Session 5aAAa

Architectural Acoustics and ASA Committee on Standards: Uncertainty in Laboratory Building Acoustic Standards

Matthew V. Golden, Cochair
Pliteq, 616 4th Street, NE, Washington, DC 20002

Daniel Urbán, Cochair
A & Z Acoustics, s.r.o., S. H. Vajanského 43, Novè Zámky, 94079, Slovakia

Chair's Introduction—7:55

Invited Papers

8:00

5aAAa1. Remarks on the definition of airborne sound insulation and consequences for uncertainties. Volker Wittstock (Physikalisch-Technische Bundesanstalt, Bundesallee 100, Braunschweig 38118, Germany, volker.wittstock@ptb.de)

Airborne sound insulation is explicitly defined as the ratio between incident and transmitted sound power. Since sound power cannot be measured directly, field quantities like sound pressure are measured to derive the desired sound power. The relation between sound pressure and sound power depends on the nature of the sound field, i.e., to which extent it is a diffuse sound field. This is the main reason why it is impossible to derive an analytic equation for the measurement uncertainty of a sound power and thus of a sound insulation. The current practice is to define standardized test facilities for the measurement of airborne sound insulation. The uncertainty of measured sound insulations is then approximated by the standard deviation of reproducibility determined by interlaboratory tests. This is equivalent to changing the definition of airborne sound insulation. It is no longer the sound power ratio but the mean value of the sound insulation measured in very many or all thinkable laboratories meeting the required specifications. Thus, laboratory specifications become part of the definition of airborne sound insulation. The contribution highlights the background of the different definitions and shows consequences for the uncertainty of airborne sound insulation.

8:20

5aAAa2. Some practical issues affecting repeatability and reproducibility in laboratory transmission loss tests. Christoph Hoeller and Jeffrey Mahn (Construction, National Res. Council Canada, 1200 Montreal Rd., Ottawa, ON K1A 0R6, Canada, christoph.hoeller@nrc.ca)

The ASTM standard E90 defines the measurement of transmission loss, equivalent to the sound reduction index defined in ISO 10140. ASTM E90 and ISO 10140 specify requirements for the laboratory, the test procedure and conditions, and for preparation and mounting of the specimen under test. Despite the strict requirements in ISO 10140 and the somewhat less strict requirements in ASTM E90, transmission loss results for nominally identical specimens often vary if measured in different laboratories, and sometimes even if measured again in the same laboratory. In practice, there are many factors that affect the repeatability or reproducibility of a transmission loss test for a given specimen. This presentation will not attempt to systematically cover all different sources of uncertainty, but instead will highlight some practical issues commonly encountered in laboratory transmission loss tests. Examples will be presented for a number of issues, including the effect of leakage through the specimen under test, the effect of varying temperature and humidity in the test chambers, and the effect of re-using gypsum board.

8:40

5aAAa3. Cross-laboratory reproducibility of sound transmission loss testing with the same measurement and installation team. Benjamin Shafer (Tech. Services, PABCO Gypsum, 3905 N 10th St., Tacoma, WA 98406, ben.shafer@quietrock.com)

Previous cross-correlative statistical research studies, combined with the results from past laboratory sound transmission loss round robin testing, illustrate that the laboratory-to-laboratory reproducibility of sound transmission loss testing is inordinately and unacceptably low. Industry building construction professionals use the results of laboratory sound transmission loss testing to determine acoustics-related building code compliance. As such, a forensic analysis of laboratory sound transmission loss is needed to narrow potential causes of cross-laboratory variability to a few primary sources. As a first step in this process, sound transmission loss measurements for two different assemblies are compared between multiple laboratories, each with their own different technicians and installation crews. Two different assemblies are then compared between multiple laboratory facilities with the same measurement and installation crew. The use of the same measurement crew at two different facilities resulted in much better statistical reproducibility than all previous reproducibility studies.

9:00

5aAAa4. Variations in impact sound level as a function of tapping machine position. John LoVerde and David W. Dong (Veneklasen Assoc., 1711 16th St., Santa Monica, CA 90404, wdong@veneklasen.com)

Impact insulation class testing per ASTM E 492 requires measurement of the sound field at exactly four tapping machine positions. Previous research by the authors [J. Acoust. Soc. Am. **121**, 3113 (2007), J. Acoust. Soc. Am. **122**, 2955 (2007)] indicated that for field tests, the variation between tapping machine positions was small. To our knowledge, a systematic investigation has not been performed for tapping machine positions in the laboratory, and some recent results indicate that the variation may be larger than expected. Large variation in sound level may be inherent to the method, or may point to problems in construction or installation of flooring materials. The variations with tapping machine position are analyzed for a set of laboratory tests, and the previous field test studies are updated with additional data. The authors investigate possible changes to the standards to mandate a maximum allowable variation between tapping machine positions, and to require additional positions as necessary.

Contributed Paper

9:20

5aAAa5. Importance of correlation between reverberation times for calculating the uncertainty of measurements according to ISO 354 and ISO 17497-1. Markus Müller-Trapet (National Res. Council, ISVR, Univ. of Southampton, Southampton SO17 1BJ, United Kingdom, M.F.Muller-Trapet@soton.ac.uk)

The calculation of measurement uncertainties follows the law of error propagation as described in the Guide to the Expression of Uncertainty in Measurements (GUM). The result can be expressed as a contribution of the variances of the individual input quantities and an additional term related to the correlation between the input quantities. In practical applications, the correlations are usually neglected. This has, e.g., led to the expression included in Annex A of ISO 17497-1 to calculate the precision of the measurement of random-incidence scattering coefficients. To determine whether it is actually justified to neglect the input correlations, this contribution investigates the correlations between the reverberation times used to determine the random-incidence absorption coefficient (ISO 354) and scattering coefficient (ISO 17497-1) in a reverberation chamber. The data used here are taken from measurements in a real-scale and a small-scale reverberation chamber. It is found that for ISO 354 correlations can be neglected. However, for ISO 17497-1, it is important to take correlations into account to obtain the correct measurement uncertainty using error propagation.

Invited Papers

9:40

5aAAa6. Addressing the lack of statistical control in acoustical testing laboratories. John LoVerde and David W. Dong (Veneklasen Assoc., 1711 16th St., Santa Monica, CA 90404, jloverde@veneklasen.com)

In order to be useful in comparing products, evaluating assemblies, or performing research, acoustical laboratory tests must be precise. This means that the “chance” variation due to any external variables must be relatively small and randomly distributed. This defines a measurement method that is in a state of statistical control, in which case the precision of the test method (the size of these small chance variations) can be measured [J. Acoust. Soc. Am. 130, 2355 (2011)]. The authors have observed many airborne and impact insulation tests performed at accredited acoustical laboratories. While controlled behavior is sometimes seen, it is also sometimes observed that a set of results shows unpredictable behavior, abrupt changes, excess scatter, or unexplained variations, which is symptomatic of a loss of statistical control [J. Acoust. Soc. Am. 137, 2216 (2015)]. It is not merely that the precision in the measurement is larger than desired; a lack of statistical control means that there are large unknown variables and the precision of the method cannot even be defined. Experiences with laboratories in which this has occurred are shared, and procedures and safeguards to address the issue are discussed.

10:00–10:20 Break

10:20

5aAAa7. Numerical study on the repeatability and reproducibility of laboratory building acoustic measurements. Arne Dijckmans, Lieven De Geetere, and Bart Ingelaere (Belgian Bldg. Res. Inst., Lombardstraat 42, Brussels B-1000, Belgium, arne.dijckmans@gmail.com)

An important issue in building acoustics is the significant variability in laboratory test results that numerous round robin tests have indicated. The current wish to include the frequency bands 50-80 Hz in the procedures to determine single-number quantities has prompted new discussions. In this paper, wave based models are used to numerically investigate the fundamental repeatability and reproducibility. Regarding sound insulation measurements, both the pressure method (ISO 10140-2) and the intensity method (ISO 15186-1 and ISO 15186-3) are investigated in the frequency range 50-200 Hz. Flanking transmission measurements (ISO 10848) are also studied. The investigation includes the repeatability of the different measurement procedures, which depends on the influence of the source and receiver positions. The reproducibility in different test facilities is studied by looking at the influence of geometrical parameters like room and plate dimensions. Increasing the number of source or receiver positions has little effect on the overall uncertainty as the reproducibility uncertainty is generally much larger than the repeatability uncertainty. For small-sized test elements, the reproducibility of the intensity method is better. For heavy walls and lightweight double constructions, however, the predicted uncertainty is similar for the three measurement methods.

Contributed Paper

10:40

5aAAa8. Reproducibility of a metric for sound reflectivity. Felicia Doggett and Sooch San Souci (Metropolitan Acoust., LLC, 40 W. Evergreen Ave., Ste. 108, Philadelphia, PA 19118, f.doggett@metro-acoustics.com)

Past attempts at developing a system able to quantify the sound reflectivity of a surface have been hindered by the shear difficulty of repeatability and reproducibility. Where and why these challenges arise, and how they can be overcome is discussed. Sound fields resulting from direct sound energy projected at various incident angles toward materials and assemblies are compared as well as how laboratory setups and specimen changeovers can affect outcomes. Can precision be increased and bias avoided? A proposed system based on software driven microprocessors and robotic precision is very fast with a high degree of precision performing at less than 0.1 mm error over 50 repeated runs. The performance in terms of repeatability and reproducibility are approximately equivalent to today's ink jet printers. As a recognized acoustic metric, the Sound Reflectivity Index could provide vital data that would accompany all products. For acousticians, a higher order of accuracy would be possible in designs involving envelopment, early reflections, speech comprehension, speech privacy, sound enhancement, and general room acoustics.

Invited Papers

11:00

5aAAa9. Comparisons of laboratory repeatability and performance when increasing reverberation chamber volume. Douglas Winker (ETS-Lindgren, Inc., 3502 Hamlet Cv, Round Rock, TX 78664, douglas.winker@ets-lindgren.com), Brian Stahke, and Michael C. Black (ETS-Lindgren, Inc., Cedar Park, TX)

ETS-Lindgren/Acoustic Systems has operated an acoustics laboratory in Austin, Texas since 1985. The original laboratory consisted of a reverberation chamber suite with a source chamber volume of 200 cubic meters and a receive chamber volume of 254 cubic meters. In 2008, ETS-Lindgren/Acoustics Systems relocated and constructed a completely new laboratory. The new lab features a reverberation chamber suite with a source chamber volume of 208 cubic meters and a receive chamber volume of 408 cubic meters. During the transition, all measurement equipment, test frames, and proficiency specimens were maintained and are constant between both laboratories. Repeatability data for ASTM E90 and ASTM C423 will be discussed and compared for each laboratory reverberation chamber suite. The changes in the results will be shown with respect to the different chamber sizes. Proficiency panels constructed to ASTM round robin guidelines have been maintained since original construction and form the baseline for this comparison. Comparison of low-frequency performance between the two chamber sizes will be emphasized with respect to current reverberation construction guidelines and limits. Intra-laboratory uncertainties over time will also be discussed and compared to the published uncertainties of the ASTM E90 and C423 standards.

11:20

5aAAa10. Update on the current ASTM building acoustics inter-laboratory studies. Matthew V. Golden (Pliteq, 616 4th St., NE, Washington, DC 20002, mgolden@pliteq.com)

In the last few years, ASTM E33 committee on Building and Environmental Acoustics has undertaken a program to improve the precision and bias statements in its laboratory and field standards. This is done through analysis of inter-laboratory studies. At the time of writing, there are approximately 10 such inter-laboratory studies either currently in process or recently completed. This paper will give a brief overview of these activates. It will highlight those studies involving laboratory building acoustics standards, including standards on sound transmission loss, impact sound transmission, sound absorption in reverberation rooms, and sound attenuation between rooms sharing a common ceiling plenum.

11:40

5aAAa11. On the uncertainty of measurement of dynamic stiffness of resilient materials. Krister Larsson (Bldg. Technology/Sound & Vibrations, RISE Res. Institutes of Sweden, Box 857, Boras SE-50115, Sweden, krister.larsson@sp.se)

The apparent dynamic stiffness of resilient materials used for example under floating floors is measured according to the standard ISO 9052-1:1989 (EN 29052-1:1992). Basically, the material under test is loaded by a load plate corresponding to 200 kg/m^2 and the resonance frequency of the first vertical mode of the mass-spring system formed by the load plate and the resilient material under test is determined. The resonance frequency then gives the dynamic stiffness. The standard allows for several excitation techniques such as swept sine, continuous noise, or impact excitation. Additionally, vibration excitation at the base as well as force excitation on the load plate is allowed. In this study, the uncertainty because of excitation of multiple vibration modes is investigated in detail. A model for the load plate on an elastic foundation representing the test setup is developed. The model is verified towards measurements and a parameter study is performed. The results show that additional modes may be excited depending on the excitation, which might lead to erroneous results. Suggestions for procedures to take the excitation of multiple modes into account and to improve the uncertainty of the method are given based on the findings.

12:00

5aAAa12. Impact of sound insulation quality in dwellings on its financial value. Andrea Vargova (Faculty of Civil Eng., Dept. of Bldg. Construction, STU Bratislava, Radlinskeho 11, Bratislava 81005, Slovakia, Andrea.Vargova@stuba.sk), Herbert Muellner (TGM Wien, Vienna, Austria), Rudolf Exel (Exel, Vienna, Austria), and Monika Rychtarikova (Faculty of Architecture, KU Leuven, Gent, Belgium)

A lot of research has been already done on the assessment and improvement of sound insulation quality of partitioning elements in dwellings. Fewer studies have addressed the impact of acoustic improvements on those elements. Very little information is available on the impact of sound insulation properties on the global real estate value of dwellings as a hole. This contribution reports on the analysis of questionnaires and interviews concerning the overall satisfaction of dwellers about the acoustic comfort at their homes. The importance that people living in apartment flats in Slovakia give to their acoustic comfort at home is addressed, with the part of their budget that they would potentially consider to spend for improving it used as a measure.

THURSDAY MORNING, 29 JUNE 2017 — ROOM 206, 8:00 A.M. TO 8:40 A.M.

Session 5aAAb

Architectural Acoustics: Topics in Architectural Acoustics Related to Materials and Modeling

Kenneth W. Good, Chair
Armstrong, 2500 Columbia Av, Lancaster, PA 17601

Contributed Papers

8:00

5aAAb1. Exploring novel beautiful, durable, hard, smooth, acoustically absorbent materials. Randall J. Rehfuss (Architecture, Virginia Tech, 224 Dunbar Ave., Dublin, VA 24084, rjrehfus@vt.edu), Michael Ermann, Martha Sullivan, Andrew Hulva (Architecture, Virginia Tech, Blacksburg, VA), and Alexander M. Kern (Mech. Eng., Virginia Tech, Darmstadt, Germany)

Acoustically absorbent materials are generally “fuzzy.” This texture converts sound energy into heat through friction, thereby reducing the amount of sound that reflects back into the room.[1] In contrast, this research explores smooth surface alternatives to speech-frequency sound absorption by retracing the efforts of Wallace C. Sabine and his partner, architect and thin-structural-tile builder Raphael Guastavino. The two, each a luminary in his respective field at the time, teamed up in the early 1900s to manufacture sound absorbent ceramic tiles. After attempting to recreate and test the Sabine-Guastavino tiles, we sought to improve upon the concept of a porous, fire-proof, and acoustically absorbent tile to include a smooth surface that is easily sanitized to be used in facilities like child care centers, schools, and hospitals. While there are materials on the market that provide a smooth surface of micro-pores that open to a “fuzzy” material behind the surface, this novel material would be a continuous unit that would enhance the aesthetics of a space. To achieve this, we used an impedance tube to test the absorption of contemporary materials ranging from uncooked ramen noodles to cementitious composites, while simultaneously applying innovative methods and technology into the firing of clay bodies to achieve the desired surface smoothness and acoustic absorption in the vocal frequency domain.

8:20

5aAAb2. Modeling the inhomogeneous reverberant sound field within the acoustic diffusion model: A statistical approach. Cedric Foy (Cerema, 11 rue Jean Mentelin, Strasbourg 67200, France, cedric.foy@cerema.fr), Vincent Valeau (Institut Pprime UPR - Bât. B17, Poitiers Cedex 9, France), Judicaël Picaut, Nicolas FORTIN (Ifsttar, Bouguenais Cedex, France), Anas Sakout (Pôle Sci. et Technologie, LaSie, La Rochelle Cedex 1, France), and Christian Prax (Institut Pprime UPR - Bât. B17, Poitiers Cedex 9, France)

In room acoustics, starting from the sound particle concept, it is now well established that the reverberant field can be modeled from a diffusion equation function of the acoustic density and a gradient equation function of the acoustic intensity. The main works on the development of an acoustic diffusion model have highlighted the major role of a coefficient of the model, the so-called diffusion coefficient. Indeed, the main phenomena influencing the reverberant sound field can be modeled by proposing an appropriate expression of this diffusion coefficient. The work presented here deals with the modeling of inhomogeneous reverberant sound fields induced by geometric disproportions, and investigates, in particular, the case of long rooms. Previously, the ability of the acoustic diffusion to model adequately the spatial variations of the sound field along the room has been demonstrated by considering a diffusion coefficient that is spatially dependent. We propose here to extend this work by determining an empirical law of the diffusion coefficient, depending on both the scattering and absorption coefficients of the walls of the room. The approach proposed here is statistical and is based on the least squares method. Several linear models are proposed, for which a rigorous statistical analysis makes it possible to assess their relevance.

Session 5aAAc

Architectural Acoustics: Simulation and Evaluation of Acoustic Environments III

Michael Vorländer, Cochair
ITA, RWTH Aachen University, Kopernikusstr. 5, Aachen 52056, Germany

Stefan Weinzierl, Cochair
Audio Communication Group, TU Berlin, Strelitzer Str. 19, Berlin 10115, Germany

Ning Xiang, Cochair
School of Architecture, Rensselaer Polytechnic Institute, Greene Building, 110 8th Street, Troy, NY 12180

Invited Papers

8:20

5aAAc1. The perceptual evaluation of acoustical environments I: Simulated environments. Stefan Weinzierl (Audio Commun. Group, TU Berlin, Strelitzer Str. 19, Berlin, Berlin 10115, Germany, stefan.weinzierl@tu-berlin.de)

The successful design and development of numerical models for the sound propagation in rooms crucially depends on the existence of appropriate measures for the quality assessment of the modeling results. In the perceptual domain, these should include single-number measures for the overall assessment of the simulation as well as differential measures, enabling experts to identify specific shortcomings in the selected modeling approach and its implementation. In the SEACEN consortium, the "authenticity" and the "plausibility" of virtual acoustic environments has been established, measuring the perceived identity with an external (given) or internal reference. Moreover, a Spatial Audio Quality Inventory (SAQI) was developed by a focus group of experts for virtual acoustics, as a metric for the differential diagnosis of virtual environments.

8:40

5aAAc2. The perceptual evaluation of acoustical environments II: Natural environments. Stefan Weinzierl (Audio Commun. Group, TU Berlin, Strelitzer Str. 19, Berlin, Berlin 10115, Germany, stefan.weinzierl@tu-berlin.de)

The perceptual evaluation of natural acoustic environments requires an assessment of the "room acoustical impression." After some theoretical considerations on the nature of this perceptional construct and the shortcomings of existing tools, we present a new, empirically substantiated approach for the development of a corresponding measuring instrument. It relies crucially on the room acoustical simulation of a representative pool of acoustical environments for speech and music and their auralization for different acoustical sources. The resulting room acoustical quality inventory, together with a database of room acoustical models and their monaural and binaural transfer functions, can be used as a ground truth for room acoustical analysis and perception.

9:00

5aAAc3. Tools for the assessment of sound quality and quality of experience in original and simulated spaces for acoustic performances. Jens Blauert (Inst. of Commun. Acoust., Ruhr-Universitaet Bochum, Bochum, North-Rhine Westphalia 44780, Germany, jens.blauert@rub.de), Jonas Braasch (Ctr. for Cognition, Commun. and Culture, Rensselaer Polytechnic Inst., School of Architecture, Troy, NY), and Alexander Raake (Inst. of Media Technol., Audio-visual Group, Tech. Univ. of Ilmenau, Ilmenau, Thuringa, Germany)

Sound quality and Quality of Experience are complex mental constructs. Their assessment, consequently, requires consideration of multiple different aspects, each of which may demand different evaluation and measurement methods going beyond current standards such as ISO 3382 for room acoustics. In this talk, relevant quality aspects are identified including signal-related, psychological, semiotic, and further cognitive ones. For each of them, available evaluation and measurement methods will be discussed, using the amount of abstraction involved as an ordering principle. Methods that can extract binaural cues from a running signal, such as those provided by BICAM (Binaurally Integrated Cross-correlation/Auto-correlation Mechanism) for room-impulse responses, or those provided by the TWO!EARS model framework (www.twoears.eu) will be highlighted, including consideration of the processing of cross-modal cues. A special focus will be put on the question of to which extent human assessors can be replaced with current algorithmic (instrumental) methods as based on computer models of the human hearing and subsequent cognitive processing. The task of collecting proper reference data will be considered, whereby results from the AABBA Initiative (Aural Assessment By means of Binaural Algorithms) and the TWO!EARS project (Reading the world with TWO!EARS) will be incorporated. [Work supported by FET-Open FP7-ICT-2013-C-618075 and NSF BCS-1539276.]

9:20

5aAAc4. Psycho-acoustic evaluation of physically-based sound propagation algorithms. Atul Rungta (Comput. Sci., Univ. of North Carolina at Chapel Hill, 250 Brooks Bldg., Columbia St., Chapel Hill, NC 27599-3175), Roberta Klatzky (Carnegie Mellon Univ., Pittsburg, PA), Ming C. Lin, and Dinesh Manocha (Comput. Sci., Univ. of North Carolina at Chapel Hill, Chapel Hill, NC, dmanocha@gmail.com)

Recently, many physically accurate algorithms have been proposed for interactive sound propagation based on geometric and wave-based methods. In terms of these applications, a key question arises whether the improved physical accuracy of these algorithms offers perceptual benefits over prior interactive methods? In this work, we present results from two studies that compare listeners' perceptual response to both accurate and approximate propagation algorithms that are used to simulate two key acoustic effects: diffraction and reverberation. For each effect, we evaluate whether increased numerical accuracy of a propagation algorithm translates into increased perceptual differentiation in interactive environments. Our results suggest that auditory perception indeed benefits from the increased accuracy, with subjects showing better perceptual differentiation when experiencing the more accurate propagation method. The diffraction experiment exhibits a more linearly decaying sound field (with respect to the diffraction angle) for the accurate diffraction method, while the reverberation experiment shows that more accurate reverberation results in better assessment of room volume and acoustic distance perception. In case of room volume, accurate reverberation, after modest user experience, results in a near-logarithmic response to increasing room volume. Finally, in the case of acoustic distance perception, accurate reverberation shows less distance compression as compared to an approximate, filter-based reverberation method.

9:40

5aAAc5. A room acoustical quality inventory. Steffen Lepa and Stefan Weinzierl (Audio Commun. Group, TU Berlin, Einsteinufer 17, Berlin, Berlin 10587, Germany, steffen.lepa@tu-berlin.de)

In a two-step procedure, a new psychological measuring instrument has been developed for the acoustical perception of room acoustical environments for speech and music. As a first step, an expert focus group of room acoustical scholars and consultants was formed in order to reach a consensus on a vocabulary as complete as possible to describe room acoustical qualities. In a second step, this inventory was used for the evaluation of 35 different simulated room acoustical environments presented by binaural synthesis to 190 subjects of different age and expert level. Based on the ratings of this room sample, a comprehensive psychometric analysis was performed in order to evaluate the preliminary item battery with respect to reliability, discriminative power, and redundancy. The resulting room acoustical quality inventory, together with the database of room acoustical models as well as their monaural and binaural transfer functions and their perceptual evaluation, can be used as a ground truth for the validation of existing and newly developed room acoustical parameters.

10:00–10:20 Break

10:20

5aAAc6. A new method for quantifying binaural decoloration based on parametrically altering spectral modulations. Andreas Haeussler and Steven van de Par (Acoust. Group, Cluster of Excellence "Hearing4all," Univ. Oldenburg, Carl-von-Ossietzky-Straße 9-11, Oldenburg 26129 Oldenburg, Germany, andreas.haeussler@uni-oldenburg.de)

The reproduction of sound in a reverberant environment leads to spectral modifications that are typically perceived as coloration. It is well known that a dichotic instead of a diotic presentation of such signals leads to reduced perception of coloration [Salomon (1995), Ph.D. Thesis, TU Delft]. In this contribution, a new method is presented for quantifying the reduction in coloration due to dichotic presentation. In this method, the first part of a Binaural Room Impulse Response, pre-dominantly responsible for the spectral envelope is extracted, and the spectral modulations are parametrically modified and reinstated on a minimum-phase Impulse Response that is convolved with pink noise and musical instruments and presented diotically to the listeners. Their task is to adaptively adjust the spectral modulations until this diotically presented stimulus sounds equally colored as a dichotically presented binaural signal. Results show that the spectral fluctuations expressed as standard deviations in decibels needs to be reduced by 1-2 dB for the diotic presentation to sound equally colored as the dichotic presentation.

10:40

5aAAc7. Inter-aural cross-correlation measured during symphony orchestra performance in big concert halls. Magne Skalevik (AKUTEK and Brekke&Strand, Bolstadtunet 7, Spikkestad 3430, Norway, msk@brekkestrand.no)

Spatial impression in concert hall listeners is known to depend on lateral reflections causing differences between the sound at the left ear and the right ear. Such differences can be measured, e.g., by the so-called inter-aural cross-correlation IACC in a binaural signal pair, i.e., a signal pair from microphones placed in the ear canal entrances. IACC data from binaural impulse responses (BRIR) are commonly reported. In contrast, little attention has been paid to running IACC, i.e. IACC(t), during music performance. Therefore, in 2011 this author launched the Binaural Project in 2011 in order to collect binaural signal data from concerts with symphony orchestras. An analysis of IACC(t) from more than 600 minutes of binaural recordings during concerts in many big concert halls is presented. Several famous halls, including Boston Symphony Hall, are included in the data. Among the questions to be answered are Can we observe from the data that concert halls make a difference to IACC(t)? If so—is this variation small or big compared to the variation from one moment to another, bar to bar, movement to movement, from one orchestra to another, and so on? On the other hand, if we cannot observe significant hall-to-hall differences, several new questions would arise, including How can we maintain that listeners are able to perceive hall-to-hall differences in ASW and LEV? And why do not the reported hall-to-hall differences in IACC from impulse responses (ISO-3382) make an observable difference to running IACC(t)?

11:00

5aAAc8. Influence of visual rendering on the acoustic judgments of a theater auralization. Barteld N. Postma (LIMSI, CNRS, Université Paris-Saclay, LIMSI, CNRS, Université Paris-Saclay, Rue John von Neumann Campus Universitaire d'Orsay, Bât 508, Orsay 91403, France, bart.postma@limsi.fr) and Brian F. Katz (Lutheries - Acoustique - Musique, Inst. ∂'Alembert, UPMC/CNRS, Paris, France)

Auralizations have become more prevalent in architectural acoustics. Auralizations in listening tests are typically presented in a unimodal fashion (audio only). However, in everyday-life one perceives complex multi-modal information. Multi-sensory research has shown that visuals can influence auditory perceptions, such as with the McGurk and ventriloquist effects. However, few studies have investigated the influence of visuals on room acoustic perception. Additionally, in the majority of previous studies, visual cues were represented by photographs either with or without visuals of the source. Previously, a virtual reality framework combining a visible animated source in a virtual room with auralizations was conceived enabling multi-modal assessments. The framework is based on BlenderVR scene graph and visual rendering with MaxMSP for the real-time audio rendering of 3rd order HOA room impulse responses (RIRs) in tracked binaural. CATT-Acoustic TUCT was used to generate the HOA RIRs. Using this framework, two listening tests were carried out: (1) a repeat of a prior audio-only test comparing auralizations with dynamic voice directivity to static orientation and (2) a test comparing dynamic voice auralizations with coherent or incoherent visuals with respect to seating position. Results indicate that judgments of several room acoustic attributes are influenced by the presence of visuals.

11:20

5aAAc9. Audio-visual room perception: Cross-modal and interaction effects put to test. Hans-Joachim Maempel and Michael Horn (Staatliches Institut für Musikforschung, Tiergartenstraße 1, Berlin, Berlin 10785, Germany, m.horn@posteo.de)

The perception of rooms involves different modalities, particularly hearing and sight. Fundamental issues such as the acoustical and optical shares in certain perceptual features have, however, not been experimentally addressed yet. We investigated to what extent the acoustical and optical properties of performance rooms influenced auditory and visual features. Specifically, cross-modal effects and interaction effects were a matter of particular interest. We also quantified the respective proportion of acoustical and optical information accounting for the perceptual features. The main preconditions for such an undertaking are the dissociation of the acoustical and optical components of the stimuli, the commensurability of these components, and rich cue conditions. We acquired binaural room impulse responses and panoramic stereoscopic images of six rooms in order to recreate these rooms virtually, and added recordings of both a music and a speech performance by applying dynamic binaural synthesis and chroma-key compositing. By the use of a linearized extra-aural headset and a semi-panoramic stereoscopic projection system we presented the scenes to test participants and asked them to rate unimodal features such as loudness, highs, lows, clarity, reverberance, and envelopment as well as brightness, contrast, color intensity, and hue. The statistical analyses indicate a straight-forward processing of low-level features.

11:40

5aAAc10. Binaural auralization of proposed room modifications based on measured omnidirectional room impulse responses. Christoph Pörschmann and Philipp Stade (Technische Hochschule Köln, Betzdorfer Str. 2, Cologne 50679, Germany, christoph.poerschmann@th-koeln.de)

The auralization of rooms with dynamic binaural synthesis using binaural room impulse responses (BRIRs) is an established approach in virtual audio. The BRIRs can either be obtained by simulations or by measurements. Up to now changed acoustical properties, as they occur when a room is altered in a renovation, cannot easily be considered in a measurement-based approach. This paper presents a new method to auralize modifications of existing rooms. The authors already have shown in a previous publication that such an auralization can be done by appropriately shaping the reverberation tail of an impulse response. Furthermore, the authors have presented an approach to synthesize BRIRs based on one omnidirectional room impulse response (RIR). In this paper, both methods are combined: A single measured omnidirectional RIR is enhanced and adapted to create a binaural representation of a modified room. A listening experiment has been performed to evaluate the procedure and to investigate differences between synthesized and measured BRIRs. The advantages of this method are obvious: Planned room modifications can be made audible without complex measurements or simulations; just one omnidirectional RIR is required to provide a binaural representation of the desired acoustic treatment.

Contributed Paper

12:00

5aAAc11. Azimuthal localization in 2.5D near-field-compensated higher order ambisonics. Fiete Winter, Nara Hahn (Inst. of Communications Eng., Univ. of Rostock, Universität Rostock - Institut für Nachrichtentechnik, R.-Wagner-Str. 31 (Haus 8), Rostock 18119, Germany, fiete.winter@uni-rostock.de), Hagen Wierstorf (Audiovisual Technol. Group, Technische Universität Ilmenau, Ilmenau, Germany), and Sascha Spors (Inst. of Communications Eng., Univ. of Rostock, Rostock, Germany)

Sound Field Synthesis approaches aim at the reconstruction of a desired sound field in a defined target region using a distribution of loudspeakers. Near-Field Compensated Higher Order Ambisonics (NFCHOA) is a prominent example of such techniques. In practical implementations different artifacts are introduced to the synthesized sound field: spatial aliasing is caused by the non-zero distance between the loudspeakers. Modal bandwidth limitation is a well-established approach to reduce spatial aliasing in 2.5D NFCHOA, but introduces temporal and spectral impairments to the reproduced sound field which strongly depend on the relative position to the center of modal expansion. Also, the dimensionality mismatch in a 2.5D synthesis scenario results in a different amplitude decay compared to the desired sound field. Listening experiments already investigated the azimuthal localization in 2.5D NFCHOA. It is however unclear, in how far individual artifacts caused by spatial sampling, modal bandwidth limitation, and the 2.5D dimensionality mismatch contribute to these localization impairments in particular. Within this contribution a mathematical framework is used together with binaural synthesis to simulate the individual effect of each artifact on the ear signals. Humans' performance is approximated by a binaural model for azimuthal localization.

Session 5aAAd

Architectural Acoustics: Recent Developments and Advances in Archeo-Acoustics and Historical Soundscapes III

David Lubman, Cochair
DL Acoustics, 14301 Middletown Ln., Westminster, CA 92683-4514

Miriam A. Kolar, Cochair
Architectural Studies; Music, Amherst College, School for Advanced Research, 660 Garcia St., Santa Fe, NM 87505

Elena Bo, Cochair
DAD, Polytechnic Univ. of Turin, Bologna 40128, Italy

Chair's Introduction—9:15

Invited Papers

9:20

5aAAd1. Music and sound of the 17th Century:Athanasius Kircher and his Phonosophia anacamptica. Lamberto Tronchin (DIN - CIARM, Univ. of Bologna, Viale del Risorgimento, 2, Bologna I-40136, Italy, lamberto.tronchin@unibo.it) and David J. Knight (Univ. of Southampton, Guelph, ON, Canada)

In the 17th Century many Physicists, Mathematicians and Musicians dealt with the experiences of harmony, music, and sound propagation in enclosed interior spaces. Among them, Athanasius Kircher was one of the most influential researchers of his time. Born in Geisa, Thüringia (Germany), he became a Jesuit in 1608 and spent a large part of his life in Rome, where he died in 1680. During his lifetime, he wrote several books spanning a wide range of topics, including sound, music, and acoustics. One of these, the Phonurgia Nova, published in 1673, was almost ignored for hundreds of years. Phonurgia Nova was translated from the original Latin. It consists of two different books, the *Phonosophia nova* and the *Phonosophia anacamptica*. The former deals with the influence of music on human beings whereas the latter analyses sound propagation in enclosed spaces. In this paper, the Authors present new achievements regarding some of the apparatuses that Kircher invented. Among all his *marvelous sound machines*, the Authors will describe some of Kircher's items, including the tuba stentorophonica (the "loud trumpet"), the statua citofonica (the "talking statue"), the obiectum phonocampticum (the "phonocentric object"), the Ruota Cembalaria (the "sounding wheel"), the ancient Egyptian singing statue of Memnon, the Aeolian Harp, and the hydraulis (hydraulic organ). Some of these apparatuses were also recently partially realized by the Polish Pavilion during the *Biennale of Venice* in 2012, achieving a Special Mention from the international jury.

9:40

5aAAd2. Seeking the sounds of ancient horns. D. Murray Campbell (Acoust. and Audio Group, Univ. of Edinburgh, James Clerk Maxwell Bldg., Mayfield Rd., Edinburgh EH9 3JZ, United Kingdom, d.m.campbell@ed.ac.uk), Joël GILBERT (Laboratoire d'Acoustique de l'Université du Maine - CNRS, LE MANS, France), and Peter Holmes (Designer in Residence, Middlesex Univ., London, United Kingdom)

Recent archaeological discoveries, most notably at the Gallo-Roman site at Tintignac in the Corrèze district of France, have thrown fresh light on the nature of some of the lip-excited wind instruments used in Europe around two thousand years ago. In particular, it has been possible to reconstruct working copies of the Celtic horn known as the carnyx, and to experiment on the reproductions both scientifically and musically. A number of Etruscan and classical Roman brasswind instruments, including the lituus and the cornu, have also been reproduced and tested under the auspices of the European Music Archaeology Project. This paper reviews some of this work, and discusses the usefulness of acoustical modeling and measurement in interpreting the possible musical functioning of these ancient horns.

10:00

5aAAd3. Reconstructing human music with hard evidence. Jelle Atema (Biology, Boston Univ., 5 Cummington Mall, Boston, MA 02215, atema@bu.edu)

Philosophers have long discussed the origins of music and language and used them as an argument to set humans apart from other animals. For evidence we rely on archaeology, anthropology and even biology, and on historical depictions and descriptions. In the case of music we have some hard evidence in the form of ancient bone flutes. Their physical reconstructions are important tools in the quest for origins, but open to interpretation and controversy. As a biologist-flutist I have been skeptical of many resulting "models" that have been proposed. Here I submit some of the main obstacles I encounter in reconstructing 4,000 to 50,000 year-old bone flutes in hopes of hearing their music. How do we define what is a flute? Which physical reconstruction is most credible? Which sounds can it make? Is there a recognizable scale and is that scale credibly constrained? Which sounds constitute music? Is “our” music related to “their” music? While flutes have been preserved in the pre-historic record and their finger holes suggest a musical scale, the oldest evidence is sufficiently weak to allow for many interpretations and vociferous debates. I will demonstrate these questions with a few reconstructions and different types of flutes.

10:20–10:40 Break

10:40

5aAAd4. Archaeological auralization as fieldwork methodology: Examples from Andean archaeoacoustics. Miriam A. Kolar (Res. Associate, Five Colleges, Inc., School for Adv. Res., 660 Garcia St., Santa Fe, NM 87505, mkolar@fivecolleges.edu)

Auralization, the computational rendering of sound for listeners, enables archaeoacoustical reconstructions. In archaeoacoustics research, computational tools and analyses frequently enmesh with human performance. Broadening the definition of archaeological auralization to encompass the investigative process of specifying and enacting the re-sounding of archaeological spaces, objects, and events positions auralization as a methodology for the sensory exploration of anthropological research questions. A foundational tool for archaeoacoustical and archaeomusicological fieldwork, auralization allows contextualized testing and measurement of spatial and instrumental acoustics, along with their perceptual evaluation. Case-study examples from Andean archaeoacoustics research include auralizations of reconstructed architectural acoustics, and in-situ loudspeaker playback of recorded performances of 3,000-year-old conch shell horns, delivered as auditory perceptual experiment stimuli within the extant ceremonial architecture at Chavín de Huántar, Peru. Performed plaza auralizations at the Inka administrative city Huánuco Pampa re-enacted sound transmission dynamics in that Pre-Columbian public space, enabling present-day listeners to evaluate verbal intelligibility, among other tests. As a fieldwork methodology, archaeological auralization is both process and product: the specification and physical sounding of concepts and data, to be observed and evaluated in relationship with archaeological materials and knowledge.

11:00

5aAAd5. The acoustics and sound environments of Early Delta Blues. Mark H. Howell (MS Dept. of Archives and History, Winterville Mounds, 2415 Hwy. 1 nort, Greenville, MS 38703, mhrabinal@gmail.com)

As part of a larger investigation into the archaeological origins of the blues, I seek to re-create the sound environments that surrounded and informed the initial emanations of this important American music genre. This study is warranted because the music has had such a pronounced influence on global musics, as well as on broader sociological concerns like race, regionalism, nationalism, capitalism, and gender. It is also of value for archaeo-acoustic studies in that it concerns a study of an archaeo-historic site that is not in the deep past. For the ASA meeting-forum I will report on two related acoustic processes: one is the digital reconstruction of the physical structures where the blues was first heard, such as wood-framed shotgun-shaped performance spaces (juke joints, general stores), some of which still exist in the Mississippi Delta (Po Monkeys near Merigold and Mary’s in Indianola), or can be archaeologically recovered (the general store at Dockery Farms); and the second process is a recreation of the sonic environment of the late 19th early 20th century south of the lower Mississippi river valley, including natural and anthropomorphic sounds. These steps precede a future goal, the archaeologically reconstructed instrumental and vocal sounds of incipient and early blues, allowing for a truer picture of this musical phenomenon than currently exists.

11:20

5aAAd6. Üneholisunn: Proposal for a new descriptive language of sound. Jeff Benjamin (Columbia Univ., P.O. 42, West Shokan, NY 12494, jlb2289@columbia.edu)

The apprehension of historic sound relies upon a philosophical shift from representation to presence: we do not need electronic capture to preserve sonic forms. In this paper, I will argue that most of the sounds we hear are old sounds, glazed with the patina of novelty. The dialog that persists between different fundamental conceptions of sound is instructive, but the assertion of sound as artifact, or sonifact (as a material thing that endures through time)provides a useful way of thinking about historic sound in particular. This points to the somewhat pressing need for an adequate descriptive language of sound, a colloquial way of expressing the abundance of sonifacts all around us. In this paper, and specifically pertaining to landscapes of industrial ruination, I intend to offer some suggestions for the possible development of such a language. After an initial description of this project, I will read a series of poems using these words to demonstrate a possible method to convey sonic information in the vernacular. Many of the root words for this agglutinative sonic language will be drawn from some of the world’s disappearing languages.

11:40–12:20 Panel Discussion

Session 5aABa

Animal Bioacoustics: Ecosystem Acoustics I

Susan Parks, Cochair
Biology, Syracuse University, Biology, 107 College Place, RM 114, Syracuse, NY 13244

Jennifer L. Miksis-Olds, Cochair
Center for Coastal and Ocean Mapping, Univ. of New Hampshire, 24 Colovos Rd., Durham, NC 03824

Denise Risch, Cochair
Ecology, Scottish Association for Marine Science (SAMS), SAMS, Oban PA371QA, United Kingdom

Chair's Introduction—7:55

Invited Papers

8:00

5aABa1. Overview: Ecoacoustics for monitoring freshwater and marine biodiversity. Denise Risch (Ecology, Scottish Assoc. for Marine Sci. (SAMS), SAMS, Oban, Argyll PA371QA, United Kingdom, denise.risch@sams.ac.uk) and Susan Parks (Biology, Syracuse Univ., Syracuse, NY)

Global marine and freshwater ecosystems are experiencing an unprecedented loss and re-distribution of biodiversity, due to far reaching effects of human activities, including accelerated climate change and over-exploitation. Such changes in aquatic diversity patterns will lead to shifting baselines with respect to species richness and distribution, which need to urgently be monitored. Due to its applicability in surveying remote areas over extended timescales, ecoacoustics play a vital part in monitoring such large-scale changes. Aquatic ecoacoustics is a field that is expanding rapidly alongside emerging underwater technologies, including gliders and real-time passive acoustic buoys, as well as analytical approaches for assessing ecosystem health. These tools can also be used to monitor changes in abiotic environmental factors, including precipitation and wind events, as well as contributions of anthropogenic noise to the overall soundscape. Concerns about the increasing impact of particularly long range and ubiquitous noise sources such as global shipping traffic or seismic surveys, necessitate approaches to monitor their relative influence on aquatic soundscapes. This review will examine the use of ecoacoustic approaches to monitor freshwater and marine environments, identify gaps of knowledge, and provide recommendations for future applications of ecoacoustic tools to aide in the conservation of freshwater and marine biodiversity.

8:20

5aABa2. Implantation of marine ecoacoustic indices. Craig A. Radford (Inst. of Marine Sci., Univ. of Auckland, PO Box 349, Warkworth 0941, New Zealand, c.radford@auckland.ac.nz)

Diversity measurement techniques can present logistical and financial obstacles to conservation efforts. Ecoacoustics has recently emerged as a promising solution to these issues, providing a mechanism for measuring diversity using acoustic indices, which have proven to be beneficial in terrestrial habitats. During summer in temperate north eastern New Zealand acoustic and traditional biodiversity surveys were conducted. Three ecoacoustic indices originally developed for terrestrial use were then compared to three species assemblage diversity measures and compared using Pearson correlations. Acoustic Complexity Index (ACI) was significantly correlated with Pielou's Evenness (J') and Shannon's index (H'). Wind did not affect any of the acoustic indices. As anthropogenic noise was included in these investigations, both ACI and H were considered robust to its presence. However, all these relationships break down using recordings taken in winter even though the traditional diversity measures remains consistent. The development of marine acoustic indices is only in its early stages, but there are significant questions around trying to implant what has been achieved in terrestrial ecosystems rather than developing specific indices for the marine environment.

8:40

5aABa3. Do bioacoustic conditions reflect species diversity? A case study from four tropical marine habitats. Erica Staaterman (Smithsonian Inst., 647 Contees Wharf Rd., Edgewater, MD 21037, staatermane@si.edu)

New tools, such as passive acoustic monitoring, can be helpful for measuring levels of biodiversity in habitats that are otherwise difficult to sample. Here, we tested the utility of acoustic measurements in shallow coastal waters by conducting simultaneous bioacoustic and biodiversity surveys in four habitat types in Panama: mangrove, reef, seagrass, and sand. We found that acoustic measurements in the "low band" ($<$1000 Hz) were positively correlated with cryptic fish species richness. However, our 24-h acoustic recordings revealed a clear toadfish chorus at dusk, which masked other fish sounds and confounded results from newer acoustic indices such as acoustic

5a THU. AM

entropy and acoustic complexity. Band level in the "high band" (3,000-10,000 Hz) did not differ across habitat types and was not significantly correlated to biodiversity measurements. Our study demonstrates that bioacoustic surveys can help scientists to identify certain cryptic, soniferous species, and should be used in tandem with traditional biodiversity surveys. Additional research is needed in the marine environment to validate the utility of the newer acoustic indices.

9:00

5aABa4. Soundscape analyses for ecosystem conservation. Amandine Gasc (Institut de Systématique, Evolution, Biodiversité, museum national d'histoire naturelle, 45 rue buffon, Paris 75005, France, amandine.gasc@gmail.com)

Large scale spatial and temporal analyses help to appreciate and subsequently act to avoid or reduce ecosystem disturbances. For example, remote sensing imagery aid in understanding large scale dynamics of ecosystems; however, the capture of the animal community component of the ecosystem is challenging. Soundscape analyses provide a new insight into animal community and ecosystem ecological levels with a detailed temporal resolution, considered here as a complementary approach. Therefore, the focus of this presentation is to: (1) summarize the recent research advances in the estimation of composition and dynamics of animal acoustic community and (2) present recent research utilizing acoustic community in disturbance detection and its potential for ecosystem restoration. Initial results from the scientific community provide robust elements to support the use of soundscape measurements to evaluate disturbance impacts on animal community and natural ecosystems. Additionally, interest from natural area managers in application of soundscape techniques is largely confirmed. However, additional research is still necessary to develop robust and calibrated methods and tools for concrete biological conservation action. Three objectives are highlighted as future directions of this research: (1) develop soundscape metrics and improve their interpretation, (2) improve the understanding of soundscape drivers, and (3) develop soundscape-based disturbance indicators.

9:20

5aABa5. How often is human auditory detection in natural environments limited by our absolute hearing thresholds? Kurt M. Fristrup (Natural Sounds and Night Skies Div., National Park Service, 1201 Oakridge Dr., Ste. 100, Fort Collins, CA 80525, kurt_fristrup@nps.gov) and Damon Joyce (Natural Sounds and Night Skies Div., National Park Service, Fort Collins, CO)

The National Park Service (NPS) has collected long-term sound level measurements from more than 800 sites, using equipment that measures 1 second, $1/3^{rd}$ octave band levels. One-third octave bands approximate the critical bands for the human auditory system, and the initial motivation for collecting these data was to predict the levels at which incoming aircraft noise would be audible. Here, we will compare the nominal human threshold of hearing—expressed in $1/3^{rd}$ octave bands—with an appropriate summary of the background or residual sound level of each environment. Several procedures have been recommended for estimating the residual sound level, or the sound level that remains after energy from nearby and transient sound sources is removed. The merits of alternative procedures for estimating residual sound levels will be assessed. The NPS data show that human hearing has evolved to take advantage of the quietest conditions that occur in all but the very quietest environments, across a substantial portion of the audible spectrum. For species whose hearing is up to 10 dB less sensitive than humans, they will be operating in a masked hearing regime in the majority of locations and hours.

9:40

5aABa6. Large scale passive acoustic recording efforts improve our understanding of long term changes in marine mammal ecology and distribution. Sofie M. Van Parijs, Danielle Cholewiak, Genevieve Davis (NOAA Fisheries, 166 Water St., Woods Hole, MA 02543, sofie.vanparijs@noaa.gov), and Mark F. Baumgartner (Biology, Woods Hole Oceanographic Inst., Woods Hole, MA)

Collaborative efforts across a large number of scientists working throughout the Western Atlantic Ocean has led to sharing of passive acoustic data spanning over a decade. These data allow for a 24/7 lens to be cast providing a long term temporal understanding of species presence. This collaborative approach has allowed for unprecedented research focusing on long term patterns and changes in distribution and movements of baleen whales and odontocetes. Results from these data shows how baleen whales migration paths can be defined and changes in these paths can be detected over time. Additionally, they show that whales are present at times of the year and in regions that were not previously documented, particularly during winter months. Beaked whale composition within and between each shelf break canyon where recordings are available varies considerably, demonstrating latitudinal as well as regional gradients in species presence. The addition of ambient noise curves to this mix provides context and allows for the evaluation of anthropogenic noise. This long term big picture view of species presence and movements improves our capacity to infer whether observed changes are a result of ecological, climatological, or anthropogenic factors.

10:00

5aABa7. Soundscape planning: An acoustic niche for anthropogenic sound in the ocean? Ilse Van Opzeeland and Olaf Boebel (Ocean Acoust. Lab, Alfred-Wegener Inst. for Polar and Marine Res., Am Alten Hafen 26, Bremerhaven 27568, Germany, ilse.van.opzeeland@awi.de)

In analogy to landscape planning, the concept of soundscape planning aims to reconcile potentially competing uses of acoustic space by managing the anthropogenic sound sources. We present here a conceptual framework to explore the potential of soundscape planning in reducing (mutual) acoustic interference between hydroacoustic instrumentation and marine mammals. The basis of this framework is formed by the various mechanisms by which acoustic niche formation occurs in species-rich communities that acoustically coexist while maintaining hi-fi soundscapes, i.e., by acoustically partitioning the environment on the basis of time, space, frequency and/or signal form. Hydroacoustic measurements often exhibit certain flexibility in the timing, signal characteristics and even instrument positioning, potentially offering the opportunity to minimize the underwater acoustic imprint. We evaluate how the principle of acoustic niches (i.e., the partitioning of the acoustic space) could contribute to reduce potential (mutual) acoustic interference based on actual acoustic data from various recording locations in polar oceans.

10:20–10:40 Break

10:40

5aABa8. Spatio-temporal distribution of beaked whales on Canada's East Coast. Bruce Martin (Oceanogr., Dalhousie Univ., 32 Troop Ave., Ste. 202, Dartmouth, NS B3B 1Z1, Canada, bruce.martin@jasco.com), Julien Delarue, Katie Kowarski (JASCO Appl. Sci., Dartmouth, NS, Canada), Hilary Moors-Murphy (Bedford Inst. of Oceanogr., Dartmouth, NS, Canada), and Joanna Mills Flemming (Mathematics and Statistics, Dalhousie Univ., Halifax, NS, Canada)

Beaked whales represent some of the least understood marine mammals worldwide with the movements and distribution of many species largely unknown. Around eastern Canada current knowledge is limited to the eastern Scotian shelf and northern bottlenose whales. The acoustic signals of the beaked whale species are recognizable and sufficiently unique to be candidates for passive acoustic monitoring. Thirteen deep-water recorders located along the shelf break off Eastern Canada from Bonnécamps Canyon (42.5 N) to Southern Labrador (55.3 N) collected acoustic data near-continuously from Aug 2015 to July 2016. A minimum of one of every 20 minutes were recorded at 250,000 samples per second to monitor for the presence echolocations clicks of odontocetes, including beaked whales. An automated detector, validated by manual analysts, identified the presence of endangered Northern Bottlenose whales (*Hyperoodon ampullatus*), Cuvier's (*Ziphius cavirostris*), and special concern Sowerby's beaked whales (*Mesoplodon bidens*). The presence data were analyzed to determine the occurrence and residency durations of beaked whales throughout the geographic range studied. We then studied the influence of currents, sea ice, surface temperatures, chlorophyll, distance to the 1000 m isobath, background noise and anthropogenic noise on the whales' acoustic occurrence. Acoustic studies such as this allow us to gain further insights on the occurrence of these notoriously difficult to study species.

11:00

5aABa9. Seasonal acoustic ecology of beluga and bowhead whale core-use areas in the Pacific Arctic. Kathleen Stafford (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, stafford@apl.washington.edu), Manuel Castellote (JISAO, Univ. of Washington, Seattle, WA), Melania Guerra (Appl. Phys. Lab., Univ. of Washington, Seattle, WA), and Catherine L. Berchok (Marine Mammal Lab, NOAA, Seattle, WA)

The acoustic ecology of Arctic marine mammals is driven by anthropogenic, biotic, and abiotic factors each of which may influence the behavioral ecology of each species. The acoustic environment of bowhead *(Balaena mysticetus*) and beluga (*Delphinapterus leucas*) whales in three core-use regions of the Pacific Arctic was examined during the months in which both species occur in these regions. The Anadyr Strait region in winter was dominated by the signals of bowhead whales, walrus and bearded seals. In Bering Strait in late fall and winter, wind noise predominated in November but once the region was ice-covered, bowhead and walrus were the main sources of noise. Barrow Canyon in late summer and fall was the only region in which anthropogenic sources overlapped with both whale species. Overall, ambient noise levels were low in the Pacific Arctic when compared to other ocean basins in which anthropogenic noise dominates low frequencies. However, climate change-driven increases in open water are leading to rising noise levels from increased human use of the Arctic, increased storminess, and increased presence of vocal subarctic whales. These "new" sources of sound may be altering the underwater soundscape and possibly influencing the acoustic ecology of Pacific Arctic cetaceans.

11:20

5aABa10. Variability in coral reef soundscapes, spatiotemporal differences, biophysical and behavioral drivers, and associations with local biota. T. Aran Mooney, Ashlee Lillis, Maxwell B. Kaplan (Biology Dept., Woods Hole Oceanographic Institution, 266 Woods Hole Rd., Woods Hole, MA 02543, amooney@whoi.edu), Justin Suca (Biology Dept., Woods Hole Oceanographic Instituition, Falmouth, MA), and Marc Lammers (HIMB, Univ. of Hawaii, Kaneohe, HI)

Coral reefs harbor some of the highest biodiversity on the planet. Their rich ecoacoustic soundscape may provide a way to track both animal activities and community level structure. To do so, it is critical to identify how reef soundscapes are influenced by biotic and abiotic parameters, and establish how soundscapes change over time and across habitats. Here we present results from 18 coral reefs in the U.S. Virgin Islands and Maui, Hawaii, with the overall goals to quantify soundscape variability across multiple spatial and temporal scales (days to years), test how soundscape parameters relate to local biological communities, and address how biophysical parameters (light, temperature, and rugosity) influence these eco-soundscapes. Acoustic measurements were made in-tandem with benthic and fish visual surveys. Analyses were carried out using high and low-frequency bands corresponding to the primary soniferous taxa on reefs, snapping shrimp and fish. Overall, these results indicate that certain acoustic metrics can be linked to visual survey results. Snapping shrimp exhibit complex spatiotemporal patterns, with strong diel rhythms shifting over time and varying substantially over short spatial scales. Furthermore, long-term recordings are necessary to provide a robust baseline measurement of acoustic variability and better quantify changes in coral reef ecosystems.

Contributed Papers

11:40

5aABa11. Rapid extraction of ecologically meaningful information from large-scale acoustic recordings. Megan F. McKenna (Natural Sounds and Night Skies Div., National Park Service, 1201 Oakridge Dr., Ste. 100, Fort Collins, CO, megan_f_mckenna@nps.gov), Rachel Buxton (Fish, Wildlife, and Conservation Biology Dept., Colorado State Univ., Fort Collins, CO), Mary Clapp (Evolution and Ecology Dept., Univ. of California, Davis, Davis, CA), and Erik Meyer (Sequoia & Kings Canyon National Parks, Three Rivers, CA)

Acoustic recordings have the potential to address a suite of important conservation questions, from assessing phenology shifts due to climate change, to examining the impact of anthropogenic noise on wildlife, to monitoring biodiversity at enormous spatio-temporal scales. However, consistent methods are required to extract meaningful information from these large datasets. Here we apply a method of calibrating recordings to standardize acoustic data collected at over 50 unique sites in a diversity of habitats across the continental U.S. using a variety of recording units and parameters. The calibration method results in a coarser data resolution, decreasing storage space and computation time of further analysis. We then apply recently developed acoustic indices to evaluate biodiversity in our recordings. A review of existing acoustic indices and degree of correlation with bioacoustic activity, species richness, functional diversity, landscape attributes, and anthropogenic influence guided our decisions about what indices to implement. Resulting indices were compared with the diversity of birds from observer point counts and from animal vocalizations observed in the recording spectrograms and to anthropogenic sounds observed in the recordings. The results provide important insight on the utility of each index, or group of indices, to investigate dynamics of ecological communities across large scales.

12:00

5aABa12. Leveraging big data: How acoustic archives facilitate ecosystem research. Carrie C. Wall, Charles Anderson (Cooperative Inst. for Res. in Environ. Sci., Univ. of Colorado at Boulder, 216 UCB, Boulder, CO 80309, carrie.bell@colorado.edu), J. Michael Jech, Sofie M. Van Parijs (Northeast Fisheries Sci. Ctr., NMFS, Woods Hole, MA), Leila Hatch (Stellwagen Bank National Marine Sanctuary, NOS, Scituate, MA), and Jason Gedamke (Sci. & Technol., NMFS, Silver Spring, MD)

The National Oceanic and Atmospheric Administration's (NOAA) National Centers for Environmental Information (NCEI) has developed archives for the long-term stewardship of active and passive acoustic data. Water column sonar data have been collected for fisheries and habitat characterization over large spatial and temporal scales around the world, and archived at NCEI since 2013. Protocols for archiving passive acoustic data are currently being established in support of the NOAA Ocean Noise Reference Station Network project, and monitoring marine mammals and fish. Archives maintain data, but access to these data is a core mission of NCEI that allows users to discover, query, and analyze the data in new and innovative ways. Visualization products continue to be developed and integrated into the data access portal so that researchers of varying backgrounds can easily understand the quality and content of these complex data. Spatially and temporally contemporary oceanographic and bathymetric data are also linked to provide an ecosystem-wide understanding of the region. Providing access and facilitating the utility of these data for ecoacoustics research are ongoing efforts at NCEI, and would benefit from input from the acoustics community.

THURSDAY MORNING, 29 JUNE 2017 BALLROOM A, 8:00 A.M. TO 12:20 P.M.

Session 5aABb

Animal Bioacoustics: Topics in Animal Bioacoustics (Poster Session)

Aaron Thode, Chair
SIO, UCSD, 9500 Gilman Dr., MC 0238, La Jolla, CA 92093-0238

All posters will be on display from 8:00 a.m. to 12:20 p.m. To allow contributors in this session to see the other posters, authors of odd-numbered papers will be at their posters from 8:00 a.m. to 10:10 a.m. and authors of even-numbered papers will be at their posters from 10:10 a.m. to 12:20 p.m.

Contributed Papers

5aABb1. Listening for right whales off Brazil: Present knowledge and future research. Julia R. Dombroski, Susan Parks (Dept. of Biology, Syracuse Univ., 107 College Pl., Syracuse, NY 13210, jribeiro@syr.edu), Karina R. Groch (Projeto Baleia Franca, Imbituba, Brazil), Paulo A. Flores (APA Anhatomirim, Instituto Chico Mendes para Conservação da Biodiversidade, Florianopolis, Brazil), and Renata S. Sousa-Lima (Departamento de Fisiologia, Universidade Federal do Rio Grande do Norte, Natal, Brazil)

In the Southwest Atlantic, a key southern right whale wintering ground is found off southern Brazil. Aiming to collect reference information on the acoustic ecology of right whale mother-calf pairs in the region, we used two complementary passive acoustic monitoring methods. Recordings from autonomous archival devices were used to obtain the description of pairs' vocal repertoire: call classes were established, temporal and frequency parameters of calls were measured and reported, and the existence of a diel pattern of vocal activity was investigated. Calling rate and contextual call usage were obtained through synchronized behavioral observations and acoustic recordings made using a dipping two-unit linear array. Current knowledge about the species vocal behavior off Brazil allows an increment in the use of PAM methods as a research tool. Therefore, our plans for future research include the use of multistory tags, sound propagation, and playback experiments, and long-term deployment of autonomous devices along the whale's concentration area. Future results will contribute to enhancing the knowledge on the species communication system and will aid habitat management decisions.

5aABb2. Variation in dawn chorus acoustic complexity across a global latitudinal gradient. Colin R. Swider, Susan Parks (Biology, Syracuse Univ., 114 Life Sci. Complex, 107 College Pl., Syracuse, NY 13244, cswider@syr.edu), and Mark V. Lomolino (Environ. and Forest Biology, SUNY College of Environ. Sci. and Forestry, Syracuse, NY)

The geography of biological sound is a largely unexplored topic that exists at the boundary of ecoacoustics and biogeography. Identification and characterization of patterns of biological sound and soundscape variation across the planet may provide insight into the potential for acoustics as a tool for biodiversity monitoring and conservation. Ecoacoustic indices are audio signal measurements that can be useful for revealing, characterizing, and comparing such soundscape patterns. Some of these indices, such as the Acoustic Complexity Index (ACI), have been shown to correlate with biodiversity and avian vocalization activity in some systems. The goal of this study was to investigate broad-scale trends in the ACI by testing whether the acoustic complexity of dawn chorus soundscapes follows the known latitudinal gradient in avian biodiversity. The ACI was calculated for dawn chorus recordings from 187 sites worldwide, spanning from -45.4° to 68.1° latitude. Acoustic complexity was expected to be highest near the equator and decrease with absolute latitude, tracking the general avian diversity gradient, because higher avian diversity should result in more elaborate dawn chorus soundscapes. However, the ACI did not track the general avian latitudinal diversity gradient. The results and potential explanations for this trend will be discussed.

5aABb3. Temporal and spatial distribution patterns of toothed whales in Massachusetts Bay using passive acoustics from ocean gliders. Tammy Silva (Biology, Univ. of Massachusetts Dartmouth, 285 Old Westport Rd., Dartmouth, MA 02747, tsilva4@umassd.edu), T. Aran Mooney, Laela Sayigh, and Mark F. Baumgartner (Biology, Woods Hole Oceanographic Inst., Woods Hole, MA)

Basic information on marine mammal habitat use is necessary for informing ecosystem-based management plans and for mitigating human impacts. Massachusetts Bay is an important marine mammal foraging area in the Gulf of Maine and a region of high human activity, but little is known about toothed whale habitat use, particularly during winter months. Passive acoustic monitoring, particularly from autonomous platforms, provides advantages in studying habitat use in unfavorable conditions. The goal of this work is to use acoustic monitoring to investigate temporal and spatial occurrence patterns of toothed whale species in Massachusetts Bay during late fall/early winter using ocean gliders equipped with passive acoustic recorders. Slocum gliders were deployed in western Massachusetts Bay in 2014, 2015, and 2016. Toothed whales were detected on 92 (72%) of 128 deployment days. Detections occurred more often at night. Ongoing work includes acoustic identification of species and assessing relationships between detections and environmental conditions. These data provide the first evidence of a consistent presence of toothed whales in Massachusetts Bay during late fall and winter, and demonstrate the potential for ocean gliders as a tool to detect toothed whales in areas/times that are difficult to survey with traditional visual methods.

5aABb4. Automated classification of Pacific white-sided dolphin (*Lagenorhynchus obliquidens*) pulsed calls for diel pattern assessment. Kristen Kanes, Stan E. Dosso (Univ. of Victoria, 3800 Finnerty Rd., Victoria, BC V8P 5C2, Canada, kristenk@uvic.ca), Tania I. Lado (Ocean Networks Canada, Victoria, BC, Canada), and Xavier Mouy (Jasco Appl. Sci., Victoria, BC, Canada)

Diel patterns in marine mammal activity can be difficult to assess visually due to the cost of ship time and limitations of daylight and weather during expeditions that can bias analysis based on sightings. Acoustic data can be a cost-effective approach to evaluate activity patterns in marine mammals. However, manual analysis of acoustic data is time consuming, and impractical for large data sets. This study seeks to evaluate diel patterns in Pacific white-sided dolphin communication through automated analysis of one year of continuous acoustic data collected from the Barkley Canyon node of Ocean Networks Canada's NEPTUNE observatory, offshore of Vancouver Island, British Columbia, Canada. In this study, marine mammal acoustic signals are manually annotated in a sub-set of the data, and used to train a random forest classifier targeting Pacific white-sided dolphin pulsed calls. Binary and multiclass classifiers are compared, and the effects of different data-balancing methods are evaluated. The results from automated classification of the full data set are used to determine whether a diel pattern in Pacific white-sided dolphin communication exists in this region.

5aABb5. Analysis of fin whale vocalizations south of Rhode Island. Jennifer Giard, James H. Miller, Gopu R. Potty (Ocean Eng. Dept., Univ. of Rhode Island, 215 S Ferry Rd., Narragansett, RI 02882, jennifer_giard@my.uri.edu), Arthur Newhall, Ying-Tsong Lin, and Mark F. Baumgartner (Woods Hole Oceanographic Inst., Woods Hole, MA)

Fin whale vocalizations were recorded south of Rhode Island during late summer through early fall of 2015 using a number of underwater recording systems. These systems were deployed to monitor broadband noise, including pile driving, from construction of the Block Island Wind Farm. Two vertical hydrophone array moorings were deployed in approximately 40 m of water each with four hydrophones. Additionally, a tetrahedral array was deployed in about 30 m of water just above the seabed. The tetrahedral array consisted of four hydrophones spaced 0.5 m apart. The spacing between each of these recording systems was approximately 7.5 km. 20-Hz fin whale vocalizations were recorded numerous times on all of the sensors both during and after construction was completed. An analysis and localization effort of these signals was performed to estimate the source level, directionality and the track of the whale over the period of the vocalizations. The results of this analysis will be discussed. [Work supported by the BOEM.]

5aABb6. LAMLA 2016: Results from the workshop listening for aquatic mammals in Latin America. Renata S. Sousa-Lima (Physiol. and Behavior, UFRN, Lab. of BioAcoust., Centro de Biociencias, Campus Universitario, Caixa Postal 1511, Natal, Rio Grande do Norte 59078-970, Brazil, sousalima.renata@gmail.com), Susannah Buchan (Universidad de Concepción, Santiago, Chile), Artur Andriolo (UFJF, Juiz de Fora, Minas Gerais, Brazil), and Julia R. Dombroski (Syracuse Univ., Syracuse, NY)

The field of bioacoustics and the applications of passive acoustic monitoring (PAM) methods to investigate aquatic mammals have grown worldwide. In Latin America, a number of researchers are using PAM to investigate different species and habitats. However, due to the lack of a proper venue to discuss research findings and priorities, collaboration within the region is scarce. Considering the clear demand for an opportunity for networking and exchange of information at a regional level, we proposed a series of workshops entitled LAMLA—Listening for Aquatic Mammals in Latin America. The aim of LAMLA is to bring together researchers, professionals, and graduate students working in bioacoustics to communicate their research, network and interact, and discuss directions for a coordinated regional bioacoustics network in order to better utilize research resources. The first edition of LAMLA was held in Natal, Brazil, in June, 2016 and the second edition was held in Valparaiso, Chile, during the XI SOLAMAC Reunion in November 2016. Outcomes and results of these two meetings will be presented as well as our goals and expectations for the 2018 meeting. [LAMLA was supported by PAEP CAPES, Office of Naval Research Global, Cetacean Society International, FAPERN, UFRN, UFJF, UNAM, University of Saint Andrews, Universidad de Concepcion, SOLAMAC, and the Acoustical Society of America.]

5aABb7. Concurrent passive and active acoustic observations of high-latitude shallow foraging sperm whales (*Physeter macrocephalus*) and mesopelagic prey layer. Geir Pedersen (Sci. and Technol., Christian Michelsen Res. AS, P.O. Box 6031, Bergen 5892, Norway, geir.pedersen@cmr.no), Espen Storheim (Nansen Environ. and Remote Sensing Ctr., Bergen, Norway), Lise D. Sivle, Olav Rune Godø (Marine Ecosystem Acoust., Inst. of Marine Res., Bergen, Norway), and Lars Alf Ødegaard (Norwegian Defence Res. Establishment (FFI), Bergen, Norway)

Using echosounder and hydrophone data from the Lofoten-Vesterålen Cabled Ocean Observatory (LoVe, N 68 °54.474', E 15 °23.145', 258 m depth) collected in 2015, we are able to concurrently quantify sperm whale (*Physeter macrocephalus)* shallow foraging behavior and the behavior of the mesopelagic prey layer. Click rate and type was detected by the passive acoustics while active acoustics monitored the distribution and vertical and horizontal movement of the prey organisms in the water column. In one instance a diving sperm whale was also detected by the active acoustics allowing TS measurements and estimation of diving speed and angle. Additional data such as ocean current and proximity of vessels, in addition to vessel noise measurements, further allowed us to examine potential links between oceanographic conditions and noise on sperm whale behavior and foraging and the presence of prey and whales. The results demonstrate the additional information obtained by combining data from active and passive acoustic sensors. The first part of the LoVe cross-disciplinary ocean observatory was established in 2013, and the extension is planned for 2017/2018 covering the Norwegian shelf to approximately 2500 m depth. This will further expand the observatories capabilities for underwater acoustic monitoring and targeted scientific studies.

5aABb8. Mining noise affects Rufous-Collared Sparrow (*Zonothichia capensis*) vocalizations. Yasmin Viana (Laboratório de Bioacústica, Museu de Ciências Naturais, Pontifícia Universidade Católica de Minas Gerais, Belo Horizonte, MG, Brazil), Robert J. Young (School of Environment and Life Sci., Univ. of Salford Manchester, Salford, United Kingdom), Renata S. Sousa-Lima (Physiol. and Behavior, UFRN, Lab. of BioAcoust., Centro de Biociencias, Campus Universitario, Caixa Postal 1511, Natal, Rio Grande do Norte 59078-970, Brazil, sousalima.renata@gmail.com), and Marina H. Duarte (Laboratório de Bioacústica, Museu de Ciências Naturais, Pontifícia Universidade Católica de Minas Gerais, Belo Horizonte, Brazil)

Mining activity generates noise through explosions, traffic, machinery, alert signals, etc. Noise affects the behavior of many species that depends on acoustic communication. Our objective was to verify if the noise produced by truck traffic affects rufous-collared sparrow vocalizations. Data were collected in an Atlantic forest fragment located close to a mine at the Peti Environmental Station, in Southeast Brazil. Two digital field recorders (SM2—Wildlife Acoustics) were installed 150m from each other and 25 m from a mining road. The SM2 were set to record at 44.1kHz, from 05:00 to 09:00 am during seven days in October 2012. Using Raven pro 1.4, maximum and minimum frequencies, number of notes and duration of the *Z. capensis* songs were extracted from the recordings one minute before, one after and during the passage of trucks. Trucks noise spectral measurements were also extracted. The species decreased the duration ($H = 17.8$, $gl = 2$, $p < 0.05$), the bandwidth ($H = 36.28$, $gl = 2$, $p < 0.05$) and the maximum frequency ($H = 24.45$, $gl = 2$, $p < 0.05$) and increased the minimum frequency of the calls ($H = 25.34$, $gl = 2$, $p < 0.05$) during exposure to truck noise. These results indicate that noise can affect the vocal behavior of the species and reveal the need to address the acoustic impact of mining on animal species.

5aABb9. Non-linear analysis as a hierarchal classification scheme for vocalizing marine biologics. Cameron A. Matthews (Panama City Div., Naval Surface Warfare Ctr., 110 vernon ave, Panama City, FL 32407, cameron.matthews@navy.mil), Anthony Matthews (EPS Corp, Panama City, FL), and Anthony Ceparano (Gumbo Limbo Res. Facility, Boca Raton, FL)

Many ocean animals use acoustic communications. Breeding, territorial aggression, and hunting actions often include some form, often complex, and can provide actionable intelligence on the vocalizing animals' intent. As a means of considering the linear features of the time series and the corresponding spectral and cyclic frequency content as it pertains to different classes of animals, a hierarchy termed periodic linear, periodic non-linear, and aperiodic non-linear structural vocalization is considered. To demonstrate the application of such a hierarchy, known vocalizations from the red hind grouper (Epinephelus guttatus), the four lined grunter (Pelates quadrilineatus), the spiny lobster (Panulirus argus) are considered and analyzed for grouping according to the hierarchy based on their time, spectral, and cyclic signal content. Finally, a vocalization collected from the invasive Lionfish (Pterois volitans), understood at the time of publication to be the first such recorded instance, is exercised against the hierarchy to show membership identification.

5aABb10. Acoustic environment of North Atlantic right whales in the Southeastern United States. Susan Parks (Dept. of Biology, Syracuse Univ., Syracuse, NY), Andrew J. Read, and Douglas P. Nowacek (Nicholas School of the Environment and Duke Univ. Marine Lab., Pratt School of Eng., Duke Univ., Beaufort, NC, doug.nowacek@duke.edu)

North Atlantic right whales are an endangered species of baleen whale that migrates along the east coast of the United States, with winter calving grounds located in the coastal waters off Florida and Georgia. This study investigated the acoustic environment experienced by individual right whales swimming through this habitat though the use of suction cup attached acoustic recording tags. Nineteen tag attachments were made between 2014 and 2016. These tags documented a range of sounds from the right whale acoustic environment, including calls produced by the tagged whale, sounds produced by conspecifics, as well as sounds from other biological (fish and dolphin) and anthropogenic sources. The call rates of individual whales were relatively low, with calls typically produced in short duration bouts. Sounds from other biological sources, particularly fish and dolphin, and anthropogenic sources, particularly vessels, were common. This project presents an initial step toward characterizing the acoustic environment experienced by individual whales to allow future comparisons to stationary acoustic recordings in the same habitat.

5aABb11. Redefining species boundaries for acoustically and morphologically distinct species of swamp breeding frilled tree frogs (*Kurixalus appendiculatus*) in the Southwestern Philippines. Taylor Broadhead (Forestry and Natural Resources, Purdue Univ., 203 S. Martin Jischke Dr., B066, West Lafayette, IN 47907, taylorbroadhead@gmail.com), Jesse Grismer, and Rafe Brown (Ecology and Evolutionary Biology, Univ. of Kansas, Lawrence, KS)

Combining analysis of male advertisement calls, multivariate analysis of continuous morphological variation, biogeographic information, and a multilocus phylogenetic estimate of relationships, we reconsider species boundaries within Philippine populations of the frilled tree frogs *Kurixalus appendiculatus*. Within the archipelago, the species spans several recognized biogeographic boundaries, with highly divergent genetic lineages isolated within formally recognized, geologically defined, faunal subregions. Given this distribution, at least four possible taxonomic arrangements are possible, varying from one to four possible evolutionary species. Simultaneous consideration of fixed external phenotypic character differences, continuously varying morphometric data, evolutionary relationships, biogeography, and statistically significant differences in mating calls converges on a solution of two Philippine species. We advocate for more widespread, regular, and deliberate sampling of acoustic data to diminish challenges for future studies, where we anticipate the validation of other likely taxonomic arrangements by differences in advertisement calls.

5aABb12. Brazilian *Cerrado* nocturnal summer soundscape. Luane S. Ferreira (Physiol., Universidade Federal do Rio Grande do Norte, Avenida Senador Salgado Filho 3000 - Campus Universitário, Natal, Rio Grande do Norte 59078-970, Brazil, fsluane@gmail.com), Eliziane G. Oliveira (Ecology, Universidade Federal do Rio Grande do Norte, Natal, Rio Grande do Norte, Brazil), Luciana H. Rocha (Physiol., Universidade Federal do Rio Grande do Norte, Natal, Rio Grande do Norte, Brazil), Flávio H. Rodrigues (General Biology, Universidade Federal de Minas Gerais, Belo Horizonte, Minas Gerais, Brazil), and Renata S. Sousa-Lima (Physiol., Universidade Federal do Rio Grande do Norte, Natal, Rio Grande do Norte, Brazil)

The Brazilian *Cerrado* is one of the world's biodiversity hotspots. Our objective was to characterize its nocturnal soundscape. 12 autonomous recorders (SongMeter2+, Wildlife Acoustics) were deployed in Canastra National Park (MG/Brazil) and recorded five consecutive nights during the rainy season. Using Arbimon II soundscape builder we identified four frequency bands with higher activity levels. The lower band (0.3-1.3 kHz) is acoustically occupied throughout the night. The second band (2.8-3.2 kHz) is highly active around sunset and almost disappears after 10 PM. The third band (3.8-6.6 kHz) splits into two near 9 PM, with the upper limit disappearing after 3 AM. The highest frequency band (9-16 kHz) is the broadest and occupied in all recordings, being comprised by unidentified background noise. Insects (mainly crickets and cicadas) are present in the three superior

bands, anura in the two lower bands, and birds in the second and third near dusk and dawn. Characterizing such protected soundscapes is vital for future monitoring and identification of changes in this important Brazilian preserve.

5aABb13. Passive acoustic monitoring finds concurrent use of an artificial reef in the New York Bight by foraging humans and odontocetes (*Tursiops truncatus*). Colin Wirth and Joseph Warren (Marine and Atmospheric Sci., Stony Brook Univ., 239 Montauk Hwy., Southampton, NY 11968, colin.wirth@gmail.com)

Passive acoustic recordings collected during summer 2015 at an artificial reef (sunken barge) south of Long Island, New York revealed regular visitation by groups of delphinid odontocetes. Detected signals included social (whistles) and foraging (short-interval echolocation) signals of odontocetes, as well as signals specific to bottlenose dolphins (*Tursiops truncatus*) and two known prey species. Visual observations, high broadband noise levels, and presence of acoustic signatures specific to boats indicated heavy use of this site by recreational fishers. Boat detections were significantly more frequent on weekends and between sunrise and sunset. Dolphin detections did not vary diurnally and were significantly lower on weekends, possibly due to avoidance of persistent noise disturbance. Bottlenose dolphins produce low-frequency, narrow-band signals that are highly susceptible to masking by boat noise. However, no significant difference was observed in the duration, average peak frequency, or frequency range of these signals when boat noise was present or absent. Our findings demonstrate the benefits of passive acoustic techniques in monitoring soniforous users (including humans) of artificial reef habitats in these waters. Attraction of both human fishers and odontocetes to artificial reefs may increase direct human-predator interactions as well as indirect ecological competition.

5aABb14. Effects of noise on avian abundance and productivity at the landscape scale. Stacy L. DeRuiter (Mathematics and Statistics, Calvin College, Grand Rapids, MI 49546, sld33@calvin.edu), Amber Bingle (Biology, Calvin College, Grand Rapids, MI), Matthew Link (Mathematics and Statistics, Calvin College, Grand Rapids, MI), Michael Pontius, and Darren Proppe (Biology, Calvin College, Grand Rapids, MI)

Songbirds, with their reliance on acoustic communication, may be especially sensitive to potential population consequences of anthropogenic noise. Regional studies suggest that many species avoid noisy areas, and in some cases non-avoiding individuals experience reduced fitness. However, multiple studies of a species sometimes produce conflicting results, perhaps due to localized processes that overpower or exacerbate noise effects. Many studies also use abundance as an imperfect proxy for population persistence. To address these issues, we paired large published datasets—from the MAPs (Monitoring Avian Productivity and Survivorship) program and the U.S. National Parks Service noise map—to assess noise levels and bird demographics across the continental United States. We modeled effects of noise on songbird diversity, abundance, productivity, and physical condition (fat score), accounting for temporal and spatial variation. At the continental scale, diversity decreased with increasing noise, but other effects varied by species. For example, least flycatchers become more abundant with increasing noise, and red-breasted nuthatches less abundant. Effects on fat and productivity also varied by species, with abundance trends not consistently matching productivity. Landscape-scale models such as those presented here may facilitate range-wide conservation measures and help identify species or groups that are most at risk.

5aABb15. Preliminary measurements of passive acoustics in Lake Superior. Jay A. Austin (Large Lakes Observatory, Univ. of Minnesota, Res. Lab Bldg., Duluth, MN 55812, jaustin@d.umn.edu)

In July 2016, a hydrophone was deployed for eight days in 54 m of water in the western arm of Lake Superior. This is, to the best of our knowledge, the first recording of passive acoustic information in a large lake. The signal is dominated by noise from passing ships (30-100 Hz) and by surface winds (broad spectrum). Noise from passing ships drops off as approximately r^{-6}, suggesting that there are significant transmission losses associated with reflections off of the bottom. The signal associated with wind is highly correlated with wind speed as measured at a nearby buoy. Intermittent "clicks" look similar to burbot calls previously observed, and appear to occur only in the absence of ship noise. This suggests a potential behavioral response to ambient acoustic energy. Ray tracing experiments suggest the acoustic environment within the lake will change drastically as the lake transitions from summer stratified conditions to unstratified, and again when inverse winter stratification sets in.

5aABb16. Correlation of direct field observations of the endangered Golden Cheeked Warbler (*Dendroica chrysoparia*) with long-term acoustical measurements. Preston S. Wilson (Mech. Eng., Univ. of Texas at Austin, 1 University Station, C2200, Austin, TX 78712-0292, pswilson@mail.utexas.edu), David P. Knobles (Knobles Sci. and Anal., LLC, Austin, TX), Lisa O'Donnell, Darrell Hutchinson, and William Reiner (Balcones Canyonlands Preserve, City of Austin, Austin, TX)

We present elements of a bioacoustics study that correlated direct field observations of the endangered Golden Cheeked Warbler (*Dendroica chrysoparia*) in the Balcones Canyonlands Preserve near Austin, TX with co-located long-term acoustical measurements. The goal is to eventually understand the effects of anthropogenic noise on the breeding success of the warblers. The anthropogenic component of the soundscape includes noise from road traffic, jet aircraft, helicopters, and urban development and utilization. During the 2016 and 2017 breeding seasons (March through May), acoustical recordings were made from sun up to sun down each day at four sites. The acoustical measurements were correlated with contemporaneous direct field observations that mapped the male territories and their degree of reproductive success. The study considered the interplay of the source levels of the warblers, the noise, and sound propagation loss in the habitat. The key result is the difference in the distribution of two song types sung by banded birds with different degrees of breeding success. The relationship between these distributions, the corresponding success rates, and the size of the acoustic active space as determined by soundscape characteristics, is discussed.

5aABb17. Spatial analysis of soundscapes of a Paleotropical rainforest. Jack T. VanSchaik (Forestry and Natural Resources, Purdue Univ., 203 S. Martin Jishke Dr., Mann Hall, B066, West Lafayette, IN 47906, jvanscha@purdue.edu), Amandine Gasc (Forestry and Natural Resources, Purdue Univ., Paris, France), Kristen M. Bellisario, and Bryan C. Pijanowski (Forestry and Natural Resources, Purdue Univ., West Lafayette, IN)

The world's biodiversity is drastically decreasing due to human activity. The paleotropical rainforests of Borneo contribute 10% of the world biodiversity but are at risk of destruction due to logging and other human interests. Soundscape Ecology, defined as the composition of sounds in an environment, is a new field that offers potential for biodiversity assessment. Spatial dynamics are an important component of an ecosystem, yet the link between spatial dynamics and soundscapes has not yet been studied. It should be possible to assess disturbance of an ecosystem by analyzing the spatial structure of the soundscape. Particularly, soundscapes in healthy ecosystems should exhibit more spatial autocorrelation than soundscapes in disturbed ecosystems. We calculated Alpha acoustic and Beta acoustic indices for 13 recorders at each site that had identical spatial configurations. We compared the resultant Alpha indices using Moran's I, Geary's C and other statistical tests. We compared Beta indices by Mantel Tests and a new technique, the Beta index Semivariogram, a traditional variogram except using means of Beta indices. Spatial statistics on Alpha and Beta indices, and Beta Index Semivariograms reveal more spatial autocorrelation at the undisturbed site. However, Beta Indices detect disturbance better presumably due to their comparative nature.

5aABb18. Musicological indices for soundscape ecological analysis. Kristen M. Bellisario, Jack T. VanSchaik (Forestry & Natural Resources, Purdue Univ., Lafayette, 195 Marsteller St., 305 FORS Bldg., West Lafayette, IN 47906), amandine gasc (Forestry & Natural Resources, Purdue Univ., Lafayette, Paris, France), Carol Bedoya, Hichem Omrani, and Bryan C. Pijanowski (Forestry & Natural Resources, Purdue Univ., Lafayette, West Lafayette, IN 47906, bpijanow@purdue.edu)

Soundscape ecologists have collected sound recordings from large-scale studies that are difficult to analyze with traditional approaches and tools. Natural soundscapes are complex and contain a diverse mixture of biological, geophysical, and anthropogenic sources that span similar frequency bands and often lack a discernible fundamental frequency. Selecting features that are responsive to signals without fundamental frequencies and that are capable of classification for multi-layer signals, or polyphonic textures, is a challenging task in soundscape ecology. Spectral timbral features in various combinations have been shown to discriminate in music classification problems, and lend support to our hypothesis; timbral features in soundscape analysis may detect and identify patterns that are inherently related to order-specific communication in frequency bands shared by biological, geophysical, and anthropogenic sounds.Combined timbral feature extractions provides a new level of information about acoustic activity within a soundscape. Current soundscape metrics assess biodiversity, functional diversity, and acoustic complexity, but may be missing crucial information if we compare musical acoustic analysis techniques used to identify genres and structures in music. This new method provides a relational approach to understanding sound event interactions within soundscapes, refining quantifiable soundscape data, and improving the resolution with which it is analyzed.

5aABb19. Acoustic competition of Serranids at a fish spawning aggregation. Katherine Cameron, Brice Semmens (Marine Physical Lab., Scripps Inst. of Oceanogr., 8622 Kennel Way, La Jolla, CA 92037, kccameron@ucsd.edu), Christy V. Pattengill-Semmens (REEF, La Jolla, CA), Steve Gittings (National Marine Sanctuaries Program, NOAA, Silver Spring, MD), Croy McCoy (Cayman Island Dept. of Environment, Little Cayman, Cayman Islands), and Ana Sõirovic¨ (Marine Physical Lab., Scripps Inst. of Oceanogr., La Jolla, CA)

Many fish species are known to produce stereotyped calls during spawning activities. It has been hypothesized that these calls play a vital role in coordination during this critical period. Competition for acoustic space can result in masking of calls and, as a result, may limit their function. Acoustic niche separation could be a solution to avoid acoustic competition for species that co-occur in a geographic area. The data from passive acoustic arrays deployed in Little Cayman, Cayman Islands, between 2015 and 2017 during the spawning of one of the largest aggregations of Nassau grouper (*Epinephelus striatus)* have been analyzed to explore acoustic niches in Serranids. At least three other vocal Serranid species are known to spawn at this location during the same time as Nassau grouper: red hind (*E. guttatus*), black grouper (*Mycteroperca bonaci*), and yellowfin grouper (*M. venenosa*). Call characteristics, such as the peak frequency, bandwidth, source level, and pulse period, were analyzed for all distinct calls from these species. In addition, temporal and spatial patterns of those calls were evaluated, allowing a detailed understanding of the communication space and acoustic competition potential for these cohabitating species.

5aABb20. Sparsified nightly fish chorusing in the Dry Tortugas during elevated background noise levels due to a tropical storm. Benjamin S. Gottesman, Dante Francomano, Taylor Broadhead, and Bryan C. Pijanowski (Forestry and Natural Resources, Purdue Univ., Ctr. for Global Soundcapes, 331 Smiley St., West Lafayette, IN 47906, bgottesm@purdue.edu)

Determining how fish respond to naturally occurring noise disturbances can provide insight into the biological mechanisms underlying the response of fish to anthropogenic noise. Data collected from passive acoustic monitoring in the Dry Tortugas, FL, showed that a tropical storm significantly increased levels of low-frequency noise over a four-day period. The nightly fish chorus occurring at this site, likely from Black Drums (*Pogonias cromis*), was significantly reduced during this storm event, with fewer than 10% of grunts detected during the storm's peak as compared to pre- and post-storm levels. We applied Soundscape Control Charts, a new method to quantify the effects of natural and anthropogenic disturbances. Most commonly used in the industrial sector as alarm systems, control charts identify when a system deviates from its normal state. In this study, Soundscape Control Charts quantified the effects of this tropical storm on the communication of fish. This serendipitous dataset suggests that fish communication is negatively impacted by naturally occurring noise disturbances, and is evidence for the need to preserve marine acoustic habitats in order to facilitate animal communication.

5aABb21. Bioacoustic analysis of penguin vocalization classification for Newport Aquarium educational exhibit. Bethany Wysocki (Univ. of Cincinnati, 3250 Jefferson Ave., Cincinnati, OH 45220, wysockby@mail.uc.edu) and Peter M. Scheifele (FETCH~LAB, Univ. of Cincinnati, Cincinnati, OH)

Newport Aquarium in Newport, Kentucky, strives to engage its visitors in the educational importance of aquatic life and conservation. Through a variety of research programs and volunteerism, the aquarium promotes local and global efforts for animal advocacy. An important component of their organization is the diversity of education offered to its daily patrons and guests. The purpose of this project was to examine the acoustic and behavioral characteristics of penguin vocalizations in an effort to create an educational exhibit targeting school-age visitors. Vocalization samples of the 5 penguin species housed at Newport Aquariums' Kroger Penguin Palooza exhibit were recorded at various times of the day, over a 5-month period. In addition to the recordings of the various penguins, behavioral characteristics were also noted in correspondence with the individual call and species. The calls were spectrally analyzed using SpectraPLUS software system, looking specifically at frequency, power, and sound pressure levels. Spectrogram plots were also documented to identify penguin vocalization variances and species identification. A Hidden Markov Model used this information to categorize and cluster vocalizations in an effort to classify them. The information extracted about the vocalizations and acoustic variances will be used for educational and exhibit purposes for the aquarium.

5aABb22. Spectrogram contour analysis of vocalizations of the Asian Small-Clawed Otter (Amblonyx cinerea). Haylea G. McCoy and Peter M. Scheifele (Commun. Sci. and Disord., Univ. of Cincinnati, 3202 Eden Ave., Cincinnati, OH 45267, roarkhg@mail.uc.edu)

Asian Small-Clawed Otters (Amblonyx cinerea) are small creatures found in many areas of southern and southeastern Asia, as well as Indonesia. They mate for life and communicate with one another using a wide variety of vocalizations from long, drawn out cries, to small yipping noises. The vocalizations of these otters were recorded in the back up area of the Newport Aquarium and Wellington Zoo and analyzed in Spectra Plus to determine the spectrogram contour of each vocalization. The goal of this research was to gather and compile data on the way the Asian Small-Clawed Otter communicates in order to better understand the way these animals live. These data will now be used to perform vocal clustering and classification using a Hidden Markov model, spectral moments, and geometric contour classification (Lofft, 2009; Williamson, 2014).

5aABb23. North Atlantic Right Whale call detection with very deep convolutional neural networks. Kele Xu (College of Electron. Sci. and Eng., National Univ. of Defense Technol., 16 Rue Flatters, Paris 75005, France, kelele.xu@gmail.com), Hengxing Cai (Guangdong Provincial Key Lab. of Intelligent Transportation Systems, Sun Yat-Sen Univ., Guangdong, China), Xi Liu (ESPCI Paris, Paris, France), Zhifeng Gao (School of Software & Microelectronics, Peking Univ., Peking, China), and Bingbing Zhang (College of Electron. Sci. and Eng., National Univ. of Defense Technol., Changsha, China)

Ship collision is one of the main threats to the North Atlantic right whale, which is in danger of extinction. One popular way to reduce the collision is monitoring for the occurrences of whales by detecting their sounds on data recordings. We explore the application of very deep convolutional neural network in this detection problem. For feature extraction, we compute Mel-frequency cepstral coefficients (MFCCs) along with their first and second temporal derivatives, and Fourier-transform-based filter-banks for all sound clips. MFCCs were calculated with Hamming window, and the filter-banks were calculated in range of 50—650 Hz, and include 72 coefficients, distributed on mel-scale, for each of the 97 time steps. For classifier modeling method, we apply the very deep convolutional Neural Network (CNN) in our task. The CNN architecture s 22 layers, which consists of alternating convolutional layer and pooling layer, while the last layers are full-connected neural network. Dropout is used in our fully connected layers with a rate of 0.4. By using the data provided by the Cornell University Whale Detection data, our model provides area under the ROC curve (AUC) performance of 0.985, which achieves the state-of-the-art performance presently.

5aABb24. Characterizing Chilean blue whale vocalizations with digital acoustic recording tags: A test of using tag accelerometers for caller identification. Mark Saddler (Biology, Woods Hole Oceanographic Inst., 5500 S University Ave. 585, Chicago, IL 60637, mark.saddler@sbcglobal.net), Alessandro Bocconcelli (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., Woods Hole, MA), Leigh S. Hickmott (Sea Mammal Res. Unit, Scottish Oceans Inst., Univ. of St. Andrews, Petersfield, Hampshire, United Kingdom), Gustavo Chiang, Rafaela Landea-Briones, Paulina A. Bahamonde, Gloria Howes (Fundación MERI, Vitacura, Santiago, Chile), and Laela Sayigh (Biology, Woods Hole Oceanographic Inst., Woods Hole, MA)

Vocal behavior of blue whales (*Balaenoptera musculus*) in the Gulf of Corcovado, Chile, was analyzed using digital acoustic recording tags (DTAGs). We report the occurrence of Southeast Pacific type 2 (SEP2) calls, which exhibit peak frequencies, durations, and timing consistent with previous reports. We also offer the first description of tonal downswept (D) calls for this population. Since being able to accurately assign vocalizations to individual whales is fundamental for studying communication and for estimating population densities from call rates, we further examine the feasibility of using DTAG accelerometers to identify low-frequency calls produced by tagged whales. We cross-correlated acoustic signals with simultaneous tri-axial accelerometer readings in order to analyze the phase match as well as the amplitude of accelerometer signals associated with low-frequency calls, which provides a reliable method of determining if a call is associated with a detectable acceleration. Our results suggest that vocalizations from nearby individuals are also capable of registering accelerations in the tagged whale's DTAG record. We cross-correlate acceleration vectors between calls to explore the possibility of using signature acceleration patterns associated with sounds produced within the tagged whale as a new method of identifying which accelerometer-detectable calls originate from the tagged animal.

5aABb25. Build-up effect of auditory streaming in budgerigars (*Melopsittacus undulatus*). Huaizhen Cai, Laurel A. Screven, and Micheal L. Dent (Psych., Univ. at Buffalo, SUNY, 206 Park Hall, Buffalo, NY 14228, huaizhen@buffalo.edu)

When listening to a rapid tone sequence of the form ABA-ABA-ABA-… (where A and B are two tones of different frequencies and "-" indicates a silence interval), listeners may either hear one coherent "gallop" of three tones grouped together or two separate auditory streams (one high frequency, one low frequency) appearing to come from two sound sources. Research on humans indicates that the tendency to perceive two streams can be built up as exposure time increases. Neural recordings in European starlings show build-up effects of auditory streaming too. A lack of behavioral data on the build-up effect in nonhumans make it difficult to draw parallels between animals and humans. The present research aims to behaviorally validate the build-up effect of auditory streaming and factors that may influence the effect in nonhuman animals. Four budgerigars were tested in a categorization task using operant conditioning. "Streaming" categorization increased as the frequency separation increased. Additionally, in some frequency separations, as the sequence duration increases, the probability of "streaming" categorizations reported by the budgerigars is higher. These results indicate that budgerigars experience a build-up effect of auditory streaming behaviorally, and this effect is influenced by, but may not be limited to, different frequency separations.

5aABb26. Acoustic characterization of sound production by the Pot-Bellied Seahorse (*Hippocampus abdominalis*). Brittany A. Hutton, Peter M. Scheifele (Commun. Sci. and Disord., Univ. of Cincinnati, 3202 Eden Ave., P.O. Box 670379, Cincinnati, OH 45267, huttonba@mail.uc.edu), and Laurel Johnson (Animal Husbandry, Newport Aquarium, Newport, KY)

Sound production is a critical component of predator-prey interactions. In order to understand why seahorses produce sound in various instances (i.e., courtship, feeding, and stress), we must first quantify the acoustic parameters of the signal. Seahorses produce sound with a stridulation of the supraoccipital bone and the coronet by moving their head upwards in a motion that is referred to as a "snick." The acoustic signal that accompanies this head movement is called a "click." We set out to analyze the sound parameters of the largest seahorse species, *Hippocampus abdominalis,* housed at the Newport Aquarium. Adult and juveniles were tested each individually and allowed one hour to acclimate to an isolated tank. Feeding on brine shrimp (Genus: *Artemia*) was observed with video and audio recordings that were collected for approximately 12 minutes. SpectraPLUS was used to evaluate the frequency, intensity, and number of clicks present in the audio recordings. The video footage allowed for analysis of the presence of the snick. By characterizing the sound production in this species of seahorse we are able to begin to answer the question of the purpose for the click and snick behavior.

5aABb27. Biotic factors influencing sound production patterns by two species of snapping shrimp (*Alpheus heterochaelis* and *A. angulosus*). Jessica N. Perelman (Biology, Woods Hole Oceanographic Inst., 266 Woods Hole Rd., MRF, WHOI, Woods Hole, MA 02543, jperelman@whoi.edu), Apryle Panyi (Univ. of Southern MS, Ocean Springs, MS), Ashlee Lillis, and T. Aran Mooney (Biology, Woods Hole Oceanographic Inst., Woods Hole, MA)

Snapping shrimp are among the most pervasive sources of biological sound in the ocean, inhabiting and sonifying many shallow temperate and tropical reefs and seagrass flats. Despite the continuous crackling sounds of snapping shrimp colonies that contribute greatly to marine soundscapes, relatively few studies have explored baseline information regarding acoustic patterns and underlying behavioral ecology. Recent field data has highlighted intricate spatiotemporal dynamics in shrimp sound production, with seasonally variable diel rhythms. However, the biotic factors (e.g., sex, size, species, or behavioral mode) underlying these patterns are unclear. This study investigated the snapping behavior of two species of snapping shrimp (*Alpheus heterochaelis* and *A. angulosus*). Previously undescribed spontaneous snap behavior was observed. The snap rates for shrimp held individually, in pairs, and in a 10-shrimp colony, were measured under natural day-night light conditions. Results show high variability in individual snap rates, with females generally snapping more than males, and higher snap rates for same-sex pairs compared to male-female pairings. Time of day was also found to variably affect snap rates. Establishing the nature of these patterns increases our understanding of a key sound-producer and driver of marine soundscapes.

5aABb28. Comparison between the marine soundscape of recreational boat mooring areas with that of a pristine area in the Mediterranean: Evidence that such acoustic hot-spots are detrimental to ecologically sensitive habitats. Juan Francisco Ruiz, Jose Miguel Gonzalez-Correa, Just Bayle-Sempere, Jaime Ramis (Univ. Alicante, Alicante, Spain), Rodney A. Rountree (Waquoit, MA), and Francis Juanes (Univ. Victoria, 3800 Finnerty Rd., Victoria, BC V8P 5C2, Canada, juanes@uvic.ca)

We investigated the potential to use passive acoustics to access the impact of recreational boat mooring areas on ecologically sensitive habitats in the Western Mediterranean. One important consequence of the tourist industry in the region is that it targets the most pristine and ecologically sensitive habitats. Underwater sounds were recorded in mooring areas in Ibiza, Formentera and Tabarca harbors during high use and low use seasons and compared to recordings in the Tabarca Marine Protected area. At each location, we recorded sounds during 20 min at three different sites, for three random sampling times during the day. The percent of time occupied by selected biological (drums and croaks) and anthropogenic sounds (boat and mooring chain noises), and call rates of selected fish sounds were measured and compared among sites and seasons. Biological sounds contributed significantly less to the soundscape in mooring areas during the tourist season, and to the reserve in both seasons. Our study demonstrates the critical need for research on the impact of acoustic noise "hot spots" such as recreational mooring areas on marine and freshwater soundscapes.

5aABb29. Auditory sensitivity of various areas of the head to underwater acoustical stimulation in odontocetes. Evgeniya Sysueva (Inst. of Ecology and Evolution, 33 Leninsky Prospect, Moscow 119071, Russian Federation, evgeniasysueva@gmail.com), Paul E. Nachtigall (Hawaii Inst. of Marine Biology, Univ. of Hawaii, Kailua, HI), Aude F. Pacini (Hawaii Inst. of Marine Biology, Univ. of Hawaii, Kaneohe, HI), Jeff Pawloski, Craig Allum (Sea Life Park, Honolulu, HI), and Alexander Supin (Inst. of Ecology and Evolution, Moscow, Russian Federation)

Sensitivity to the local underwater acoustic stimulation of the ventro-lateral head surface was investigated in a bottlenose dolphin (Tursiops truncatus). The stimuli were tone pip trains of carrier frequencies ranging from 32 to 128 kHz with a pip rate of 1 kHz. Auditory evoked potentials (the rate following responses) were recorded. For all the tested frequencies, a low-threshold region was revealed at the lateral side of the middle portion of the lower jaw. This result differed from that obtained in a beluga whale, Delphinapterus leucas (Popov *et al*., JASA 2016, 140: 2018) revealed a maximal sensitivity region next to the medial side of the middle portion of the lower jaw. The comparative analysis of these data and their extrapolation to all odontocetes in generally is discussed.

5aABb30. Influence of fatiguing noise on auditory evoked responses to stimuli of various levels in a beluga whale, *Delphinapterus leucas*. Vladimir Popov, Evgeniya Sysueva, Dmitry Nechaev, and Alexander Supin (Inst. of Ecology and Evolution, 33 Leninskij prosp., Moscow, 119071, Russian Federation, popov.vl.vl@gmail.com)

The post-exposure effect of fatiguing noise (half-octave band-limited noise centered at 32 kHz) on the evoked responses to test stimuli (rhythmic pip trains with a 45-kHz center frequency) at various levels (from threshold to 60 dB above threshold) was investigated in a beluga whale *Delphinapterus leucas*. For baseline (pre-exposure) responses, the magnitude-vs-level function featured a segment of steep magnitude dependence on level that was followed by a segment of little dependence (plateau). Post-exposure, the function shifted upward along the level scale. Due to the plateau in the magnitude-vs-level function, post-exposure suppression of responses depended on the stimulus level such that higher levels corresponded to less suppression. The experimental data may be modeled based on the compressive non-linearity of the cochlea. According to the model, post-exposure responses of the cochlea to high-level stimuli are minimally suppressed compared to the pre-exposure responses, despite a substantially increased threshold. [The study was supported by The Russian Foundation for Basic Research (Grant No. 15-04-01068).]

5aABb31. Psychophysical audiogram of a California sea lion (*Zalophus californianus*) listening for airborne tonal sounds in an acoustic chamber. Colleen Reichmuth, Jillian Sills, and Asila Ghoul (Inst. of Marine Sci., Long Marine Lab., Univ. of California, Long Marine Lab., 115 McAllister Way, Santa Cruz, CA 95060, coll@ucsc.edu)

Many species-typical audiograms for marine mammals are based on data from only one or a few individuals that are not always tested under ideal conditions. Here, we report auditory thresholds across the frequency range of hearing for a healthy, five-year-old female California sea lion identified as *Ronan*. *Ronan* was trained to enter a hemi-anechoic acoustic chamber to perform a go/no-go audiometric experiment. Auditory sensitivity was measured first by an adaptive staircase procedure and then by the method of constant stimuli. Minimum audible field measurements were obtained for 500 ms frequency-modulated tonal upsweeps with 10% bandwidth and 5% rise and fall times. Thresholds were measured at 13 frequencies: in one-octave frequency steps from 0.1 to 25.6 kHz, and additionally at 18.0, 22.0, 36.2, and 40.0 kHz. Sensitivity was greatest between ~0.9 and 23 kHz, with best hearing of 0 dB re 20 μPa at 12.8 kHz. Hearing range, determined at the 60 dB re 20 μPa level, extended from ~0.2 kHz to 38 kHz. Sensitivity was comparable to that of three sea lions tested in similarly controlled conditions, and much better than that of two sea lions tested in less controlled conditions. [Work supported by ONR.]

5aABb32. Differential temporal structures in mouse ultrasonic vocalizations relay sex and age information about the producer. Daniel Calbick (Neurosci., Yale Univ., 333 Cedar St., New Haven, CT 06510, daniel.calbick@yale.edu), Gregg A. Castellucci (Linguist, Yale Univ., New Haven, CT), and David A. McCormick (Neurosci., Yale Univ., New Haven, CT)

Mice produce a variety of ultrasonic vocalizations (USVs) during social interactions among conspecifics. However, it remains unclear which features of these calls mice utilize to distinguish one type of USV from another. In this study, we examine male courtship USVs, neonatal isolation USVs, and female social contact USVs, and find that the temporal structure of the calls alone is sufficient for a high level of discriminability. Specifically, we found that males produce temporally distinct short and long duration USVs with resulting short and long duration call intervals, while females produce nearly exclusively short USVs with short call durations. Young pups were found to produce medium duration USVs with long call intervals only. Interestingly, as the pups aged, their USV durations and call intervals decreased and approached values observed in juvenile male courtship USVs. Therefore, the gross USV rhythmic structure carries a high degree of information about both sex and age of the producer, and may be utilized by mice during call discrimination. These findings are reminiscent to some pinniped species, who face similar challenges in the transmission of their calls (i.e., a high density of calling conspecifics), who also use gross temporal features to distinguish call types.

5aABb33. Comparison between sub-ms fast spatio-temporal imaging and electrical recordings from the bat inferior colliculus using a micro-endoscope. Hidetaka Yashiro (Life and Medical Sci., Doshisha Univ., Kyotanabe, Kyoto 6180011, Japan, hidetaka.yashiro@gmail.com), Kazuo Funabiki (Inst. of Biomedical Res. and Innovation, Kobe, Japan), Andrea Simmons, James A. Simmons (Brown Univ., Providence, RI), and Hiroshi Riquimaroux (Shandong Univ., Jinan, Shandong, China)

Recently, we demonstrated that our micro-endoscopic system enabled acquiring optical image in submillisecond temporal resolution (132 μs/line) by line-scan. The micro-endoscope was fabricated from optical fiber bundle as an endoscope tip and the bundle was coated by gold and enamel. The tip was beveled as cone-shape for minimal invasion and the surrounding edge became electrical-conductive for being used as an electrode. Then, this system can record electrical activities (local field potentials; LFP, and multi-unit activities; MUA) at a sampling rate of 20 kHz and optical responses (calcium response derived from Oregon green BAPTA-1, AM; ΔF/F) simultaneously. We recorded these responses from the inferior colliculus (IC) of two species of bats (*Carollia perspicillata* and *Eptesicus fuscus*) to tone bursts (10, 20, 40, 60, and 80 kHz), noise bursts (5-100 kHz), and downward FM sweeps (80-20 and 100-30 kHz). Along the same scanning line, several activated areas (hot spots) were found responding to a single sound stimulus, while different areas were also found to be activated dependent on types of sound stimulus. We analyzed spatio-temporal characteristics at a single recording site and compared calcium response to electrical activities (LFP and MUA). [Work supported by JSPS, MEXT Japan, ONR, Capita Foundation.]

5aABb34. Neurobehavioral studies of *Dumetella carolinensis* in the Northeast United States. Babak Badiey (Newark Charter School, 200 Mcintire Dr., Newark, DE 19711, babak.badiey@gmail.com)

Behavioral studies of Grey Catbirds (*Dumetella carolinensis*) have been conducted over two seasons using data collection, including playback calls, to learn about the neurobehavior of mimicking birds. First birds are identified by their signals, and detailed signal processing techniques are used to examine the characteristics of the bird calls using call time-frequency signal dispersion and relating them to behavior. Catbird songs often include frequency-modulated notes sweeping through a wide range of frequencies and different sweeps related to their syrinx and are identified in the signals. The broadband signals are grouped into similar song patterns and are checked versus the bird's position, the background noise, and weather conditions. These experiments are conducted in controlled background noise conditions to quantify the effect of noise on the birds. It is found that background noise causes production of louder calls while it does not affect that playback behavioral response of the bird.

5aABb35. Computational acoustics for soundscape ecology: Simulation and normalization. Cris Graupe (Multidisciplinary Eng., Purdue Univ., 272 Littleton St., Unit 561, West Lafayette, IN 47906, cgraupe@purdue.edu)

One of the major research themes of soundscape ecology is to understand ecosystem dynamics by measuring and analyzing patterns in biological acoustic communication in the context of the local environment. The emerging field focuses at landscape scales to quantify compositional, spatial, and temporal variation in these patterns. This is reflected by variation in specific acoustic qualities that must be taken into account to more accurately compare diverse soundscape recordings. A new software is being developed that aims to overcome this challenge. The MATLAB-based software utilizes geometrical and parabolic methods of computational acoustics to simulate terrestrial sound propagation, quantifying the impact of variable environmental filtration on acoustic recordings. This software provides both the ability to quantitatively compare this impact across ecosystems, and the ability to normalize acoustic recordings so that soundscape data from distinct environments can be compared and analyzed more accurately.

5aABb36. Bioacoustic monitoring station in underwater sculpture. Heather R. Spence (GRACIASS, 801 S. 25th St., Arlington, VA 22202, info@heatherspence.net)

In the waters surrounding Cancun, pressures from development, tourism, and shipping threaten the second largest coral reef system in the world. Passive bioacoustic monitoring provides information about presence and activity of marine life and other environmental information, including during the night and inclement weather. The development and deployment of underwater reef-forming art structures provided a unique opportunity to listen to the stages of growth. Starting in 2012, an Ecological Acoustic Recorder was used to sample sounds at "The Listener" sculpture site in the Punta Nizuc Marine Protected Area in Quintana Roo, Mexico. Snapping shrimp and fish sounds were prevalent as well as sounds from boat motors. Diel patterns were characterized to inform management practices and assess efficacy of nighttime noise restrictions.

5aABb37. Non-whistle sounds used in bottlenose dolphin aggressive interactions recorded on digital acoustic tags. Laela Sayigh, Austin Dziki (Biology Dept., Woods Hole Oceanographic Inst., MS #50, Woods Hole, MA 02543, lsayigh@whoi.edu), Vincent Janik (Scottish Oceans Inst., Univ. of St. Andrews, St. Andrews, United Kingdom), Edward Kim (Univ. of Pennsylvania, Philadelphia, PA), Katherine McHugh (Chicago Zoological Society, Sarasota Dolphin Res. Program, Sarasota, FL), Peter L. Tyack (Scottish Oceans Inst., Univ. of St. Andrews, St. Andrews, Fife, United Kingdom), Randall Wells (Chicago Zoological Society, Sarasota Dolphin Res. Program, Sarasota, FL), and Frants H. Jensen (Aarhus Univ., Woods Hole, MA)

Bottlenose dolphins (*Tursiops truncatus*) produce a wide array of sounds, including clicks for echolocation and whistles for communication, both of which have been studied intensively. However, sounds other than whistles and echolocation clicks have received less attention, probably due to their high variability. These include the class of sounds loosely described as "burst pulses," which in several studies of dolphins under human care have been linked to aggressive interactions. Few studies have been carried out in the wild, beyond those describing basic acoustic parameters of sounds. Here we use acoustic and movement recording tags (DTAGs) placed simultaneously on both members of pairs of free-ranging bottlenose dolphins in Sarasota Bay, Florida, USA, to investigate acoustic behavior during aggressive interactions between male alliances and female-calf pairs. Using unsupervised clustering and discriminant function analysis on parameters such as frequency content, duration and rise time, we separate three different sound types recorded during aggressive interactions: broadband burst pulses, highly resonant cracks, and low-frequency narrowband quacks. We demonstrate how these different signal types appear to be used in the context of male-to-female aggression and/or male-male coordination. These characterizations may assist researchers analyzing acoustic recordings in assigning behavioral states to animals that are largely out of sight.

5aABb38. Potential to use passive acoustics to monitor the invasion of the Hudson River by freshwater drum. Rodney A. Rountree (23 Joshua Ln., Waquoit, MA 02536, rrountree@fishecology.org) and Francis Juanes (Biology, Univ. of Victoria, Victoria, BC, Canada)

We conducted a preliminary passive acoustic survey of the occurrence of freshwater drum in the New York State Canal System (NYSCS). Similar to more well studied marine members of the Sciaenidae, freshwater drum calls are composed of highly variable trains of 1 to 119 knocks/call (mean = 25 knocks/call), a mean knock period of 33 knocks/s, mean peak frequency of 400 Hz, and mean duration of 0.8 s. The occurrence of reproductively active freshwater drum, as evidenced by the presence of chorus calling throughout the canals, suggests that native drum populations from Lake Champlain, Lake Erie, and Lake Ontario likely all contribute to the Hudson River invasive population. We suggest that freshwater drum most likely also invaded the finger lakes through the NYSCS. The invasion of the Hudson River by freshwater drum is significant because the species had previously been geographically excluded from the entire east coast. It is a prolific species with the widest distribution of any native species in the Americas and will likely have a strong impact on the Hudson River ecosystem. We conclude that passive acoustic surveys are a highly effective non-invasive tool to monitor the future spread of freshwater drum in the Hudson River system.

Session 5aAO

Acoustical Oceanography: Tools and Methods for Ocean Mapping I

Scott Loranger, Cochair
Earth Science, University of New Hampshire, 24 Colovos Road, Durham, NH 03824

Philippe Blondel, Cochair
Physics, University of Bath, University of Bath, Claverton Down, Bath BA2 7AY, United Kingdom

Chair's Introduction—8:15

Invited Papers

8:20

5aAO1. Forty years of progress in multibeam echosounder technology for ocean investigation. Xavier Lurton (Underwater Acoust. Lab., Ifremer, IMN/NSE/ASTI, CS 10070, Plouzane 29280, France, lurton@ifremer.fr)

Along four decades, multibeam echosounders (MBES) have continuously progressed in their technology and application fields, and are today the favorite active sonar tool of several communities for ocean investigation. Measurement capabilities extended progressively from seafloor bathymetry to interface imagery and reflectometry, and to water column imaging and target quantification; the concerned domains spread from hydrography and seafloor-mapping to geosciences, offshore industry, biology and fisheries, coastal engineering, habitat mapping, environmental science and monitoring. The paper proposes an overview of this history, based on the experience of the author's oceanography institute since 1977. Technologically MBES first extended the classical single-beam echosounder toward a narrow fan of elementary beams, then to a wide-coverage design similar to sidecan sonars, to multiple swaths, and to various geometries of 3-D insonification. Various aspects of MBES technology evolution are presented: frequencies and powers; array configuration and beam features; electronics; sounding detection methods; reflectometry techniques. The progress in performance level includes: coverage extent, bathymetry accuracy, resolution, and sampling density; reflectometry reliability; water column data specificities. The importance of ancillary sensors and on-carrier platform installation is emphasized. Recent trends and possible future evolutions are finally presented, as well as some facts related to MBES environmental impact.

8:40

5aAO2. Regional seabed backscatter mapping using multiple frequencies. John E. Hughes Clarke, Anand Hiroji (Ctr. for Coastal and Ocean Mapping, Univ. of New Hampshire, Jere A. Chase Ocean Eng. Lab, 24 Colovos Rd., Durham, NH 03824, jhc@ccom.unh.edu), Glen Rice (HSTP, NOAA , Durham, NH), Fabio Sacchetti, and Vera Quinlan (INFOMAR, Marine Inst., Renville, Co. Galway, Ireland)

The frequency dependence of seabed backscatter has previously been assessed at static sites. While dependencies have been identified, the restricted range of seabed types and issues of absolute calibration have limited inferences. Until recently seabed backscatter measurements from underway mapping sonars (sidescans and multibeams) have predominantly been at a single center frequency, dictated by the range versus resolution compromise best suited for the sonar altitude. With improved range performance using FM pulses, the depth range over which a specific frequency is usable have expanded. Taking advantage of that, two national seafloor mapping programs have switched to routine collection of seabed backscatter with wavelength differences of almost an order of magnitude. The NOAA ship Thomas Jefferson and the Irish government vessel Celtic Explorer now are acquiring data at 45 and 300 kHz and 30 and 200 kHz, respectively. Even though absolute calibration remains a concern (particularly for multi-sector systems), the spatial variation of relative backscatter strength clearly provides extra discrimination between seafloors that would not separate using a single frequency. The discrimination occurs both over grazing-angle-normalized scattering strength as well as the variable shape of the angular response curves. For these two programs, data manipulation procedures and preliminary results are presented.

9:00

5aAO3. Integrating dual frequency side-scan sonar data and multibeam backscatter, angular response and bathymetry, for benthic habitat mapping in the Laganas Gulf MPA, Zakinthos Isl., Greece. Elias Fakiris, Xenophon Dimas, Nikolaos Georgiou, Dimitrios Christodoulou (Geology, Univ. of Patras, University Campus, Rio 26504, Greece, fakiris@upatras.gr), Yuri Rzhanov (Ctr. for Coastal and Ocean Mapping, Univ. of New Hampshire, Durham, NH), and George Papatheodorou (Geology, Univ. of Patras, Patras, Greece)

The preferred procedure nowadays for benthic habitat mapping is combining marine acoustic and ground truthing methods, with the former ones tending to be the swath sonars, such as the Multi Beam Echo Sounders (MBES) and the Side-Scan Sonars (SSS). Both above acoustic systems, in conjunction with an extensive underwater video footage, were employed to map in detail benthic habitats in the marine area of Laganas Gulf in the National Marine Park of Zakynthos Isl., Greece, including key protected habitats such as *P.oceanica* beds and coralligenous formations. Object oriented seafloor classification was achieved taking advantage of the multi-layer information available, including the two individual frequencies of SSS and the MBES backscatter, angular response, and bathymetry data. The extracted statistical derivatives regarded: (1) textural analysis of the multi-frequency backscatter imagery, (2) angular range analysis of the MBES backscatter, as well as (3) application of various bathymetric indices. Those derivatives were classified individually or fuzzed together, to explore the rate of improvement when choosing one system or another and to investigate their best combinations and practices towards valid seafloor classification. Benthic habitat classification has been comprehended using the NATURA and EUNIS classification schemes to be compatible with the EU regulations.

Contributed Paper

9:20

5aAO4. Target detection with multisector and multiswath echosounders. Christian de Moustier (10dBx LLC, PO Box 81777, San Diego, CA 92138, cpm@ieee.org)

Discontinuities in the ensonification pattern of certain multibeam echosounders can hamper the detection of targets in the water column and on the bottom. Such echosounders operate in single or multiswath modes with several independent transmit sectors per swath. Each sector has a unique acoustic frequency band and is steered across and along track to compensate for the vessel's pitch and yaw and to achieve a specified spatial density of soundings. This sonar geometry is optimized for bathymetry but less favorable for processing and interpretation of acoustic backscatter intensities because the sectors have different acoustic absorption profiles and are disjoint across and along track. This paper presents a method of target detection in the water column that involves frequency- and depth-dependent compensation for transmission loss in each sector, equalization of gains across all the sectors, and adaptive removal of sidelobe interferences. Echo intensities in each sector are then aggregated along track and displayed as signal-to-clutter ratio vs. altitude and distance along track. This technique allows detection of targets that are in the water column between the bottom echo trace and the ring of sidelobe interference—a region that is often deemed "blind" or too noisy for target detection.

Invited Papers

9:40

5aAO5. Multibeam sonar water column data processing tools to support coastal ecosystem science. Ian Church (Geodesy and Geomatics Eng., Univ. of NB, 15 Dineen Dr., Fredericton, NB E3B5A3, Canada, ian.church@unb.ca)

Multibeam sonar water column data are routinely collected by hydrographic survey vessels to observe and validate minimum depth measurements over anthropogenic submerged objects, which may pose a hazard to navigation. A large volume of additional data is collected during this process but mostly goes unused. This project investigates the development of processing tools and algorithms to automate the extraction of oceanographic and ecological features from the water column data files to enhance the usefulness of these datasets. Sample data from a Gulf of Mexico Research Initiative (GoMRI) project is explored where multibeam data is collected simultaneously with a towed profiling high resolution in situ ichthyoplankton imaging system (with CTD, dissolved oxygen, PAR, and chlorophyll-a fluorescence sensors). The objective is to identify and map spatial and temporal variations in biomass throughout the water column by correlating the acoustic data with imagery and sensor output from the towed profiling system. Processing the dataset to isolate and identify objects and signal patterns of interest, which might normally be considered noise, is investigated. The developed tools will aid in biological and physical feature extraction to further enhance the application of multibeam acoustic water column data.

10:00

5aAO6. Mapping methane gas seeps with multibeam and split-beam echo sounders. Thomas C. Weber (Univ. of New Hampshire, 24 Colovos Rd., Durham, NH 03824, tom.weber@unh.edu)

Modern multibeam echo sounders (MBES), which are most widely used for collecting high-resolution bathymetry and seabed imagery, often have the capability of recording acoustic backscatter from the full water column. This capability has enabled several new applications for MBES including the study of marine organisms, the quantification of suspended sediments, imaging physical oceanographic structure, and the detection, localization, and characterization of methane gas seeps. Split-beam echo sounders (SBES) are widely used in fisheries applications, but have a similarly diverse range of other applications. Here, we review the use of both MBES and SBES systems for mapping different phenomena in the water column, with a focus on mapping methane gas seeps. In doing so, we attempt to highlight the many advantages of these systems, but also discuss some of the limitations including the masking of targets by high seafloor reverberation levels in MBES systems. We also discus some of the challenges associated with wide bandwidth SBES systems, including our attempts to maintain a frequency-independent field-of-view using constant-beamwidth transducers.

10:20–10:40 Break

10:40

5aAO7. Use of acoustic estimates of euphausiid biomass in trophic dynamics and ecosystem models of the Georges Bank/Gulf of Maine Region from 1999 to 2012. J. Michael Jech (Northeast Fisheries Sci. Ctr., 166 Water St., Woods Hole, MA 02543, michael.jech@noaa.gov), Gareth L. Lawson (Biology, Woods Hole Oceanographic Inst., Woods Hole, MA), Michael Lowe (Renewable Natural Resources, Louisiana State Univ. Agricultural Ctr., Baton Rouge, LA), Sean Lucey, and Paula Fratantoni (Northeast Fisheries Sci. Ctr., Woods Hole, MA)

Euphausiids are a key link between primary production and higher-level predators in the Gulf of Maine, but are not well sampled during standard fisheries surveys. Multifrequency acoustic data may provide useful estimates of euphausiid distribution and biomass, as long as automated classification of acoustic backscatter is reliable and robust. Estimates of euphausiid biomass in the Georges Bank region of the Gulf of Maine were derived from annual acoustic/midwater trawl surveys from 1999 through 2012. Acoustic data were collected continuously with Simrad EK500 and EK60 echo sounders operating at 18, 38, and 120 kHz. Four different methods were used to classify euphausiids from the acoustic data: multifrequency single beam imaging, "dB-differencing" of 120- and 38-kHz volume backscatter, multifrequency z-score, and a multifrequency index. Scattering model predictions of euphausiid target strength and biological metrics were used to scale acoustic data to biomass for each classification method. Biomass estimates were compared among classification methods and to depth-stratified quantitative net samples to evaluate whether the acoustically-derived biomass estimates were commensurate with historical estimates. Biomass estimates were also incorporated in ecosystem models and in calculations of euphausiid consumption by fish and other predators to assess their importance in the ecosystem and their trophic significance.

Contributed Papers

11:00

5aAO8. Assessment of three sonars to evaluate the downstream migration of American Eel in the St. Lawrence River. Christopher W. Gurshin (Normandeau Assoc. Inc., 30 Int., Dr., Ste. 6, Portsmouth, NH 03801, cgurshin@normandeau.com), David J. Coughlan (Normandeau Assoc., Inc., Stanley, NC), Anna-Maria Mueller, Don J. Degan (AquAcoust., Sterling, AK), and Paul T. Jacobson (Electric Power Res. Inst., Glenelg, MD)

This study assessed the feasibility of three sonar technologies to estimate eel abundance, determine distribution, and describe approach behavior to advance strategies for providing safe downstream passage of out-migrating American Eels at hydroelectric facilities on the St. Lawrence River. A Simrad EK60 split-beam echosounder (120 kHz), Sound Metrics ARIS Explorer multibeam sonar (1100/1800 kHz), and Mesotech M3 multi-mode multibeam sonar (500 kHz) were deployed at Iroquois Dam for experimentally testing their capabilities in detecting and identifying known numbers and sizes of live adult eels tethered to surface floats released upstream of the sonar beams and allowed to swim through at known locations and times. In addition, sonars collected data continuously to monitor wild, out-migrating eels during July 15-22 and September 17-19, 2015. Results highlight several challenges in acoustically monitoring eels in a large, fast-moving river with a few orders of magnitude higher abundance of other targets that can lead to a high false positive error rate. The ARIS multibeam sonar, operating with 48 beams, holds the most promise for correctly identifying eels out to 16-20 m in range, but the M3 multibeam sonar has some value for tracking previously identified targets over larger areas.

11:20

5aAO9. Acoustic observations of the fish assemblage during a multibeam hydrographic survey of a cod spawning area. Christopher W. Gurshin (Normandeau Assoc. Inc., 30 Int. Dr., Ste. 6, Portsmouth, NH 03801, cgurshin@normandeau.com)

A two-day hydrographic survey using a dual-head Kongsberg EM3002 multibeam echosounder was conducted to map the seafloor features within 9 square kilometers of the Gulf of Maine Cod Spawning Protection Area, where Atlantic Cod (*Gadus morhua*) were known to concentrate during their spring spawning season. Acoustic backscatter from the water column collected during the first day of this survey was used to describe the fish assemblage associated with a corridor bounded by elevated features when the area was closed to commercial fishing. During the survey, a bottom trawler under a research permit made six tows where fish were observed by the hydrographic survey vessel. The species assemblage was dominated (>80%) by Atlantic Cod. Acoustic observations from the second day, when the area was open to an active commercial fishery, were compared to those during the fishing area closure. Echo statistics and position in the water column were used to characterize categories of fish detections and associate them spatially to the seafloor features and temporally to fishing closure.

11:40

5aAO10. Sediment sound speed dispersion inferences from broadband reflection coefficient measurements. Charles W. Holland (Appl. Res. Lab., The Penn State Univ., P.O. Box 30, State College, PA 16804, cwh10@psu.edu), Samuel Pinson (State College, PA), and Derek R. Olson (Appl. Res. Lab., The Penn State Univ., State College, PA)

The frequency dependence, or dispersion, of sound speed in marine sediments has been a topic of considerable interest and remains a research topic. While experiments on well-sorted sediments (having a narrow range of grain sizes) show promising concordance with theory, the more typical continental shelf sediments exhibit a rather wide range of grain sizes. A major experimental challenge is to measure *in-situ* sound speed over a sufficiently wide frequency range, such that the underlying mechanisms (e.g., viscous or friction) that control intrinsic dispersion can be isolated. Broadband 1.8-10 kHz seabed reflection measurements in the TREX13 experiment show a critical angle that is very nearly frequency independent. When effects of wavefront curvature, sound speed gradients, layering, and roughness are taken into account, this observation indicates that sediment sound speed must also be nearly independent of frequency. [Research supported by the ONR Ocean Acoustics Program.]

12:00

5aAO11. Development of a new acoustic mapping method for eelgrass and macroalgae using a multi-beam echo-sounder. Ashley R. Norton and Semme J. Dijkstra (Ctr. for Coastal and Ocean Mapping, Univ. of New Hampshire, 24 Colovos Rd., Durham, NH 03824, anorton@ccom.unh.edu)

Eelgrass and various macroalgae play important roles in temperate coastal ecosystems, including as habitat for many species, and as a bio-indicator for water quality. However, in turbid or deeper waters, the optical remote sensing methods commonly used for mapping eelgrass do not provide the necessary range for analysis. We are developing a methodology for detecting and characterizing eelgrass and macroalgae beds using water column backscatter data from multi-beam echosounder systems. We are specifically developing methods to map the maximum depth limit, percent cover, functional type (i.e., macroalgae or eelgrass) and canopy height of the beds, because these are difficult to characterize using existing optical and acoustic methods. Water column data was collected using an Odom MB1 sonar in 2014 and 2015 over a variety of vegetated sites selected to represent a range

of conditions: dense/sparse eelgrass, long/short eelgrass, mixed macroalgae and eelgrass, eelgrass on muddy or hard substrates, etc. In addition to sonar data, drop camera data was collected, and data from a regional aerial mapping program also exist for comparison. Initial data analysis shows good agreement between drop camera and sonar detections, and patches as small as $1m^2$ and as short as 20 cm are detectable.

THURSDAY MORNING, 29 JUNE 2017 BALLROOM B, 7:55 A.M. TO 12:20 P.M.

Session 5aBAa

Biomedical Acoustics and Signal Processing in Acoustics: Diagnostic and Therapeutic Applications of Ultrasound Contrast Agents I

Tyrone M. Porter, Cochair
Boston University, 110 Cummington Mall, Boston, MA 02215

Klazina Kooiman, Cochair
Thoraxcenter, Dept. of Biomedical Engineering, Erasmus MC, P.O. Box 2040, Room Ee2302, Rotterdam 3000 CA, Netherlands

Chair's Introduction—7:55

Invited Papers

8:00

5aBAa1. High frame rate imaging of microbubble contrast agents. Mengxing Tang (Dept. of BioEng., Imperial College London, London, N/A SW7 2AZ, United Kingdom, mengxing.tang@imperial.ac.uk)

The advent of microbubble contrast agents has transformed the way blood flow and tissue perfusion can be imaged using ultrasound in cardiovascular and oncological applications. The high echogenicity offered by the microbubbles greatly enhances ultrasound echoes from within the blood, making imaging of micro-vascular flow and tissue perfusion possible. Recent advances in high frame rate (HFR) ultrasound, which uses non-focused wave transmission, parallel data acquisition and digital beamforming, enable an imaging frame rate two orders of magnitude higher than the existing line-by-line scanning approach. A combination of HFR ultrasound with microbubble contrast enhanced ultrasound (CEUS) offers new exciting opportunities in imaging with unprecedented resolution and contrast. The HFR allows better tracking of fast moving target such as arterial flow and cardiac motion. Even for slow moving target, by processing the large amount of data afforded by HFR ultrasound significant improvement in image contrast/signal to noise ratio can be gained. We have developed HFR CEUS imaging methodology based on a HFR ultrasound research platform using both linear and phased array probes, and achieved a frame rate of up to tens of thousands of frames per second over multiple centimeter depth. We have demonstrated that HFR CEUS, together with signal processing, significantly improves macro- and micro-vascular flow imaging *in vivo*.

8:20

5aBAa2. Ultrafast ultrasound localization microscopy. Claudia Errico, Olivier Couture, and Mickael Tanter (CNRS, INSERM, ESPCI Paris, PSL Res. University, Institut Langevin, 17, rue moreau, Paris, IDF 75012, France, cerrico87@gmail.com)

The resolution of ultrasound imaging is limited by the classical wave diffraction theory and corresponds roughly to the ultrasonic wavelength (from 0.2 to 1 mm for clinical applications). Current methods for *in vivo* microvascular imaging are limited by trade-offs between the depth of penetration and resolution. Inspired by the optical localization techniques (FPALM), we developed the technique ultrafast ultrasound superlocalization (uULM) where the resolution is not limited by the wavelength (Couture *et al*. 2011, Desailly *et al*. 2013). The use of ultrafast ultrasound acquisitions, based on plane wave transmissions at the rate of thousand frames per second, enabled the separation of million microbubbles achieving a resolution of about 8 μm at 12 mm depth for the vascular structure of the rat brain *in-vivo* (Errico *et al*., 2015). Moreover, we have lately demonstrated that by combining acoustic vaporization of composite droplets and rapid ultrasound monitoring, ultrasound drug-delivery can also be attained with subwavelength precision (Hingot *et al*., 2016).

8:40

5aBAa3. Effects of a mono-disperse bubble population on the cumulative phase delay between second harmonic and fundamental component. Libertario Demi (TMC Europe, Da Vincilaan 5, Brussel 1903, Belgium, libertario.demi@tmceurope.com), Wim van Hoeve (Tide Microfluidics, Enschede, Netherlands), Ruud J. van Sloun (Eindhoven Univ. of Technol., Eindhoven, Netherlands), Hessel Wijkstra (Eindhoven Univ. of Technol., Amsterdam, Netherlands), and Massimo Mischi (Eindhoven Univ. of Technol., Eindhoven, Netherlands)

A positive cumulative phase delay (CPD) between the second-harmonic and fundamental component of ultrasound waves is a marker specific to ultrasound contrast agents. Dynamic contrast-enhanced ultrasound images generated by using this marker have already been reported in the literature. However, only results obtained with a poly-disperse contrast agent (SonoVue®) have been presented. In this study, we compared CPD values obtained with standard SonoVue® and with a mono-disperse contrast agent (MDCA); the latter consisted of 4-μm microbubbles produced with a MicroSphereCreator®. The same as with SonoVue®, MDCA microbubbles were made of SF6 gas encapsulated in a monolayer phospholipid shell. An ULA-OP research platform equipped with an LA332 linear-array probe was employed to image a dedicated gelatin flow-phantom. Different frequencies (1.5 to 3.5 MHz) with a mechanical index of 0.7 were used. Images were reconstructed in a tomographic fashion, and the contrast to tissue ratio (CTR) evaluated. When imaging at 1.8 MHz (close to the resonance frequency of a 4 μm microbubble), results show significantly stronger CPD values for the mono-disperse microbubbles, resulting in an overall CTR improvement by 5 dB. In conclusion, mono-disperse UCAs can be used to improve imaging performance of cumulative phase delay imaging.

9:00

5aBAa4. Subharmonic response of lipid-coated monodisperse microbubbles. Qian Li (Biomedical Eng., Boston Univ., 44 Cummington Mall, Rm. B01, Boston, MA 02215, qianli@bu.edu) and Tyrone M. Porter (Mech. Eng., Boston Univ., Boston, MA)

Subharmonic emissions from lipid-coated microbubbles, which is not radiated by tissues, can be leveraged to improve contrast-enhanced diagnostic ultrasound. In our work, we investigated subharmonic emissions from monodisperse lipid-coated microbubbles under different acoustic pressure and frequencies. First, the resonance frequency of microbubble monodispersion was determined from measured attenuation spectrum. Next, acoustic emissions from the microbubbles were detected by a transducer positioned orthogonal to the excitation transducer. Our study showed that subharmonic emissions were maximized when bubbles were driven at nearly twice the pressure-dependent resonance frequency rather than the linear resonance frequency. We also observed subharmonic emission at low excitation acoustic pressure ($<= 30$ kPa) for microbubbles coated with densely packed lipid shells, which suggests that minimizing the initial surface tension can enable subharmonic emissions at very low excitation pressures. Further studies were conducted with shells composed of varying lipids to test the influence of shell composition on the initial surface tension and subharmonic emissions. Theoretical simulations were carried out and agreed with the experimental trends. Implications of these results on the use of monodisperse lipid-coated microbubbles for subharmonic imaging will be discussed.

9:20

5aBAa5. The dynamic of contrast agent near a wall under the excitation of ultrasound wave. Nima Mobadersany and Kausik Sarkar (Mech. Eng., George Washington Univ., 800 22nd St. NW, Washington, DC 20052, sany@gwu.edu)

In this study, the behavior of contrast agents near a wall subjected to ultrasound wave have been studied numerically using boundary integral method. Contrast agents are gas filled microbubbles coated with layer of lipid or protein to prevent them against dissolution in the blood stream. Under the exposure of ultrasound wave, the contrast agents oscillate and collapse. The oscillation or collapse of the contrast agent near a wall generates shear stress resulting in the perforation of the wall to better uptake of large molecules and drugs by the tissue. In this research, we are studying the dynamics of the contrast agent, the surrounding velocity and pressure filed and the shear stress exerted on the wall due to bubble collapse for different shell rheology parameters, different standoff distances, different excitation pressures and frequencies. The study has been conducted at excitation pressures beyond the threshold of inertial cavitation where the bubble forms a jet during the collapse phase. The strain softening exponential elasticity model has been used for the interfacial rheology of the coating.

9:40

5aBAa6. Impact of temperature on the size distribution and shell properties of ultrasound contrast agents. Himanshu Shekhar, Nathaniel Smith (Dept. of Internal Medicine, Univ. of Cincinnati, 3933 Cardiovascular Ctr., 231 Albert Sabin Way, Cincinnati, OH 45267, himanshu.shekhar@uc.edu), Jason L. Raymond (Dept. of Eng. Sci., Univ. of Oxford, Oxford, United Kingdom), and Christy K. Holland (Dept. of Internal Medicine and Biomedical Eng. Program, Univ. of Cincinnati, Cincinnati, OH)

Physical characterization of ultrasound contrast agents (UCAs) is important for their efficacious use in theragnostic applications. The goal of this study was to elucidate the impact of temperature on the size distribution and shell properties of Definity®, an FDA-approved clinical UCA. A Coulter counter (Multisizer IV) was modified to enable size measurements of UCAs at elevated temperatures. The size distribution and attenuation spectrum of Definity® was measured at room temperature (25 °C) and physiological temperature (37 °C), and used to estimate the shell stiffness and viscosity of the agent at both temperatures. The attenuation coefficient of Definity® increased by as much as 5 dB at 37 °C relative to 25 °C. The highest increase in attenuation was observed at 10 MHz, the resonance frequency of Definity®. However, no significant difference was observed in the size distribution of Definity® at 25 °C and 37 °C. The estimated shell stiffness and viscosity decreased from 1.76 ± 0.18 N/m and $0.21 \times 10^{-6} \pm 0.07 \times 10^{-6}$ kg/s at 25 °C to 1.01 ± 0.07 N/m and $0.04 \times 10^{-6} \pm 0.04 \times 10^{-6}$ kg/s at 37 °C. These results indicate that the change in shell properties mediates the change in acoustic behavior of Definity® at physiological temperature.

10:00–10:20 Break

Invited Paper

10:20

5aBAa7. Gas vesicles: Acoustic biomolecules for ultrasound imaging. Mikhail G. Shapiro (Chemistry and Chemical Eng., California Inst. of Technol., 1200 E. California Blvd., Mail Code 210-41, Pasadena, CA 91125, mikhail@caltech.edu)

Expanding the capabilities of ultrasound for biological and diagnostic imaging requires the development of contrast agents linked to cellular and molecular processes *in vivo*. In optical imaging this is commonly accomplished using fluorescent biomolecules such as the green fluorescent protein. Analogously, we recently introduced gas vesicles (GVs) as the first acoustic biomolecules for ultrasound. GVs are physically stable gas-filled protein nanostructures (~250 nm) naturally expressed in aquatic photosynthetic microbes as a means to regulate buoyancy. Purified GVs produce robust ultrasound contrast across a range of frequencies at picomolar concentrations, exhibit nonlinear scattering to enable enhanced detection *versus* background *in vivo*, and have species-dependent thresholds for pressure-induced collapse to enable multiplexed imaging. Here, I will present our recent progress on understanding the biophysical and acoustic properties of these biomolecular contrast agents, engineering their mechanics and targeting at the genetic level, developing ultrasound pulse sequences to enhance their detection *in vivo* and expressing them heterologously as acoustic reporter genes. 1. Shapiro, M.G. *et al. Nat. Nanotechnol.* 9, 311-316 (2014). 2. Cherin, M. *et al. U.M.B.* (In press). 3. Lakshmanan, A. *et al. ACS Nano* 10, 7314-7322 (2016). 4. Maresca, D. *et al. In revision.* 5. Bourdeau, R.W. *et al. Submitted.* More information at http://shapirolab.caltech.edu.

Contributed Papers

10:40

5aBAa8. *In vitro* acoustic characterization of echogenic polymersomes with PLA-PEG and PLLA-PEG shells. Lang Xia, Krishna Kumar (Mech. and Aerospace Eng., George Washington Univ., 800 22nd St. NW, SEH 3961, Washington, DC 20052, langxia.org@gmail.com), Fataneh Karandish, Sanku Mallik (Dept. of Pharmaceutical Sci., North Dakota State Univ., Fargo, ND), and Kausik Sarkar (Mech. and Aerospece Eng., George Washington Univ., Washington, DC)

Echogenic liposomes (ELIPs), lipid bilayer-coated vesicles, have been widely studied as an acoustically triggerable drug delivery agent or an ultrasound contrast agent. Instead of liposomes, polymersomes, amphiphilic vesicles, offer additional stability and chemical flexibility. Here, we report the acoustic behaviors of echogenic polymersomes made of block copolymers PLA-PEG and PLLA-PEG, which are stereo-isomers. Polymersomes were excited with three different frequencies, 2.25 MHz, 5 MHz and 10 MH, and their scattered responses were measured. Both PLA-PEG and PLLA-PEG shell polymersomes produce strong acoustic responses as high as 50 dB in the fundamental component, thus demonstrating their potential as contrast agents. Significant subharmonic as well as second harmonic responses were observed at excitation frequencies of 2.25 MHz and 5 MHz. The gas dissolved in the suspension was found to be essential for the echogenicity of polymersomes.

11:00

5aBAa9. Acoustic vaporization threshold of lipid coated perfluoropentane droplets. Mitra Aliabouzar, Krishna N. Kumar, and Kausik Sarkar (Mech. & Aerosp. Eng., George Washington Univ., 800 22nd St. NW, Washington, DC 20052, mitraaali@email.gwu.edu)

Phase shift droplets that can be vaporized in situ by acoustic stimulation offer a number of advantages over microbubbles as contrast agents due to their higher stability and smaller size distribution. The acoustic vaporization threshold (ADV) of droplets with perfluoropentane (PFP) core has been investigated extensively via optical and acoustical means. However, there are noticeable discrepancies among reported ADV thresholds between the two methods. In this study, we thoroughly discuss the criteria and the experimental methodology of determining the ADV threshold. In addition, we explain the possible reasons for the discrepancies between the optical and acoustical studies of the droplet vaporization. The ADV threshold was measured as a function of the excitation frequency by examining the scattered signal from PFP droplets (400-3000 nm). The threshold increases with frequency¾2 MPa at 2.25 MHz, 2.5 MPa at 5 MHz, and 3 MPa at 10 MHz. The scattered response from droplets was also compared with the scattered response from a microbubble at the corresponding excitation pressure and frequency. We found the ADV threshold to increase with frequency. The ADV threshold values determined here are in agreement with past values obtained using an optical technique.

11:20

5aBAa10. Study of acoustic droplet vaporization using classical nucleation theory. Krishna N. Kumar, Mitra Aliabouzar, and Kausik Sarkar (George Washington Univ., 800 22nd St. NW, SEH 3000, Washington, DC 20052, krishnagwu@gwmail.gwu.edu)

Lipid coated perfluorocarbon (PFC) nanodroplets can be vaporized by an external ultrasound pulse to generate bubbles in situ for tumor imaging and drug delivery. Here we employ classical nucleation theory (CNT) to investigate the acoustical droplet vaporization (ADV), specifically the threshold value of the peak negative pressure required for ADV. The theoretical analysis predicts that the ADV threshold increases with increasing surface tension of the droplet core and frequency of excitation, while it decreases with increasing temperature and droplet size. The predictions are in qualitative agreement with experimental observations. We also estimate and discuss energy required to form critical cluster to argue that nucleation occurs inside the droplet, as was also observed by high-speed camera.

11:40

5aBAa11. A model for acoustic vaporization of droplets encapsulated within a nonlinear hyperelastic shell. Thomas Lacour, Tony Valier-Brasier, and François Coulouvrat (Institut Jean Le Rond d'Alembert UMR CNRS 7190, Université Pierre et Marie Curie, 4 Pl. Jussieu, Paris 75005, France, thomas.lacour@upmc.fr)

Nanometric particles with a liquid core arise an increasing interest for medical applications such as contrast imaging and targeted drugs delivery. Unlike conventional ultrasonic contrast agents with a gaseous core (ACU), their dimensions are smaller than the gap endothelium of tumor blood vessels, allowing them to reach tumor tissues. However, because of this liquid core, these particles are poorly compressible and therefore less efficient for imaging than standard ACUs. Their echogenicity may be enhanced by vaporizing the liquid core when applying an ultrasonic field. The investigated particles are made of a liquid core encapsulated within a visco-elastic shell. One assumes the existence of a vapor nucleus at initial time. The shell confines the liquid core and ensures greater stability, but it usually inhibits the spontaneous growth of vapor core. The dynamic of the liquid-vapor interface is described by a generalized Rayleigh-Plesset equation coupled with the heat diffusion equation in the dense phases. The vapor growth induces a significant volume expansion (typically about 100 times) and therefore, the shell is assumed to behave as a hyperelastic soft material to account for large deformation. The complete vaporization is shown to be dependent on the product of the shell shear modulus by its thickness, but also on the nonlinear elastic parameter. Numerical simulations reveal that the

bubble dynamic can be distributed into three families, thus enabling to define a threshold and an optimum of acoustical parameters.

12:00

5aBAa12. Effect of diluent fluid viscosity on acoustic droplet vaporization-mediated dissolved oxygen scavenging. Karla P. Mercado (Internal Medicine, Univ. of Cincinnati, 231 Albert Sabin Way, Cardiovascular Ctr. 3944, Cincinnati, OH 45267, karlapatricia.mercado@uc.edu), Deepak S. Kalaikadal (Mech., Industrial, and Nuclear Eng., Univ. of Cincinnati, Cincinnati, OH), John N. Lorenz (Molecular and Cellular Physiol., Univ. of Cincinnati, Cincinnati, OH), Raj M. Manglik (Dept. of Mech., Industrial, and Nuclear Eng., Univ. of Cincinnati, Cincinnati, OH), Christy Holland (Internal Medicine and Biomedical Eng. Program, Univ. of Cincinnati, Cincinnati, OH), Andrew N. Redington, and Kevin J. Haworth (Internal Medicine, Biomedical Eng. Program, and Pediatrics, Univ. of Cincinnati, Cincinnati, OH)

Acoustic droplet vaporization (ADV) can be used to scavenge dissolved oxygen and reduce the partial pressure of oxygen (pO_2) in a fluid containing perfluoropentane droplets. The impact of the diluent fluid's viscosity on ADV-mediated pO_2 reduction was investigated. Polyvinylpyrrolidone (PVP) was dissolved in saline to modify the solution's viscosity. The diluent fluid viscosity (η) and surface tension (γ) were measured. Droplets were manufactured using amalgamation and differential centrifugation to yield diameters between 1-6 μm. Droplets were diluted to 6.5x10^6 droplets/mL in saline ($\gamma = 68$ mN/m, $\eta = 0.7$ cP), 3 mg/mL PVP solution ($\gamma = 65$ mN/m, $\eta = 1.2$ cP), or 15 mg/mL PVP solution ($\gamma = 65$ mN/m, $\eta = 4$ cP). The viscosities of the 3 mg/mL and 15 mg/mL PVP solutions mimicked those of plasma and whole blood, respectively. Droplet solutions were exposed to ultrasound (5 MHz, 4.25 MPa peak negative pressure *in situ*, 10 cycles) in a 37°C *in vitro* flow system. The initial pO_2 in the fluids was 113±2 mmHg, similar to human arterial pO_2. After ultrasound exposure, the pO_2 in saline, 3 mg/mL PVP, and 15 mg/mL PVP solutions were reduced by 39.9±0.8 mmHg, 31.9±0.7 mmHg, and 16.0±0.4 mmHg, respectively. These studies indicated that ADV-mediated pO_2 reduction increased with decreasing viscosity.

THURSDAY MORNING, 29 JUNE 2017 ROOM 312, 8:20 A.M. TO 12:00 NOON

Session 5aBAb

Biomedical Acoustics: Imaging III

Kevin J. Parker, Chair
Department of Electrical & Computer Engineering, University of Rochester, Hopeman Engineering Building 203, PO Box 270126, Rochester, NY 14627-0126

Contributed Papers

8:20

5aBAb1. Characterization of modular arrays for transpinal ultrasound application. Shan Qiao (Dept. of Eng. Sci., University of Oxford, Biomedical Ultrason., Biotherapy & Biopharmaceuticals Lab. (BUBBL) Inst. of Biomedical Eng., Rm. 205, Bldg. A, 418 Guiping Rd., Xuhui District, Shanghai 2000233, China, shan.qiao03@gmail.com), Constantin Coussios, and Robin Cleveland (Dept. of Eng. Sci., University of Oxford, Biomedical Ultrason., Biotherapy & Biopharmaceuticals Lab. (BUBBL) Inst. of Biomedical Eng., Oxford, United Kingdom)

Chronic low back pain is one of the most prevalent musculoskeletal conditions worldwide, and is normally caused by the degeneration of intervertebral discs. High intensity focused ultrasound can be used to mechanically fractionate degenerate disc tissue by inertial cavitation. Due to the complexity of the spine structure, delivering sufficient focused acoustic energy to the target zone without damaging surrounding tissue is challenging and further exacerbated by patient-to-patient variability. Here we designed modular arrays, each consisting of 32 elements at 0.5MHz, which can be configured to optimize delivery for a specific patient by the means of time-reversal using the patient geometry derived from CT scans. In this study, the performance of the modular array was measured with a hydrophone and simulated numerically. For a four-module configuration the size of the focus was 4 mm in diameter and 30 mm long with a focal gain of approximately 35 and steering range of the focus was +/-30 mm in azimuth, +/-5mm in elevation, providing the required focusing flexibility and focal pressure for the transpinal application. The numerical simulations agreed well with the measurements suggesting simulations can be used for treatment planning. [Work supported by EPSRC, UK.]

8:40

5aBAb2. The H-scan analysis and results in tissues. Kevin J. Parker (Dept. of Elec. & Comput. Eng., Univ. of Rochester, Hopeman Eng. Bldg. 203, PO Box 270126, Rochester, NY 14627-0126, kevin.parker@rochester.edu)

The H-scan is based on a simplified framework for characterizing scattering behavior, and visualizing the results as color coding of the B-scan image. The methodology begins with a standard convolution model of pulse-echo formation from typical situations, and then matches those results to the mathematics of Gaussian Weighted Hermite Functions. The *n*th successive differentiation of the Gaussian pulse $G = \exp(-t^2)$ generates the *n*th order Hermite polynomial (Poularikas 2010). The function $H_4(t)G$ resembles a broadband pulse. Assuming a pulse-echo system has a round trip impulse response of $A_0H_4(t)G$, then we expect that a reflection from a step function of acoustic impedance will produce a corresponding received echo proportional to $GH_4(t)$. However, a thin layer of higher impedance, or a small scatterer or incoherent cloud of small scatterers would produce higher order Hermite functions as echoes. In this framework, the identification task is simply to classify echoes by similarity to either $GH_4(t)$, or $GH_5(t)$, or $GH_6(t)$. The resulting B-scan image is examined and echoes can be classified and colored according to their class. Results from tissue scans also demonstrate groups of echoes separated by Hermite order H_n. A theoretical framework is introduced where reflections are characterized by their similarity to *n*th order Hermite polynomials.

9:00

5aBAb3. Passive elastography in soft-tissues: Phase velocity measurement. Bruno Giammarinaro and Stefan Catheline (LabTau Inserm U1032, 151 cours Albert Thomas, Lyon cedex 03 69424, France, bruno.giam@hotmail.fr)

Elastography is an imaging technique used on medical ultrasound devices. It consists in measuring shear waves in soft tissues in order to give a tomography reconstruction of the shear elasticity. A method of measurement, usually referred as passive elastography, is to use noise correlation techniques on diffuse shear wave fields present in the medium. In human body, these fields can be naturally created by activities like heart beating, arteries pulsatility. Passive elastography allows to locally estimate the group velocity in the medium and this method has already been used for some organs as the liver, the thyroid or the brain. The present study is therefore devoted to improve this method with the calculation of the phase velocity. For example, this information, obtained for each frequency, could allow to measure the attenuation with Kramers-Krönig relations.

9:20

5aBAb4. Optical imaging of propagating ultrasonic wave fronts resulting from ultrasonic pulses incident on heel bones using refracto-vibrometry. Thomas M. Huber (Phys., Gustavus Adolphus College, 800 W College Ave., Saint Peter, MN 56082, huber@gac.edu), Matthew T. Huber, and Brent Hoffmeister (Phys., Rhodes College, Memphis, TN)

Ultrasonic measurements of the heel bone (calcaneus) are used commonly for osteoporosis screening. Pulses emitted by an ultrasound transducer are incident on the calcaneus, and the transmitted wave fronts are detected with a separate transducer. In the current *in-vitro* study, full field videos of propagating ultrasonic wave fronts incident on a calcaneus sample, along with transmitted and backscattered waves were obtained using refracto-vibrometry. Pulses were emitted by a 500 kHz Panametrics V303 transducer. To optically detect ultrasonic wave fronts, the measurement beam from a Polytec PSV-400 scanning laser Doppler vibrometer laser was directed through a water tank towards a stationary retroreflective surface. Acoustic wave fronts (density variations) which pass through the measurement laser cause variations in the integrated optical path length between the vibrometer and retroreflector. The time-varying signals detected by the vibrometer at numerous scan points were used to determine the time evolution of ultrasonic wave fronts. The resulting videos enable visualization of the propagating wave fronts incident on the calcaneus and the backscattered and transmitted wave fronts. These videos enable direct investigation of wave front distortion due to reflection, refraction and diffraction effects for pulses transmitted through the calcaneus during ultrasonic heel scanning.

9:40

5aBAb5. The effect of respiratory gas composition on kidney stone detection with the color Doppler ultrasound twinkling artifact. Julianna C. Simon (Graduate Program in Acoust., Penn State Univ., Penn State, 201E Appl. Sci. Bldg., University Park, PA 16802, jcsimon@psu.edu), Yak-Nam Wang, Jeffrey Thiel, Frank Starr, and Michael R. Bailey (Appl. Phys. Lab., Ctr. for Industrial and Medical Ultrasound, Univ. of Washington, Seattle, WA)

The color Doppler ultrasound twinkling artifact, which is thought to arise from microbubbles on and within the stone, has the potential to improve kidney stone detection in space; however, bubbles are known to be sensitive to the elevated levels of carbon dioxide (CO_2) found on space vehicles. Here, we investigate the effect of respiratory gas composition on twinkling in swine implanted with kidney stones. Thirteen swine were initially exposed to either 100% oxygen (O_2) or room air and then to air with elevated CO_2 at 0.8%, 0.54%, or 0.27%. Stones were imaged with a Verasonics ultrasound system and ATL P4-2 transducer. The 9 swine initially breathing 100% O_2 showed a significant reduction in twinkling when exposed to air with elevated CO_2, with the degree of decrease in twinkling occurring in the order: 0.8%>0.54%>0.27% CO_2. An additional 4 swine were oscillated between air with 0.04% CO_2 (normal air) and 0.5% CO_2. A reduction in twinkling was observed over the course of the experiment. The effect of respiratory gas composition should be further investigated before using twinkling to diagnose a kidney stone in space. [Work supported by NSBRI through NASA NCC 9-58 and NIH DK043881.]

10:00

5aBAb6. The native frequency of B-lines artifacts may provide a quantitative measure of the state of the lung. Libertario Demi (TMC Europe, Da Vincilaan 5, Brussel 1903, Belgium, libertario.demi@tmceurope.com), Wim van Hoeve (Tide Microfluidics, Enschede, Netherlands), Ruud J. van Sloun (Eindhoven Univ. of Technol., Eindhoven, Netherlands), Marcello Demi (Medical Imaging Processing, Fondazione Toscana Gabriele Monasterio, Pisa, Italy), and Gino Soldati (Emergency Medicine Unit, Valle del Serchio General Hospital, Lucca, Italy)

B-lines are ultrasound-imaging artifacts, which correlate with several lung-pathologies. However, their understanding and characterization is still largely incomplete. To further study B-lines, ten lung-phantoms were designed. A layer of microbubbles was trapped in tissue-mimicking gel. To simulate the alveolar size reduction typical of various pathologies, 166 and 80-micrometer bubbles were used for phantom-type 1 and phantom-type 2, respectively. A normal alveolar diameter is around 280 micrometer. A LA332 linear-array connected to the ULA-OP platform was used for imaging. Standard ultrasound imaging at 4.5 MHz was performed. Next, a multi-frequency approach was used: images were sequentially generated using orthogonal sub-bands centered at different frequencies (3, 4, 5, and 6 MHz). Results show that B-lines appear predominantly with the phantom-type 2, suggesting a link between increased artifact formation and the reduction of the alveolar size. Moreover, the multifrequency approach revealed that the B-lines have a native frequency: B-lines appeared with significantly stronger amplitude in one of the 4 images, and spectral-analysis confirmed B-lines to be centered at specific frequencies. These results can find relevant clinical application since, if confirmed by *in-vivo* studies, the native frequency of B-lines could serve as a quantitative-measure of the state of the lung.

10:20–10:40 Break

10:40

5aBAb7. Full 3D dynamic functional ultrasound imaging of neuronal activity in mice. Claire Rabut, Mafalda Correia, Victor Finel, Thomas Deffieux, Mathieu Pernot, and Mickael Tanter (INSERM U979, Institut Langevin, CNRS UMR 7587, ESPCI Paris, PSL Res. Univ., Inserm U979, 17 rue Moreau, PARIS 75012, France, claire.rabut@espci.fr)

Introduction: *In vivo* neuronal activity imaging is key to understand the mechanisms of complex brain behavior. 2D Functional Ultrasound (fUS) Imaging is a powerful tool for measuring brain activation with high spatiotemporal sampling (80μm, 1 ms) using neurovascular coupling. Here, we demonstrated the proof of concept of *in vivo* full 3-D fUS and extend the research work toward 3D functional connectivity imaging in mice. Method: A fully programmable 1024-channel ultrasound platform was used to drive 32 x 32 matrix phased arrays (9 MHz central frequency) at ultrafast frame rates (500 volumes per second). Successive plane wave emissions were compounded to produce high sensitivity vascular volumetric images of the brain of anesthetized (Kétamine-Xylazine) and craniotomized mice. Whiskers were alternatively (6 seconds ON/OFF) stimulated during acquisition. Results: High-quality 3D images of cerebral blood volume were obtained and showed the feasibility of task-activated 3D fUS imaging. The activation maps depict the spatiotemporal distribution of the hemodynamic response to whiskers stimulation at high spatial (150μm x 150μm x 150 μm) and temporal resolution (400 ms). Conclusion: We demonstrated for the first time the feasibility of full 3D dynamic functional brain imaging in mice. This paves the way toward a full-fledged neuro-imaging modality of the entire brain using ultrasound.

11:00

5aBAb8. *In-vivo*demonstration of a self-contained ultrasound-based battery charging approach for medical implants. Inder Raj S. Makin (A. T. Still Univ. of Health Sci., 5850 E Still Circle, Mesa, AZ 85206, imakin@atsu.edu), Leon Radziemski, Harry Jabs (Piezo Energy Technologies, Tucson, AZ), and T. Doug Mast (Univ. of Cincinnati, Cincinnati, OH)

Ultrasound electrical recharging (USER^{TM}) has been developed to demonstrate application-specific charging of a 200 mA-h Li-ion battery currently used in a clinical device for lower esophageal sphincter stimulation. In refining earlier developments [JASA 134(5) 4121, 2013], the receiver transducer and charging chip circuitry was miniaturized by an order of magnitude to a volume of 1.1 cc, the transducer attached directly to the 0.4 mm thick titanium device casing. Transmitter was a 1 MHz, 25 mm diameter piezo-composite disk, while the 1 MHz frequency matched receiver was either a 15 mm diameter disk or a 15 mm square tile. During a series of acute *in vivo* porcine experiments, the titanium prototype was implanted 10—15 mm deep in the subcutaneous tissue, the battery being successfully charged at a current of up to 75 mA, whereby the nominal transmitted RF power was 2 W. Maximum tissue temperature increase during the 4-hour charging cycle was 2.5 0C, directly in front of the receiver face, with no histologic thermal changes noted in the tissue post-mortem. The ultrasound approach with 10-15% system efficiency is a potentially favorable option for charging a Li-ion battery for next generation gastric stimulation implants. [Work supported by the NIH/NIBIB R43EB019225.]

11:20

5aBAb9. Model-based ultrasound attenuation estimation. Natalia Ilyina and Jan D'hooge (Cardiovascular Imaging and Dynam., KU Leuven, UZ Herestraat 49 - box 7003 50, Leuven, België 3000, Belgium, natalia.ilyina@uzleuven.be)

Ultrasound attenuation coefficient (α) has shown a potential to provide quantitative information on the pathological state of the tissue. However, the main difficulty in the estimation of α consists in the need for diffraction correction that is currently done by means of a reference measurement. Previously, we proposed an alternative attenuation reconstruction technique, wherein the attenuation coefficient was estimated by iteratively solving the forward wave propagation problem and matching the simulated signals to measured ones. The simulation procedure involved modeling of the diffraction effects and allowed to avoid several assumptions made by conventional methods. The proposed method showed promising results when applied to the data recorded using a single-element transducer. In the present study, this methodology was extended to a case of a clinically used phased array transducer. The proposed approach was validated on simulated in Field II data and data recorded in a tissue mimicking phantom with varying focal position. For the simulated data, exact α estimates were obtained regardless of the focal position, while the relative error of α for the phantom data remained below 10 %. Currently, the performance of the proposed method is validated on the phantom an *in-vivo* liver data.

11:40

5aBAb10. Transcranial ultrasound detection of intracranial hemorrhages: time-frequency analysis with empirical and variational mode decomposition. Michael P. Jordan, Amber B. Bennoui, Anjelica A. Molnar-Fenton (Chemistry and Phys., Simmons College, 300 Fenway, Boston, MA 02115, michael.jordan@simmons.edu), and Phillip J. White (Radiology, Brigham and Women's Hospital, Harvard Med. School, Boston, MA)

Ultrasound imaging has become a standard method for evaluating patients who have suffered blunt trauma. Specifically, focused assessment with sonography in trauma, commonly known as the "FAST exam," is regularly employed to detect free intraperitoneal, intrathoracic, and pericardial fluid in the setting of trauma. Currently, evaluation of the intracranial space remains a significant missing component of the FAST exam because the skull bone acts as a physical barrier to ultrasound transmission for existing approaches in pulse-echo ultrasound imaging. To address this shortcoming, we have explored a grounds-up re-evaluation of the way in which ultrasound can be used in neurosonography based on the hypothesis that backscattered signal from the intracranial space can be analyzed to identify the unique signature of pooling and coagulated blood. Preliminary experiments with tissue phantoms, *ex vivo* animal blood, and *ex vivo* human calvaria have established the parameters that optimize the SNR of pulse-echo transcranial ultrasound. The need to improve the sensitivity in isolating the desired signals have led to the exploration of several time-frequency signal analysis techniques, including the recently introduced empirical and variational mode decomposition methods. Our results have established the basis of a much-needed novel intracranial assessment technique to incorporate into the FAST exam.

Session 5aEA

Engineering Acoustics: Engineering Acoustics Topics III

Jordan Cheer, Cochair
Institute of Sound and Vibration Research, University of Southampton, University Road, Highfield, Southampton SO17 2LG, United Kingdom

Andrew W. Avent, Cochair
Mechanical Engineering, University of Bath, University of Bath, Claverton Down, Bath BA2 7AY, United Kingdom

Contributed Papers

8:00

5aEA1. Feedback control of an active acoustic metamaterial. Jordan Cheer and Stephen Daley (Inst. of Sound and Vib. Res., Univ. of Southampton, University Rd., Highfield, Southampton, Hampshire SO17 2LG, United Kingdom, j.cheer@soton.ac.uk)

Acoustic metamaterials have recently been of significant interest due to their potential ability to exhibit behavior not found in naturally occurring materials. This extends to the realization of acoustic cloaks, but perhaps of greater industrial impact is their ability to achieve high levels of noise control performance. In particular, previous research has demonstrated the high levels of transmission loss that can be achieved by an array of locally resonant elements. However, these passive metamaterials are inherently limited in performance due to both losses and their static nature. Therefore, there has been an increasing interest in active acoustic metamaterials, which can allow both increased performance and adaptability. Recent work has investigated the integration of active elements into a passive resonator-based metamaterial, and it has been demonstrated using a feedforward control architecture that significant increases in the level and bandwidth of transmission loss are achievable. However, in many practical noise control applications, it is not possible to obtain a time advanced reference signal that is required for a feedforward control implementation. Therefore, this paper will explore the design of a feedback control architecture that is applicable to the active resonator based acoustic metamaterial and demonstrate the potential performance of such a system.

8:20

5aEA2. A thermoacoustic test rig for low-onset temperature gradients and the validation of numerical models. Andrew W. Avent and Christopher R. Bowen (Mech. Eng., Univ. of Bath, Claverton Down, Bath, Sommerset BA2 7AY, United Kingdom, a.avent@bath.ac.uk)

The design and implementation of a low-onset temperature gradient standing wave thermoacoustic test rig is presented together with results which validate previous numerical models. A summary of on-going development of innovative fabrication methods and the novel use of composite materials for thermoacoustic stacks and regenerators is provided. An evaluation of the performance of these components is presented and further research and development work proposed. A novel design for selective laser sintered (SLS) aluminium thermoacoustic heat exchangers to deliver heat to from the thermoacoustic core is also presented together with numerical models and experimental validation from a purpose-built test rig.

8:40

5aEA3. Model based prediction of sound pressure at the ear drum for an open earpiece equipped with two receivers and three microphones. Steffen Vogl, Matthias Blau (Hearing Technol. and Audiol., Jade Univ., Ofenerstr. 16/19, Oldenburg 26121, Germany, steffen.vogl@jade-hs.de), and Tobias Sankowsky-Rothe (Hearing Technol. and Audiol., Jade Univ., Oldenburg, Niedersachsen, Germany)

In future hearing systems, one or more microphones and one or more receivers located within the earmold or ear canal are feasible. In order to predict the sound pressure at the ear drum in such a scenario, a one-dimensional electro-acoustic model of a prototype open earpiece with two integrated receivers and three integrated microphones was developed. The transducers were experimentally characterized by their (frequency-dependent) sensitivity (microphones) and Norton equivalents (receivers). The remaining acoustical system was modeled by 12 frequency-independent parameters which were fitted using a training set-up with well-defined loads at both sides of the ear piece. Put on an individual subject, the model could then be used to determine the acoustic impedance at the medial end of the earpiece, based on measured transfer functions between the integrated components. Subsequently, a model of the ear canal and its termination was estimated from the measured ear canal impedance, which could eventually be used to predict the drum pressure in the individual subject. Comparison to probe tube measurements of the drum pressure in 12 human subjects showed good agreement (less than ± 3 dB up to 3 kHz, less than ± 5 dB up to 6…8 kHz).

9:00

5aEA4. Development of a novel sound pressure level requirement for characterizing noise disturbances from theater and opera stages. Anton Melnikov (SBS Bühnentechnik GmbH, Bosewitzerstr. 20, Dresden 01259, Germany, anton.melnikov@sbs-dresden.de), Marcus Guettler, Monika Gatt (Tech. Univ. of Munich, Munich, Germany), Michael Scheffler (Univ. of Appl. Sci., Zwickau, Germany), and Steffen Marburg (Tech. Univ. of Munich, Munich, Germany)

In theaters or opera houses, the stage elevator is one of the most important machinery elements. It can be used for holding decorations in place and moving them between scenes or providing effects by elevating large decorations or choirs directly in a play. This scenic movement is a crucial situation from a machinery acoustics point of view. To localize and understand the sources of sound from these elevators, suitable experiments can be performed, e.g. sound pressure level (SPL) measurements, experimental and operational modal analysis. From previous measurements, the drive train could be identified as an important sound source. To quantify suitable quality requirements, the SPLs were measured in various theaters throughout Germany. The new key idea was to do the measurements while playing a drama or opera and to evaluate time periods of calm breaks during the play.

This procedure published herein, where the masking effect of noises caused by the auditorium listening to a play is taken into account, lead to a new SPL requirement for the noise radiation of the stage machinery of theaters and opera houses. Last but not least, the results of these measurements in different German theaters, are compared and discussed.

9:20

5aEA5. Analysis of the feasibility of an array of MEMS microphones to machinery condition monitoring or fault diagnosis. Lara del-Val (Industrial Eng. School. Mech. Eng. Dept., Univ. of Valladolid, Paseo del Cauce, Valladolid, Spain, lvalpue@eii.uva.es), Alberto Izquierdo, Juan J. Villacorta (TeleCommun. Eng. School. Signal Theory Dept., Univ. of Valladolid, Valladolid, Spain), Luis Suarez (Superior Tech. College, Civil Eng. Dept., Univ. of Burgos, Burgos, Spain), and Marta Herráez (Industrial Eng. School. Mech. Eng. Dept., Univ. of Valladolid, Valladolid, Spain)

During the last decades, vibration analysis has been used to evaluate condition monitoring and fault diagnosis of complex mechanical systems. The problem associated with these analysis methods is that the employed sensors must be in contact with the vibrant surfaces. To avoid this problem, the current trend is the analysis of the noise, or the acoustic signals, which are directly related with the vibrations, to evaluate condition monitoring and/or fault diagnosis of mechanical systems. Both, acoustic and vibration signals, obtained from a system can reveal information related with its operation conditions. Using arrays formed by digital MEMS microphones, which employ acquisition/processing systems based on FPGA, allows creating systems with a high number of sensors paying a reduced cost. This work studies the feasibility of the use of acoustic images, obtained by an array with 64 MEMS microphones (8x8) in a hemianechoic chamber, to detect, characterize, and, eventually, identify failure conditions in machinery. The resolution obtained to spatially identify the problem origin in the machine under test. The acoustic images are processed to extract different feature patterns to identify and classify machinery failures.

9:40

5aEA6. Numerical investigation of normal mode radiation properties of ducts with low Mach number incoming flow. José P. de Santana Neto, Danilo Braga (Dept. of Mech. Eng., Federal Univ. of Santa Catarina, Rua Monsenhor Topp, 173, Florianópolis, Santa Catarina 88020-500, Brazil), Julio A. Cordioli, and Andrey R. da Silva (Dept. of Mech. Eng., Federal Univ. of Santa Catarina, Florianópolis, Santa Catarina, Brazil, andrey.rs@ufsc.br)

Normal mode radiation properties of ducts issuing a subsonic mean flow have been thoroughly investigated in the past. The behavior of the radiation characteristics, such as the magnitude of the reflection coefficient and the length correction, are well understood. Nevertheless, the behavior of the same features in the presence of an incoming low Mach number flow has not been investigated in detail, particularly in the case of the length correction. This work presents a numerical study of the reflection coefficient and length correction of unflanged pipes and pipes terminated by circular horns of different radii in the presence of an incoming flow. The investigations are conducted with a tridimensional lattice Boltzmann scheme. The results suggest that the detachment and reattachment of the incoming flow inside the duct play a significant role on the transfer of kinetic energy from the flow to the internal acoustic field. Moreover, the results show that the mechanism of flow detachment and reattachment is highly sensitive to the geometrical characteristics at the open end.

10:00

5aEA7. Effect of taper angle on the performance of jet pumps for a loop-structured thermoacoustic engine. Ye Feng, Ke Tang, and Tao Jin (Inst. of Refrigeration and Cryogenics, Zhejiang Univ., Rd. 38 West Lake District, Yuquan Campus, Hangzhou, Zhejiang 310027, China, 972390816@zju.edu.cn)

Gedeon streaming, which circulates throughout the loop configuration with a time-averaged manner in the oscillating flow, can considerably deteriorate the efficiency of traveling-wave thermoacoustic engine. A jet pump is characterized by a tapered channel with different opening areas, which can produce a time-averaged pressure drop to suppress Gedeon streaming. Three jet pumps with different taper angles, i.e., 5°, 9°, 15°, are studied. Following Iguchi's hypothesis, the turbulent oscillating flow can be considered as a steady flow. The flow structures though the jet pumps are numerically simulated. Meanwhile, an experimental apparatus has been built to measure the performance of jet pump operating in the turbulent oscillating flow. The results show that the simulation is in a good agreement with the experiment data for the jet pumps with small taper angles. For the jet pump with 15° taper angle, the simulation results match well with the experiment data when the velocity at small opening of jet pump is higher than 50 m/s. However, when the velocity is lower, the simulation results deviate from the experiment data considerably, which can be attributed to the flow separation in the diverging direction operated differently in the steady flow and oscillating flow at lower velocity.

10:20–10:40 Break

10:40

5aEA8. Coherence resonance and stochastic bifurcation in standing-wave thermoacoustic systems. Xinyan Li and Dan Zhao (Aerosp. Eng. Div., Nanyang Technolog. Univ., 50 Nanyang Ave., Singapore, Singapore 639798, Singapore, zhaodan@ntu.edu.sg)

In this work, we develop a noisy nonlinear model and conduct experimental tests to study the stochastic bifurcation and coherence resonance in standing-wave thermoacoustic systems. When white Gaussian noise is added and its intensity is varied, it is found that the stochastic behaviors of the system can be better described by stochastic P bifurcation. When the noise intensity is chosen as a bifurcation parameter, it is shown to the bi-modal region and reduce the bi-modal area are shifted. In addition, the noise-induced coherence motions are examined and confirmed. Resonance-like behaviors of signal to noise ratio (SNR) are clearly observed. When the system approaches the critical bifurcation point, SNR is found to become larger and the optimal noise intensity is decreased to a smaller value. This property can be used as a precursor to the Hopf bifurcation in standing-wave thermoacoustic systems. Experiments are then conducted on a Rijke-type thermoacoustic system with 3 loudspeakers implemented. Transition to instability is found to be subcritical. Comparison is then made between the present theoretical and experimental results. Good qualitative agreements are obtained in terms of (1) SNR, (2) the peak height of power spectrum, and (3) the width of the frequency.

11:00

5aEA9. Analysis and design of Fresnel zone plates with multiple foci. Pilar Candelas (Centro de Tecnologías Físicas, Universitat Politècnica de València , Camino de Vera s/n, Valencia 46022, Spain, pcandelas@fis.upv.es), José Miguel Fuster (Departamento de Comunicaciones, Universitat Politècnica de València, Valencia, Spain), Constanza Rubio (Centro de Tecnologías Físicas, Universitat Politècnica de València , Valencia, Spain), Sergio Castiñeira-Ibáñez (Departamento de ingeniería Electrónica, Universitat de València, Valencia, Spain), and Daniel Tarrazó-Serrano (Centro de Tecnologías Físicas, Universitat Politècnica de València , Valencia, Spain)

Fresnel Zone Plates (FZPs) become an interesting alternative to traditional lenses when planar fabrication is advantageous, and are used in a wide range of physical disciplines such as optics, microwave propagation, or ultrasounds. Conventional FZPs produce a single focus, which is optimal in most applications. However, certain medical applications, such as MRI (magnetic resonance imaging) guided ultrasound surgery, require multiple foci ultrasound exposures. In this work, new multi-focus Fresnel lenses (MFFLs) based on conventional FZPs are presented. The advantages and drawbacks of these new MFFL structures are thoroughly analyzed. There is a tradeoff on the number of foci achieved in a single MFFL and its focusing efficiency. Therefore, the most efficient MFFL is that with two foci. A procedure for designing 2-foci MFFLs, in which the focal length of both foci may be selected independently, is established. For each 2-foci MFFL, there are several physical implementations. The focusing properties of all implementations are compared among them. From this comparison, the most efficient MFFL is selected and fully characterized.

11:20

5aEA10. Actively passive control of thermoacoustic instability. Dan Zhao, Ashique Akram Tarique (Aerosp. Eng. Div., Nanyang Technolog. Univ., 50 Nanyang Ave., Singapore, Singapore 639798, Singapore, zhaodan@ntu.edu.sg), and Shen Li (School of Energy and Power Eng., Jiangsu Univ. of Sci. and Technol., Zhenjiang City, China)

In this work, experimental studies of actively passive control of perforated liner with a bias flow on mitigating thermoacoustic instability are performed. For this, a well-designed Rijke-type thermoacoustic combustor with a perforated liner implemented is designed. A premixed propane-fueled flame is confined in the bottom half. A mean cooling flow (known as bias flow) generated from a centrifugal pump is forced to pass through the lined section. To maximize the damping capacity of the perforated liners, the bias flow rate is optimized by implementing a real-time tuning algorithm. The algorithm determines the optimum actuation signal to drive the centrifugal pump. On implementing the tuning algorithm, the unstable thermoacoustic combustor is successfully stabilized by reducing sound pressure level over 64 dB. To evaluate the off-design performance of the developed control approach, an extension tube is added/removed to the Rijke-type thermoacoustic combustor to give rise to the dominant unstable mode frequency being changed by approximately 17\%. It is found that the present actively passive control approach is able to mitigate the new limit cycle. And sound pressure level is reduced by about 41 dB. This confirms that the developed actively passive control scheme is sufficiently robust for use in real combustion systems.

11:40

5aEA11. Frequency dependence of Fresnel zone plates focus. José Miguel Fuster (Departamento de Comunicaciones, Universitat Politècnica de València, Camino de Vera s/n, Valencia 46022, Spain, jfuster@dcom.upv.es), Pilar Candelas, Constanza Rubio (Centro de Tecnologías Físicas, Universitat Politècnica de València, Valencia, Spain), Sergio Castiñeira-Ibáñez (Departamento de Ingeniería Electrónica, Universitat de València, Valencia, Spain), and Daniel Tarrazó-Serrano (Centro de Tecnologías Físicas, Universitat Politècnica de València, Valencia, Spain)

Fresnel zone plates (FZPs) focus waves through constructive interference of diffracted fields. They are used in multiple fields, such as optics, microwave propagation or acoustics, where refractive focusing by conventional lenses is difficult to achieve. FZPs are designed to work and focus at a design frequency. At this frequency, the behavior of the FZP is optimum and focusing at a certain focal length is achieved. In most medical applications using lenses, it is critical to have a fine and dynamic control on the lens focal length. In this work, the variation of the FZP focus parameters when working at operating frequencies different from the design frequency is analyzed, and a focal length control mechanism is proposed. It is shown that the FZP focal length shifts linearly with the operating frequency, becoming a dynamic control parameter that can be useful in many different applications. However, other focusing parameters, such as focal depth and distortion, are also affected by the operating frequency. These parameters establish a limit on the frequency span in which the operating frequency can be shifted, and therefore they restrict the range of focal lengths available with a single FZP.

12:00

5aEA12. Experimental study of aeroacoustic damping performance of in-duct perforated orifices with different geometric shapes. Dan Zhao, C. Z. Ji, Nuomin Han, and Xinyan Li (Aerosp. Eng. Div., Nanyang Technolog. Univ., 50 Nanyang Ave., Singapore, Singapore 639798, Singapore, zhaodan@ntu.edu.sg)

In this work, 11 in-duct perforated plates are experimentally tested in a cold-flow pipe. These plates have the same porosities but different number and geometric shaped orifices: (1) circle, (2) triangle, (3) square, (4) pentagon, (5) hexagon, and (6) star. The damping effect of these orifices is characterized by power absorption and reflection coefficient from to. It is found that the orifice shape has little influence on and at lower frequency. However, as the frequency is increased, star-shaped orifice is shown to be with much lower Δ in comparison with that of other shapes orifices. For the perforated plates with the same shaped orifices, increasing does not lead to an increase of power absorption at lower frequency. However, the orifice with the same shape and porosity but larger is found to be associated with more power absorption at approximately. Maximum power absorption is approximately at about, as. The optimum depends on the orifice shape. The present parametric measurements shed light on the roles of the number and geometric shapes of orifices and the flow parameters on its noise damping performance.

Session 5aMU

Musical Acoustics: General Topics in Musical Acoustics II

Tim Ziemer, Cochair
Institute of Systematic Musicology, University of Hamburg, Neue Rabenstr. 13, Hamburg 20354, Germany

James P. Cottingham, Cochair
Physics, Coe College, 1220 First Avenue NE, Cedar Rapids, IA 52402

Contributed Papers

8:00

5aMU1. Influence of the musician's position on the radiation impedance for transverse and notch flutes. Augustin Ernoult (LAM, Institut Jean le Rond d'Alembert, Université Pierre et Marie Curie, ∂'Alembert, boîte 162, 4, Pl. Jussieu, Paris 75252, France, ernoult@lam.jussieu.fr), Patricio de la Cuadra (Chair thématique Sorbonne universités, Pontificia Universidad Católica, Santiago, Chile), Cassandre Balosso-Bardin (Chair thématique Sorbonne universités, Univ. of Lincoln, Lincoln, United Kingdom), and Benoit FABRE (LAM, Institut Jean le Rond d'Alembert, Université Pierre et Marie Curie, Paris, France)

To play a transverse or notch flute, musicians place their mouth near an open end of the instrument where a sharp edge or labium is the target of the air-jet blown by the musicians. The jet/labium interaction is responsible for generating sound. Musicians can control the geometry of the air-jet during their performance. They can, for example, decrease the distance from their lips to the labium when increasing the pitch. The presence of the musician and the variation of his position while playing modifies the boundary conditions at the opening which, in turn, impacts the passive resonances of the instrument/player system. Flute manufacture adapted empirically to take into account this effect but, until now, it was never systematically quantified. The main goal of this study is to quantify and model the influence of the musician's presence and his movements as a modification on the radiation impedance of the instrument. Finite element simulations and experimental mockups are implemented and described in order to fit experimental models. Such models are useful as a complement to physical models of flutes as well as to understand the choices made in flute manufacture.

8:20

5aMU2. Reed chamber resonances in free reed instruments. James P. Cottingham (Phys., Coe College, 1220 First Ave. NE, Cedar Rapids, IA 52402, jcotting@coe.edu)

Western free reed instruments such as the accordion, harmonica, and harmonium do not normally employ pipe resonators to determine the pitch, but all do feature some sort of reed chamber or cavity in which the reed is mounted. The reed chamber will necessarily have resonances which can affect the tone quality and the pitch, but, since the cavity volumes are small and the resonances have high frequencies, the effects on the reed vibration generally tend to be small. In some cases, however, a resonance of the reed chamber can be close to the vibration frequency of the reed tongue. In this case, the cavity air vibration can become large enough to influence the self-excitation mechanism, possibly interfering with tongue vibration and the resulting musical tone, and in some cases preventing the sounding of the reed at all. For various configurations of the reed chamber, reed motion during the initial transient stage of free reed vibration has been analyzed, exploring effects on the rise time and final amplitude of vibration due to changes in reed chamber configurations. [Work partially supported by United States National Science Foundation Grant PHY-1004860.]

8:40

5aMU3. Numerical simulations of the turbulent flow and the sound field of the Turkish *ney* end-blown flute. Jost L. Fischer and Rolf Bader (Inst. of Systematic Musicology, Univ. Hamburg, Neue Rabenstr. 13, Hamburg, Hamburg D-20354, Germany, jost.leonhardt.fischer@uni-hamburg.de)

The Turkish *ney* is an end-blown flute in which sound generation is generated as an interplay between the air jet caused by the player and the sound pressure inside the flute tube. Sound pressure measurements inside the instrument show considerable nonlinear behavior which are crucial for the operation of the instrument and its sound character. Numerical simulations of the dynamics of both the turbulent flow field and the sound field solving the compressible Navier-Stokes equations, are performed. The transient process as well as the quasi steady-state operation mode of the instrument are performed. Varying the initial conditions of the blowing velocity as well as its attack time and shape result in the normal and in the overblown tones, which are characteristic of the instrument. Active elements like the turbulent air jet and rotating vortices around the labium as well as passive elements like the role of the mouthpiece and the resonator are discussed. A high agreement between numerical simulations and measurements is achieved.

9:00

5aMU4. Modulation and instability in the sound of plastic soprano recorders. Péter Rucz (Dept. of Networked Systems and Services, Budapest Univ. of Technol. and Economics, 2 Magyar Tuodósok körútja, Budapest H1117, Hungary, rucz@hit.bme.hu), Judit Angster (Dept. of Acoust., Fraunhofer Inst. for Bldg. Phys., Stuttgart, Baden-Württemberg, Germany), and András Miklós (Steinbeis Transfer Ctr. of Appl. Acoust., Stuttgart, Baden-Württemberg, Germany)

Soprano recorders are among the most popular wind instruments in music education for children. As these instruments are played by amateurs, they should be easy to play even by beginner players. The investigations reported in this paper were initiated by an instrument maker company. It was found that their plastic soprano recorders often had unsatisfactory sound quality and exhibited hard playability and strange, unstable steady state sounds, especially in the low regime. The experiments reported here were performed in order to examine the characteristic properties of these unusual sounds and to identify the background causes of the observed phenomena. In this contribution the results of various measurements carried out on different plastic soprano recorders are presented. Recorder sounds and edge tones are analyzed in the steady and attack transient states and their key properties are compared. The observed instabilities, strong amplitude modulations and the appearance of subharmonic components in the steady state sound are discussed. Finally, possible physical explanations of the experimental results are examined.

9:20

5aMU5. Analysis of the tonehole lattice of the northern xiao flute. Michael Prairie and Da Lei (Elec. and Comput. Eng., Norwich Univ., 158 Harmon Dr., Northfield, VT 05663, mprairie@norwich.edu)

The northern Chinese xiao is an end-blown flute characterized by an array of two to three pairs of holes that separate the main bore from an extended foot. The top pair are tuning holes and the lower ones are described as vent holes, but the design of the latter have shown an influence in the attainability of the third octave as well as the timbre of the notes (G. Ellis, personal communication, 3 March 2014). We analyze these holes in the context of an infinite lattice with a cutoff frequency f_c determined by the geometry and spacing of the top holes, and compare the results to that of the pole frequency of the calculated impedance of the actual, irregular lattice below the tuning holes. The input impedance of a cylindrical pipe with a tonehole lattice is calculated, and pressure standing waves are predicted for the peaks of the resulting admittance spectra. The standing waves are compared to experimental results to confirm the high-pass filter properties of the lattice above f_c. The effects of varying lattice dimensions on f_c and the alignment of upper harmonics with peaks in the spectra will be presented.

9:40

5aMU6. Non-planar vibrations of an ideal string against a smooth unilateral obstacle. Dmitri Kartofelev (Dept. of Cybernetics, Tallinn Univ. of Technol., Ehitajate Rd. 5, Akadeemia Rd. 21, Tallinn, Harju 19086, Estonia, dima@ioc.ee)

In this paper, we study the various possible motions of a string free to vibrate in two mutually perpendicular planes in the presence of a finite unilateral curved obstacle. We consider an obstacle which is curved only along the direction of the string at rest, and which is located at one of the ends of the string. The nonlinear problem of a non-planar string vibration against an obstacle is investigated using a kinematic numerical model under a number of simplifying assumptions. The complex interaction of the string with rigid obstacle is studied without the interfering effects of wave dissipation and dispersion. Also, it is assumed that no energy is lost due to friction and collision of the string with the obstacle. In this paper, we are especially interested in strings that are excited primarily parallel to surface of the obstacle. The modeling results show that presents of the obstacle changes dynamics of the string motion qualitatively. The conclusions of this studied idealized scenario are relevant to string vibrations in Indian stringed musical instruments like sitar or in Japanese shamisen. These lutes are equipped with finite curved bridges, and their strings are primarily excited parallel to those bridges.

10:00–10:20 Break

10:20

5aMU7. Acoustic characterization of chichin bass-drum. Sergio E. Floody and Luis E. Núñez (Universidad de Chile, Compañia 1264, Santiago, Metropolitana, Chile, eddiefloody@u.uchile.cl)

The objective of this work is to present an acoustic characterization of the *chinchín*, a bass drum type instrument native of Chile. The instrument is played by the *chinchinero*, an urban street performer in Chile, who dances acrobatically and plays the instrument simultaneously. Carried like to a backpack, the musician play the instrument with long drumsticks, which also involves a rope tied around the performer's foot to play the hi-hat cymbals. The string that drives the hi-hat cymbals passes through two holes in the cylindrical drum shell, modifying the expected sound of this type of instruments. We present detailed vibro-acoustic characterization in frequency and spatial domain using the Finite Element Method (FEM). Natural frequencies are compared with a usual bass-drum layout of similar dimensions as a baseline. Our preliminary results show differences in the fundamental frequency and the overtones of the chinchín bass drum are different from the normal one.

10:40

5aMU8. Physical model of a drum using an edge-diffraction approach. Sara R. Martín Román, U. Peter Svensson (Acoust. Res. Ctr., Dept. of Electronics and TeleCommun., Norwegian Univ. of Sci. and Technol., Trondheim, Norway, O.S. Bragstads plass 2a, Trondheim 7034, Norway, sara.martin@ntnu.no), Mark Rau, and Julius O. Smith (Music Dept., Ctr. for Comput. Res. in Music and Acoust. (CCRMA), Stanford Univ., STANFORD, CA)

Physical modeling of musical instruments is of great interest, both due to the interpretation of the parameters that make the model as similar to reality as possible, and because of the faithful sound synthesis. In this paper we will focus on the modeling of percussion instruments and more concretely drums consisting of a vibrating membrane and a rigid body. One recently published physical model [S. Bilbao and C. J. Webb, J. Audio Eng. Soc. 61 (2013) 737-748] uses finite difference time domain schemes to solve both the system of equations describing the dynamics of the membrane, and the influence of the surrounding air. The main drawback of that approach is the growth of computation cost with the scale of the problem. A new approach is suggested here where an edge diffraction-based method [A. Asheim, U. P. Svensson, J. Acoust. Soc. Am. 133 (2013) 3681-3691] is used to compute both the air loading on the surface and the sound radiation. The finite difference time domain method is used in this approach too but only to solve the dynamics of the membrane. The computational complexity is evaluated and results are compared to reference solutions.

11:00

5aMU9. Viscoelastic internal damping finite-difference model for musical instruments physical model sound production. Rolf Bader (Inst. of Musicology, Univ. of Hamburg, Neue Rabenstr. 13, Hamburg 20354, Germany, R_Bader@t-online.de)

A viscoelastic model for the internal damping of musical instrument parts, like membranes or plates is implemented within a Finite-Difference Time Domain (FDTD) method. Internal damping of wood, leather, nylon, mylar, glue, or varnish strongly change the timbre of musical instruments and the precise spectrum of this damping contributes strongly to the individual instrument character. The model assumes a complex, frequency-dependent and linear stiffness in the frequency domain, which is analytically transferred into the time-domain using a Laplace transform. The resulting mass-weighted restoring force integral of the respective membrane or plate differential equation is solved using a circular buffer accumulation method for each spatial node on the geometry, which is effective, as the model is implemented on a massive parallel Graphics Processing Unit (GPU). The model is able to reproduce arbitrarily shaped internal damping frequency responses with sharp bandwidth and fast response. The model is also able to reproduce other energy distribution problems, like energy loss or even energy supply by different parts of musical instruments through coupling, time-dependent energy loss and supply behavior, or nonlinear damping behavior, like amplitude-dependent loss strength. So also internal damping of metamaterials can be calculated with this model.

11:20

5aMU10. High-resolution directivities of played musical instruments. Timothy W. Leishman and William Strong (Phys. and Astronomy, Brigham Young Univ., N247 ESC, Provo, UT 84602, tim_leishman@byu.edu)

Recent experimental developments have enabled directivity measurements of played musical instruments with high angular resolution. The measurement system assesses directional radiation characteristics while including diffraction and absorption effects of seated musicians. The results are advantageous to better understand and visualize sound produced by the instruments, provide benchmarks for physical modeling, predict and auralize sound in rehearsal and performance venues, and improve microphone placement techniques. This paper explores steady-state directivities, contrasting key differences between brass, tone-hole, and string instruments. As expected, brass instruments generate relatively predictable results because of their single radiation elements. Tone-hole instruments produce notable interference patterns due to radiation from multiple instrument openings. String instruments produce even more complex directivities because of radiation from distributed vibrating structures and instrument openings. To illustrate the effects, the paper focuses on an instrument from each category, within the same pitch range.

11:40

5aMU11. Improvement of method for tone wood properties examination using the very near field sound pressure scanning for mode visualization. Filip Pantelic (Audio and Video Technologies, The School of Elec. and Comput. Eng. of Appl. Studies, Vojvode Stepe 283, Belgade 11000, Serbia, filip_pantelic@yahoo.com), Miomir Mijic, and Dragana Sumarac Pavlovic (School of Elec. Eng., Belgrade, Serbia)

The ability to predict behavior of some type of wood as part of a musical instrument is of a great importance. One of the important characteristics of a material, from the standpoint of musical acoustics, is Young's modulus of elasticity, which can be determined by observing the wooden beam test sample response to an excitation. In this process visualization of modes helps pairing frequencies from response spectra with vibration modes of sample, which allows us to numerically calculate Young's modulus for different frequencies. In this paper, visualization of vibration modes are achieved using the microphone Very Near Field scanning of excited samples.

THURSDAY MORNING, 29 JUNE 2017 — ROOM 203, 8:35 A.M. TO 12:20 P.M.

Session 5aNSa

Noise and Signal Processing in Acoustics: Statistical Learning and Data Science Techniques in Acoustics Research

Jonathan Rathsam, Cochair
NASA Langley Research Center, MS 463, Hampton, VA 23681

Edward T. Nykaza, Cochair
ERDC-CERL, 2902 Newmark Dr., Champaign, IL 61822

Laure-Anne Gille, Cochair
Direction territoriale Ile de France, Cerema, rue Maurice Audin, Vaulx-en-Velin 69120, France

Chair's Introduction—8:35

Invited Papers

8:40

5aNSa1. Noise forecasting: A machine-learning and probabilistic approach. Carl R. Hart, D. Keith Wilson (U.S. Engineer Res. and Development Ctr., 72 Lyme Rd., Hanover, NH 03755, carl.r.hart@usace.army.mil), Chris L. Pettit (Aerosp. Eng. Dept., U.S. Naval Acad., Annapolis, MD), and Edward T. Nykaza (U.S. Engineer Res. and Development Ctr., Champaign, IL)

Forecasting the transmission of transient noise is a challenge since several sources of uncertainty exist: source and receiver positions, meteorology, and boundary conditions. These sources of uncertainty are considered to be model parameters. Experimental observations of noise, such as peak sound pressure level, or C-weighted sound exposure level, are data parameters with their attendant sources of uncertainty. Forward models, relating model parameters to the data parameters, are also imprecise. We quantify all of these sources of uncertainty by a probabilistic approach. Probability density functions quantify a priori knowledge of model parameters, measurement errors, and forward model errors as states of information. A conjunction of these states of information is used to generate the joint probability distribution of model and data parameters. Given a forecast of model parameters, say, from a numerical weather prediction model, the joint probability distribution is marginalized in order to forecast the noise field. In this study, we examine the feasibility of this approach using, instead of numerical weather predictions, point measurements of meteorological observations and peak sound pressure level collected during a long-range sound propagation experiment. Furthermore, we examine different types of forward models based on machine learning.

9:00

5aNSa2. In-situ estimation of transmission loss based on learned dictionaries and sparse reconstruction. Jonathan Botts, Mark A. Ross (ARiA, 209 N. Commerce St., Ste 300, Culpeper, VA 22701, jonathan.botts@ariacoustics.com), Jason E. Summers, and Charles F. Gaumond (ARiA, Washington, DC)

Transmission loss is an important and notoriously difficult quantity to estimate in real underwater environments. Used to evaluate detections and threshold exceedances, estimates of transmission loss may be affected by approximate knowledge of weather conditions, bathymetry, bottom properties, and sound-speed profile—among other sources of environmental uncertainty. For applications in multistatic antisubmarine warfare, a relatively small number of in-situ measurements are available. In this work, in-situ measurements of transmission loss are used to reconstruct transmission loss fields using matching pursuit over a learned dictionary of transmission-loss fields for a given operational area. The dictionary is trained on simulated transmission-loss fields with inputs derived from historical data and plausible variations of environmental parameters. A principal challenge is constraining the problem in such a way that fields may be meaningfully reconstructed given a small set of measured data. The feasibility of reconstruction is evaluated for both full-field and depth-independent transmission-loss fields. Plausibility of practical application is evaluated with respect to realistic sampling conditions. [Portions of this material are based upon work supported by the Naval Air Systems Command.]

9:20

5aNSa3. Deep convolutional neural networks for semi-supervised learning from synthetic aperture sonar (SAS) images. Johnny L. Chen (Appl. Res. in Acoust., LLC, 209 N. Commerce St., Ste 300, Culpeper, VA 22701-2780, johnny.chen@ariacoustics.com) and Jason E. Summers (Appl. Res. in Acoust., LLC, Washington, District of Columbia)

Advancements in deep neural networks for computer-vision tasks have the potential to improve automatic target recognition (ATR) in synthetic aperture sonar (SAS) imagery. Many of the recent improvements in computer vision have been made possible by densely labeled datasets such as ImageNet. In contrast, SAS datasets typically contain far fewer labeled samples than unlabeled samples—often by several orders of magnitude. Yet unlabeled SAS data contain information useful for both generative and discriminative tasks. Here results are shown from semi-supervised ladder networks for learning to classify and localize in SAS images from very few labels. We perform end-to-end training concurrently with unlabeled and labeled samples and find that the unsupervised-learning task improves classification accuracy. Ladder networks are employed to adapt fully convolutional networks used for pixelwise prediction based on supervised training to semi-supervised semantic segmentation and target localization by pixel-level classification of whole SAS images. Using this approach, we find improved segmentation and better generalization in new SAS environments compared to purely supervised learning. We hypothesize that utilizing large unsupervised data in conjunction with the supervised classification task helps the network generalize by learning more invariant hierarchical features. [Work supported by the Office of Naval Research.]

Contributed Paper

9:40

5aNSa4. Talker age estimation using machine learning. Mark Berardi, Eric J. Hunter (Communicative Sci. and Disord., Michigan State Univ., 1026 Red Cedar Rd., Rm. 211D, East Lansing, MI 48824, mberardi@msu.edu), and Sarah H. Ferguson (Dept. of Commun. Sci. and Disord., Univ. of Utah, Salt Lake City, UT)

As a person ages, the acoustic characteristics of their voice change. Understanding how the sound of a voice changes with age may give insight into physiological changes related to vocal function. Previous work has shown changes in acoustical parameters with chronological age as well as differences between perceived age and chronological age. However, much of this previous work was done using cross-sectional speech samples, which will show changes with age but may average out important individual variability with regard to aging differences. The current study used a longitudinal recording sample gathered from a corpus of speeches from an individual spanning about 50 years (48 to 97 years of age). This study investigates how the voice changes with age using both chronological age and perceived age as independent variables; perceived age data were obtained in a previous direct age estimation study. Using the longitudinal recordings, a range of voice and speech acoustic parameters were extracted. These acoustic parameters were fitted to a supervised learning model to predict chronological age and perceived age. Differences between the chronological age and perceived age models as well as the usefulness of the various acoustic parameters will be discussed.

10:00–10:20 Break

Invited Papers

10:20

5aNSa5. Automated assessment of bird vocalization activity. Paul Kendrick, Mike Wood (Univ. of Salford, Salford, Lancashire m54wt, United Kingdom, p.kendrick@salford.ac.uk), and Luciana Barçante (Univ. of Salford, Brasilia, Brazil)

This paper presents a method for the automated acoustic assessment of bird vocalization activity using a machine learning approach. Acoustic biodiversity assessment methods use statistics from vocalizations of various species to infer information about the biodiversity. Manual annotations are accurate but time-consuming and therefore expensive, so automated assessment is desirable. Acoustic Diversity indices are sometimes used. These are computed directly from the audio and comparison between environments can provide insight about the ecologies. However, the abstract nature of the indices means that solid conclusions are difficult to reach and methods suffers from sensitivity to confounding factors such as noise. Machine learning based methods are potentially are more powerful because they can be trained to detect and identify species directly from audio. However, these algorithms require large quantities accurately labeled training data, which is, as already mentioned, non-trivial to acquire. In this work, a database of soundscapes with known levels of

vocalization activity was synthesized to allow training of the algorithm. Comparisons show good agreement between manually annotated and automatic estimates of vocalization activity in simulations and data from a field survey.

10:40

5aNSa6. Improved feature extraction for environmental acoustic classification. Matthew G. Blevins (U.S. Army Engineer Res. and Development Ctr., 2902 Newmark Dr., Champaign, IL 61822, matthew.g.blevins@usace.army.mil), Steven L. Bunkley (U.S. Army Engineer Res. and Development Ctr., Vicksburg, MS), Edward T. Nykaza (U.S. Army Engineer Res. and Development Ctr., Champaign, IL), Anton Netchaev (U.S. Army Engineer Res. and Development Ctr., Vicksburg, MS), and Gordon Ochi (Columbia College Chicago, Chicago, IL)

Modern automated acoustic classifiers have been shown to perform remarkably well with human speech recognition and music genre classification. These problems are well defined; there is a deep understanding of the source signal, and the required robustness of the model can be decreased without significantly sacrificing accuracy. Unfortunately, this simplification creates models that are insufficient when tasked with classifying environmental noise, which is inherently more variable and difficult to constrain. To further close the gap between human and computer recognition, we must find feature extraction techniques that address the additional set of complexities involved with environmental noise. In this paper, we will explore sophisticated feature extraction techniques (e.g., convolutional autoencoders and scattering networks), and discuss their effect when applied to acoustic classification.

11:00

5aNSa7. Deep learning for unsupervised separation of environmental noise sources. Bryan Wilkinson (Comput. Sci., UMBC, 1000 Hilltop Circle, Baltimore, MD 21250, bwilk7@gmail.com), Charlotte Ellison (ERDC-GRL, Alexandria, VA), Edward T. Nykaza (ERDC-CERL, Champaign, IL), Arnold P. Boedihardjo (ERDC-GRL, Alexandria, VA), Anton Netchaev (ERDC-ITL, Vicksburg, MS), Zhiguang Wang (Comput. Sci., UMBC, Baltimore, MD), Steven L. Bunkley (ERDC-ITL, Vicksburg, MS), Tim Oates (Comput. Sci., UMBC, Baltimore, MD), and Matthew G. Blevins (ERDC-CERL, Champaign, IL)

With the advent of reliable and continuously operating noise monitoring systems, we are now faced with an unprecedented amount of noise monitor data. In the context of environmental noise monitoring, there is a need to automatically detect, separate, and classify all environmental noise sources. This is a complex task because sources can overlap, vary by location, and have an unbounded number of noise sources that a monitor device may record. In this study, we synthetically generate datasets that contain Gaussian noise and overlaps for several pre-labeled environmental noise monitoring datasets to examine how well deep learning methods (e.g., autoencoders) can separate environmental noise sources. In addition to examining performance, we also focus on understanding which signal features and separation metrics are useful to this problem.

Contributed Paper

11:20

5aNSa8. Machine listening in combination with microphone arrays for noise source localization and identification. Markus Müller-Trapet (Inst. of Sound and Vib. Res., Univ. of Southampton, Southampton SO17 1BJ, United Kingdom, M.F.Muller-Trapet@soton.ac.uk), Jordan Cheer, Filippo M. Fazi (Inst. of Sound and Vib. Res., Univ. of Southampton, Southampton, Hampshire, United Kingdom), Julie Darbyshire, and J. Duncan Young (Nuffield Dept. of Clinical NeuroSci., Univ. of Oxford, Oxford, United Kingdom)

In a recent project, a large microphone array system has been created to localize and quantify noise sources in an Intensive Care Unit (ICU). In the current state, the output of the system is the location and level of the most dominant noise sources, which is also presented in real-time to the nursing staff. However, both staff as well as patients have expressed the need for information about the types of noise sources. This additional source identification can also help to find means of reducing the overall noise level in the ICU. To accomplish the source identification, the approach of machine listening with a deep neural network is chosen. A feed-forward pattern recognition network is considered in this work. However, it is not clear which types of features are best suited for the given application. This contribution thus examines the problem from a practical point of view, comparing different features including those related to sound perception, such as specific loudness, Mel-frequency cepstral coefficients, as well as the output of a gamma-tone filter bank. Additionally, the concept of time-delay networks is tested to see whether a better classification of the signals can be achieved by including their time history.

Invited Papers

11:40

5aNSa9. A likelihood-based method for point and interval estimation on auditory filter data. Andrew Christian (Structural Acoust. Branch, NASA Langley Res. Ctr., 2 N. Dryden St., Rm. 117A, M/S 463, Hampton, VA 23681, andrew.christian@nasa.gov)

A novel method for determining auditory filter (AF) shapes given a set of n-alternative forced choice (nAFC) responses from a single human subject or set of subjects is discussed. The method works by developing a function which maps individual nAFC responses into likelihood values—either supporting or conflicting with a proposed model AF shape. The aggregate of these likelihoods is then used as an objective function for optimization schemes for point estimation, or as the basis function for Metropolis Hastings-like algorithms for interval estimation, both of either parameters of the AF model or of the entire AF shape. The method is demonstrated on simulated up-down staircase data. The consistency of the method is discussed in the context of canonical methods for AF data analysis, some of which are shown to produce systematic errors. Other possible benefits of this approach are discussed including the ability of the method to: combine data from heterogeneous nAFC tasks (e.g., notched-noise maskers with tone masking) into single AF models; combine data

from different frequency-AFs into single analyses—shedding light on effects due to off-frequency listening; produce consistent results between different AF basis models.

12:00

5aNSa10. Structural equation modeling of partial and total annoyances due to urban road traffic and aircraft noises. Laure-Anne Gille (Direction territoriale Ile-de-France, Cerema / Univ Lyon, ENTPE, Laboratoire Génie Civil et Bâtiment , ENTPE/LGCB, rue Maurice Audin, Vaulx-en-Velin 69120, France, laureanne_gillechambo@yahoo.com), Catherine Marquis-Favre (Univ. Lyon, ENTPE, Laboratoire Génie Civil et Bâtiment, Vaulx-en-Velin, France), and Kin-Che Lam (Geography and Resource Management, The Chinese Univ. of Hong Kong, Hong Kong, China)

Data of a French *in situ* socio-acoustic survey were used to model partial annoyance due to urban road traffic noise, partial annoyance due to aircraft noise and total annoyance due to these combined noises. Structural equation modeling carried out on the *in situ* data showed that long-term noise annoyance depends on noise exposure but also on noise disturbance, dwelling satisfaction, visibility of a main road from the dwelling and noise sensitivity. Both noise exposure and noise sensitivity were introduced as independent variables in structural equation modeling of partial and total noise annoyances. Their contributions to the models highlight the necessity to consider these two variables in annoyance model prediction. Finally, in total noise annoyance models, whereas partial annoyance due to aircraft noise contributes to total noise annoyance as much as partial road traffic annoyance, aircraft noise exposure contributes to total noise annoyance much more than road traffic noise. Several reasons may explain this difference, such as the event character of aircraft noise or the fact that aircraft noise exposure reflects also the city exposure to aircraft noise. These hypothesis need to be confirmed on wider samples.

THURSDAY MORNING, 29 JUNE 2017 ROOM 202, 9:15 A.M. TO 12:00 P.M.

Session 5aNSb

Noise, Architectural Acoustics, Speech Communication, and Psychological and Physiological Acoustics: Effects of Noise on Human Comfort and Performance I

Z. Ellen Peng, Cochair
Waisman Center, University of Wisconsin-Madison, 1500 Highland Avenue, Madison, WI 53711

Lily M. Wang, Cochair
Durham School of Architectural Engineering and Construction, University of Nebraska - Lincoln, PKI 100C, 1110 S. 67th St., Omaha, NE 68182-0816

Anna Warzybok, Cochair
Department of Medical Physics and Acoustics, Medical Physics Group, University of Oldenburg, Universität Oldenburg, Oldenburg D-26111, Germany

Chair's Introduction—9:15

Invited Papers

9:20

5aNSb1. The quest for good, quiet spaces: Evaluating the relationship between office noise annoyance, distraction, and performance. Martin S. Lawless, Michelle C. Vigeant (Graduate Program in Acoust., The Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, msl224@psu.edu), and Andrew Dittberner (GN Hearing, Glenview, IL)

To facilitate office work performance, acousticians must design spaces that minimize annoyance from background noise, primarily from HVAC equipment, and reduce worker distraction caused by intermittent sounds, e.g., ringing telephones. Increasing background noise can mask intermittent sounds and mitigate distraction, but negatively affects annoyance. Additionally, some disrupting sounds, such as alarms, contain informational content necessary for workplaces. Balancing worker annoyance and distraction can be difficult since the definition of what constitutes a good, quiet space is yet unclear. The goal of the present work was to perform a literature review to inform ideal office noise conditions and develop an experimental procedure to test such environments. The review included papers

about indoor environmental quality and the effects of acoustics on environmental satisfaction, job performance, and noise annoyance, as well as cognitive, neurobehavioral, and physiological measures that can quantify work performance. The results of the literature survey will be used to form the basis of a future subjective study. In particular, an experimental design will be discussed that aims to evaluate the effects of various simulated environments, reproduced using higher-order Ambisonics, on work performance, annoyance, and distraction. The data from these future studies will be used to investigate ideal office acoustic environments.

9:40

5aNSb2. A simple sound metric for evaluating sound annoyance in open-plan offices. Etienne Parizet (Laboratoire Vibrations Acoustique, Univ. Lyon, INSA-Lyon, 25 bis, av. Jean Capelle, Villeurbanne 69621, France, etienne.parizet@insa-lyon.fr), Patrick Chevret, and Krist Kostallari (INRS, Vandoeuvre les Nancy, France)

Noise in open-plan offices has become a major health issue. Intelligible speech is considered as the most annoying noise sources by the occupants of such offices. Speech level fluctuations prevent people from achieving some high-demanding tasks, thus inducing annoyance and tiredness. Many studies were conducted in order to identify a sound metric closely related to this Irrelevant Speech Effect. Hongisto *et al.* have shown that Speech Transmission Effect is appropriate for evaluating the annoyance due to a neighbor in the office. More recently, Schlittmeier *et al.* suggested that the Fluctuation Strength can be used to evaluate the effect of the fluctuations of the ambient noise on task performance. This paper intends to present a new metric. It is based on the measurement of short-term temporal modulation of sound level. Results indicate that it seems to be as efficient as STI or FS, while being more suitable for in-situ experiments and usable by practitioners.

10:00

5aNSb3. Speech intelligibility under realistic classroom acoustics. Giuseppina E. Puglisi (Dept. of Energy, Politecnico di Torino, Torino, Italy), Anna Warzybok, Birger Kollmeier (Medizinische Physik and Cluster of Excellence Hearing4All, Carl von Ossietzky Universität Oldenburg, Universität Oldenburg, Oldenburg D-26111, Germany, a.warzybok@uni-oldenburg.de), and Arianna Astolfi (Dept. of Energy, Politecnico di Torino, Turin, Italy)

Speech recognition is fundamental in everyday communication environments, especially in school classrooms where the teaching-learning process takes place. Extensive literature is available that refers to speech intelligibility studies on the effect of artificially added reverberation and speech-shaped noise, whereas there is a little number of works that account for realistic acoustics. This work investigates the effect of measured classroom acoustics on speech intelligibility, accounting for the effect of informational and energetic masking, distance teacher-to-listener between and binaural unmasking. Speech reception threshold (SRT) corresponding to signal-to-noise ratio yielding 80% of speech intelligibility was measured in two acoustically different primary school classrooms. The acquired binaural room impulse responses were convolved with anechoic speech and noise stimuli, then presented via headphone to a group of adult normal-hearing listeners. The results show that SRTs were lower (better) under optimal classroom acoustics, i.e., with reverberation time of 0.4 s. Also, SRTs under informational masking noise were averagely higher (worse) by 6.6 dB than SRTs under energetic masking, proving that the former masker is more competing than the latter and needs to be deepen in future research. The binaural unmasking was observed only for a short teacher-to-listener distance in the room with a shorter reverberation time.

10:20

5aNSb4. Potential audibility and side effects of ultrasonic surveillance monitoring of PA and Life Safety Sound Systems. Peter Mapp (PMA, 101 London Rd., copford, Colchester co61lg, United Kingdom, peter@petermapp.com)

Ultrasonic surveillance monitoring, to check the operational integrity of PA and Emergency Communication Systems, has been in existence for well over 30 years—particularly in Europe. Since its inception, there has been debate as to the potential audibility that these systems may have. As the vast majority of PA systems engineers and designers have not heard or experienced any effects, is has generally been assumed that the general public do not either. Recently however, concern has been raised and claims of ill effects have been reported. There is however, little or no data as to the ultrasonic sound levels that PA systems actually emit. The paper discusses the results of an initial survey of ultrasound radiated by PA systems and compares the results with a number of international standards—there currently being little or no specific guidance. The paper reviews the technology involved, typical emission levels and concludes by making a number of recommendations to assist with the control of ultrasonic emissions from PA systems that should help to mitigate unintended side effects.

10:40–11:00 Break

11:00

5aNSb5. Relations between acoustic quality and student achievement in K-12 classrooms. Lily M. Wang, Laura C. Brill (Durham School of Architectural Eng. and Construction, Univ. of Nebraska - Lincoln, PKI 100C, 1110 S. 67th St., Omaha, NE 68182-0816, lwang4@unl.edu), Houston Lester, and James Bovaird (Educational Psych., Univ. of Nebraska - Lincoln, Lincoln, NE)

Prior work by Ronsse and Wang (2013) found that, in elementary schools, higher unoccupied background noise levels do correlate to lower student achievement scores in reading comprehension, but that study did not include detailed logs of acoustic conditions taken during the school day nor did it investigate middle or high school classrooms. More recently, measurements of the indoor environmental conditions in 110 K-12 classrooms, logged over a period of two weekdays three times seasonally, were taken over the 2015-16 academic year. Assorted acoustic metrics have been calculated from the raw measurements and a confirmatory factor analysis has been conducted to statistically create a comprehensive construct of "acoustic quality" that includes three general components: room characteristics (including reverberation times), occupied noise levels, and unoccupied noise levels. Standardized test scores of students who learned in the measured classrooms that year have also been gathered as an indicator of student achievement. Results from a structural equation model are presented to show how the various components of the proposed acoustic quality construct relate to student achievement. [Work supported by the United States Environmental Protection Agency Grant Number R835633.]

11:20

5aNSb6. Development of a methodology for field studies on the effects of aircraft noise on sleep. Sarah McGuire and Mathias Basner (Div. of Sleep and Chronobiology, Dept. of Psychiatry, Univ. of Pennsylvania Perelman School of Medicine, 423 Guardian Dr., 1013 Blockley Hall, Philadelphia, PA 19104, smcgu@upenn.edu)

An inexpensive yet sound study methodology is needed for field studies on the effects of aircraft noise on sleep. These studies are needed for developing exposure-response relationships that are representative of noise exposed communities around multiple airports and that can be used to inform policy. A methodology of monitoring sleep and identifying awakenings using ECG and actigraphy has been developed. An advantage of this approach is that ECG electrodes can be applied by investigated subjects themselves therefore reducing the need for staff in the field and the methodologic cost of the study. In addition, an automatic algorithm based on ECG and actigraphy data which identifies awakenings based on both body movements and changes in heart rate has been developed. The automatic scorings of the algorithm agree closely with awakenings identified using polysomnography which is the current gold standard for measuring sleep and related events. This ECG and actigraphy approach for monitoring sleep has been implemented in a pilot study conducted around 1 U.S. airport to evaluate its feasibility, in this study participants completed 3 nights of unattended sleep and noise measurements. Based on lessons learned, the study methodology has been further refined and implemented in a second pilot study.

Contributed Paper

11:40

5aNSb7. Study of the impact of aircraft noise on annoyance and cognitive task performance regarding the distance from the airport. Anne-Laure Verneil, Catherine Lavandier (ETIS, Université de Cergy-Pontoise, 5 mail Gay Lussac, CS 20601 Neuville, Cergy-Pontoise cedex 95031, France, anne-laure.verneil@env-isa.com), and Emilia Suomalainen (ENVISA, Paris, France)

Aviation traffic is expected to increase by 30% by 2025. Aircraft noise annoyance of people living near airports is one of the parameters which could limit this industry. Indeed, occurrence of flights takes part in noise annoyance. In addition to non-acoustic factors, acoustic parameters such as noise level or temporal and spectral aspects are involved in annoyance. In this study, two assumptions are studied: (1) close to the airport, the emergence of the flyovers constitutes the most important parameter, (2) far from the airport, this is the flyover duration. This paper presents a perceptive experiment which aims at verifying these assumptions, questioning about short term annoyance in laboratory. Three sequences of aircraft noise have been recorded at three different distances from the airport. Short term annoyance after each sound sequence has been assessed by about 50 participants. During the experiment, cognitive tasks were performed: (1) the reading of a text presented on a computer screen equipped with an eye tracker in order to measure the velocity of reading and the number of retro-saccades, (2) a memorization task where the performance is assessed with the number of errors and the reaction time. The acoustic characteristics, perceived annoyance and performance measurements are then crossed. The results are presented and discussed in this paper.

THURSDAY MORNING, 29 JUNE 2017 — ROOM 210, 9:00 A.M. TO 12:00 NOON

Session 5aPA

Physical Acoustics: General Topics in Physical Acoustics III

Bart Lipkens, Chair
Mechanical Engineering, Western New England University, 1215 Wilbraham Road, Box S-5024, Springfield, MA 01119

Contributed Papers

9:00

5aPA1. Array-based inhomogeneous soundwave generation to enhance sound transmission into solids. Trevor Kyle, Daniel C. Woods, Rahul Tiwari, Jeffrey F. Rhoads, and J. S. Bolton (School of Mech. Eng., Purdue Univ., 102 N Chauncey Ave., Apt. 203, West Lafayette, IN 47906, kylet@purdue.edu)

The acoustic excitation of energetic materials has been demonstrated to be useful in detection and defeat applications, but its efficacy is hindered by the inability to transmit a high percentage of incident acoustic energy across the air/energetic material interface. While large acoustical impedance differences usually prevent energy transmission from air into a solid, inhomogeneous incident waves have been found to transmit a significant percentage of their energy into the target material. Thus, inhomogeneous waves, whose amplitudes decay spatially in a direction different from the propagation direction, are an optimal choice for this application; however, it is difficult to create such a waveform by using a simple source. The objective of the present work is to demonstrate that by tuning the strengths and phases of sound sources in a linear array, an interference pattern can be generated such that an inhomogeneous wave forms on a surface of interest. Furthermore, it is demonstrated that by adjusting the level of inhomogeneity of the wave and its incidence angle, one can target the parameters associated with optimal sound transmission, and that these waves can be generated even in the presence of small errors in the powers and phases of the sources.

9:20

5aPA2. A spheroid model for the sound radiation of a loudspeaker on a sound bar. Vincent Roggerone (LMS, Ecole polytechnique, Laboratoire de Mécanique des Solides, École polytechnique, Palaiseau 91128, France, rogger@lms.polytechnique.fr), Etienne Corteel (Sonic Emotion Labs, PARIS, France), and Xavier Boutillon (LMS, Ecole Polytechnique, Palaiseau, France)

The sound radiation of a loudspeaker on a sound bar with a slender shape is analyzed. Measurements and boundary element method (BEM) simulations of a rectangular rigid enclosure with a flat piston turn out to be in close agreement up to the frequency limit imposed by the discretization chosen for the BEM. Looking up for a shorter computation time, we consider an analytic model based on a geometrical approximation of the sound bar by a prolate spheroid. The corresponding spheroidal coordinate system allows for an analytical solution of the sound-radiation problem. The following parameters are adjusted: geometry of the ellipse-based spheroid, size and location of the circular piston, minimum order of the spheroidal wave functions that ensures convergence. In the light of the BEM results, we also predict the frequency validity of the analytic model. In order to improve the control of the acoustical field radiated by a sound bar, we discuss the influence of the enclosure edges on the regularity of the sound field pattern. [Work supported by the ANR-13-CORD-0008 EDISON 3D grant from the French National Agency of Research.]

9:40

5aPA3. Computer simulation of synthetic apertures radar for classroom demonstration. Kathryn P. Kirkwood and Murray S. Korman (Phys. Dept., U.S. Naval Acad., 572 C Holloway Rd., Annapolis, MD 21402, m193342@usna.edu)

Synthetic Aperture Radar (SAR) has many civilian and military applications as a high resolution imaging system. A Mathematica® simulation of Synthetic Aperture Acoustic Radar will demonstrate how two-dimensional point targets on a ground plane can be imaged from a collection of acoustic echoes. A transmitter and receiver will be modeled as a one point element that stops along a linear track at collection points and hops to the next location (stop and hop approximation). The transmitter on the hypothetical apparatus will transmit acoustic signals that reflect off the targets as echoes to be collected by the receiver at each location of the track. A matched filter correlation process will complete pulse compression of the LFM (linear frequency modulated) chirp. The image reflectance of the point targets will be constructed using a time correlation/backprojection algorithm developed by Yegulap ["Fast Backprojection Algorithm for Synthetic Aperture Radar," In *Proceedings 1999 IEEE Radar Conference*, Waltham, MA, April 20-22, 1999, 60-65]. Image resolution may be improved by increasing chirp bandwidth.

10:00

5aPA4. Low-order modeling of fan broadband interaction noise. Dorien O. Villafranco and Sheryl Grace (Mech. Eng., Boston Univ., 110 Cummington Mall, Boston, MA 02215, dvillafr@bu.edu)

For commercial aircraft equipped with a modern high-bypass turbofan engine, a primary noise source on take-off and at approach has been attributed to the fan stage of the engine. The largest contribution to fan noise is rotor wake impingement on the fan exit guide vanes (FEGVs). The interaction creates both tonal and broadband noise. Engine designers have reliable tools for the prediction of tonal noise while prediction methods for broadband noise are still being developed. In the current work, a low-order method for simulating the broadband noise downstream of the fan stage of the engine is presented. Comparisons between computational results and experimental data are shown. The method produces good predictions of the spectral shape when compared to the experimental measurements. The basic low-order method models the FEGVs as flat plates. While reasonable predictions are attained with this simplification, increased fidelity is sought through inclusion of the real vane geometry in the low order model.

10:20–10:40 Break

10:40

5aPA5. The effects of diffraction on the frequency difference and frequency sum autoproducts. Brian M. Worthmann (Appl. Phys., Univ. of Michigan, 1231 Beal Ave., Ann Arbor, MI 48109, bworthma@umich.edu) and David R. Dowling (Mech. Eng., Univ. of Michigan, Ann Arbor, MI)

Previously, a remote sensing technique termed frequency difference matched field processing was developed for source localization in the shallow ocean (Worthmann *et al.*, 2015, **138**, 3549-3562). In this technique, field measurements at the in-band frequency are shifted down (or up) in frequency through the use of the bandwidth-averaged frequency-difference (or frequency-sum) autoproduct, a nonlinear construction made from field amplitudes and averaged over the available signal bandwidth. These bandwidth-averaged autoproducts may have phase structure similar to genuine acoustic fields at out-of-band frequencies when the original acoustic field is well-described by a sum of ray-path contributions. While ray theory may be a useful field description in many situations, it does not include diffraction. In this presentation, the effects of acoustic diffraction on the autoproduct are analyzed in an environment where diffraction varies in importance depending on the spatial location. Specifically, the behavior of the autoproducts is investigated in Sommerfeld's half-plane problem, where a plane wave is incident on a thin, semi-infinite rigid barrier. The bandwidth-averaged frequency difference and sum autoproduct fields are calculated in this environment, and their correlation with exact out-of-band acoustic fields are provided as a function of distance from the barrier and scattering angle. [Sponsored by NSF and ONR.]

11:00

5aPA6. Enhancing the convergence of fast multipole expansion at intermediate frequency. Hui Zhou (Univ. of Massachusetts Lowell, 1 University Ave., Lowell, MA 01854, hui_zhou@student.uml.edu)

In this work, we examined acoustic wave scattering from media having a spacial variation in its compressibility contrast. Typically, the pressure in the scattered field can be expressed as a Neumann series when the compressibility contrast is relatively small. However, divergence can occur due to resonant scattering. It has been shown that Padé Approximants method can be used to extend the range of validity of the solution. Fast multipole expansion method is applied to evaluate the terms of Neumann series. Particular interest is paid to the numerical convergence of the translation operator used in the fast multipole method.

11:20

5aPA7. Computer design, 3D printing, testing, and commercialization of a revolutionary machine gun suppressor (silencer) design*. William Moss (WCI, Lawrence Livermore National Lab., 7000 East Ave., Livermore, CA 94551, moss4@llnl.gov) and Andrew Anderson (ENG, Lawrence Livermore National Lab., Livermore, CA)

Since their invention over 100 years ago, firearm suppressors have achieved acoustic suppression using baffles and chambers to trap and delay propellant gases from exiting the muzzle of a weapon. A modern suppressor is functionally identical to the original 1908 design, with most of the improvements made by lawyers trying to circumvent extant patents. We have produced a flow-through suppressor that functions completely differently from all previous suppressors. We used a few rapid design cycles of high performance computing, 3D printing of titanium prototypes, testing, and analysis to create our suppressor, which has been patented and licensed for commercialization. Ours is the only design to simultaneously limit blowback, flash, noise, and temperature. It will last the lifetime of the barrel on single shot and fully automatic weapons, requires minimal maintenance, and therefore, is the first practical suppressor for battlefield use. If adopted for general use, the main benefit would be the reduction of debilitating long-term hearing loss, one of the most prevalent injuries in the military. [This work was performed under the auspices of the U.S. Department of Energy by Lawrence Livermore National Laboratory under Contract DE-AC52-07NA27344.]

11:40

5aPA8. Utilizing a discontinuous Galerkin method for solving Galbrun's equation in the frame of aeroacoustics. Marcus Guettler (Faculty of Mech. Eng., Tech. Univ. of Munich, Boltzmannstr. 15, Munich 85748, Germany, marcus.guettler@tum.de) and Steffen Marburg (Faculty of Mech. Eng., Tech. Univ. of Munich, Muenchen, Germany)

In the research field of aeroacoustics, scientists and engineers developed broad varieties of mathematical formulations to investigate numerically flow-induced noise in an early stage of product design and development. Besides the already established theories such as the linearized Euler equations (LEE), the linearized Navier-Stokes equations (LNSE) and the acoustic perturbation equations (APE) which are all written in a pure Eulerian frame, Galbrun utilized a mixed Eulerian-Lagrangian frame to reduce the number of unknowns. Despite the advantages of fewer degrees of freedom and the reduced effort to solve the system equations, using the usual finite element method suffers from instabilities called spurious modes that pollute the solution leaving useless results. In this work, the authors apply a discontinuous Galerkin method to overcome the difficulties related to spurious modes when solving Galbrun's equation in a mixed and pure displacement based formulation. The results achieved with the novel approach are compared with results from former attempts to solve Galbrun's equation.

THURSDAY MORNING, 29 JUNE 2017 ROOM 311, 7:55 A.M. TO 12:20 P.M.

Session 5aPPa

Psychological and Physiological Acoustics, Speech Communication, ASA Committee on Standards, Architectural Acoustics, and Signal Processing in Acoustics: Speech Intelligibility in Adverse Environments: Behavior and Modeling I

Virginia Best, Cochair
Dept. Speech, Language and Hearing Sciences, Boston University, 635 Commonwealth Ave., Boston, MA 02215

Mathieu Lavandier, Cochair
ENTPE/LGCB, Univ. Lyon, Rue M. Audin, Vaulx-en-Velin 69518, France

Chair's Introduction—7:55

Invited Papers

8:00

5aPPa1. Speech intelligibility in complex environments: Modeling and possible applications. H. Steven Colburn and Jing Mi (Biomedical Eng., Boston Univ., 44 Cummington Mall, Boston, MA 02215, colburn@bu.edu)

A summary of recent modeling work from our lab on the topic of speech intelligibility in complex environments will be presented, including some results from experiments designed to evaluate the models. The primary focus will be models of binaural processing, with the dual goals of (1) understanding the processing within the brain and (2) suggesting possible strategies for external processing that could provide more useful acoustic inputs for listeners with impaired hearing. Several processing algorithms based on allocation of individual time-frequency slices will be considered, including one based on EC processing and one based on local (in time-frequency) estimates of interaural time and intensity differences and interaural coherence. Performance of these algorithms will be evaluated using statistics of source-separation accuracy (with the ideal binary mask as the "golden standard") and also using human listening experiments. In the listening experiments, the waveforms generated by combining the time-frequency slices selected for the target location are presented to the subject in tests of speech intelligibility. Performance with these waveforms are compared to performance with the original binaural waveforms and to performance in collocated conditions. [Work supported by NIH/NIDCD Grant 2R01DC000100.]

8:20

5aPPa2. Blind modeling of binaural unmasking of speech in stationary maskers. Christopher F. Hauth, Stephan D. Ewert, and Thomas Brand (Medizinische Physik and Cluster of Excellence Hearing4All, Universität Oldenburg, Ammerländer Heerstr. 114-118, Oldenburg D-26129, Germany, thomas.brand@uni-oldenburg.de)

The equalization cancellation (EC) model predicts the binaural masking level difference by equalizing interaural differences in level and time and increasing the signal-to-noise ratio (SNR) using destructive and constructive interferences. The EC model has been successfully combined with the speech intelligibility index (SII) to predict binaural speech intelligibility. Here a blind EC model is introduced that relies solely on the mixture of speech and noise, replacing the unrealistic requirement of the separated clean speech and noise signals in previous versions. The model uses two parallel EC paths, which either maximize or minimize the EC output level in each frequency band. If SNR is negative, minimization improves the SNR by removing the interferer component from the mixed signal. If SNR is positive, maximization improves the SNR by enhancing the target component. Either the minimizing or maximizing path in each frequency band is selected blindly based on envelope frequency-selective amplitude modulation (AM) analysis. The model is evaluated for speech in stationary speech shaped noise in different spatial configurations. The suggested AM-steered selection in the EC stage demonstrates that a simple signal driven process can be used to explain binaural unmasking of speech in humans, disregarding localization and higher-level processes.

8:40

5aPPa3. Speech intelligibility and spatial release from masking in maskers with different spectro-temporal modulations. Wiebke Schubotz, Thomas Brand, and Stephan D. Ewert (Medizinische Physik and Cluster of Excellence Hearing4all, Universität Oldenburg, Carl-von-Ossietzky Str. 9-11, Oldenburg 26129, Germany, Stephan.ewert@uni-oldenburg.de)

Speech-reception thresholds (SRTs) decrease as target and maskers are spatially separated (spatial release from masking, SRM) even if two maskers are symmetrically placed around the listener's head. In this case, speech intelligibility (SI) cannot be explained by an improved long-term signal-to-noise-ratio (SNR) caused by the head shadow at one better ear alone, but could be facilitated by short-term spectro-temporal segments ("glimpses") in each ear that provide favorable SNRs. Here it was systematically assessed how SRT and SRM depend on the spectro-temporal masker properties and on the availability of specific binaural cues for a frontal target in a symmetric masker setup. Maskers ranged from stationary noise to single, interfering talkers. Maskers were modified by head-related transfer functions providing different binaural cues (interaural level and time differences; ILD, ITD, both), by presenting only glimpses derived with a fast-switching better-ear mechanism, and an "infinite ILD," removing crosstalk of the maskers between the ears. Results were compared to model predictions showing that spectral cues contribute to SRM for all maskers, while IPD and ILD cues were more important for modulated maskers. The "infinite ILD" condition suggests binaural processing limitations, resulting in a maximal SRM of 12 dB for low or absent informational masking.

9:00

5aPPa4. An account for the spatial advantage in multitalker situations based on glimpses. Esther Schoenmaker and Steven van de Par (Acoust. Group, Cluster of Excellence "Hearing4All," Univ. of Oldenburg, Carl von Ossietzkystrasse 9-11, Oldenburg D-26129, Germany, esther.schoenmaker@uni-oldenburg.de)

Spectro-temporal regions with a high local signal-to-noise ratio (SNR), so-called *glimpses*, play a vital role in the intelligibility of target speech against fluctuating interferers (e.g., concurrent speech signals). These glimpses provide access to reliable information on local signal properties. In a situation with spatially separated speech sources, a spatial advantage relative to a situation with collocated sources can be observed. This advantage is generally conceived to be composed of a monaural contribution due to better-ear listening, and a binaural contribution due to either binaural unmasking or segregation supported by spatial cues. A previous study [Schoenmaker and van de Par (2016), Adv. Exp. Med. Biol. 894, 73-81] provided evidence against the use of binaural unmasking and in favor of spatial segregation based on spatial cues extracted from glimpses. New data suggest that the better-ear contribution relies on the amount of target speech in glimpses, rather than the global SNR of the masked target speech. Together this suggests that all cues used for speech intelligibility in spatial multitalker situations are obtained from well-audible glimpses. Specifically, better-ear listening provides monaural cues to the target speech, while binaural listening provides spatial cues that improve allocation of extracted information to the correct talkers.

9:20

5aPPa5. The speech-based envelope power spectrum model (sEPSM) family: Development, achievements, and current challenges. Helia Relaño-Iborra (Dept. of Elec. Eng., Tech. Univ. of Denmark, Ørsteds Plads, Bldg. 352, Kgs. Lyngby 2800, Denmark, heliaib@elektro.dtu.dk), Alexandre Chabot-Leclerc (Dept. of Elec. Eng., Tech. Univ. of Denmark, Kongens Lyngby, Denmark), Christoph Scheidiger, Johannes Zaar, and Torsten Dau (Dept. of Elec. Eng., Tech. Univ. of Denmark, Kgs. Lyngby, Denmark)

Intelligibility models provide insights regarding the effects of target speech characteristics, transmission channels and/or auditory processing on the speech perception performance of listeners. In 2011, Jørgensen and Dau proposed the speech-based envelope power spectrum model [sEPSM, Jørgensen and Dau (2011). J. Acoust. Soc. Am. **130**(3), 1475-1487]. It uses the signal-to-noise ratio in the modulation domain (SNRenv) as a decision metric and was shown to accurately predict the intelligibility of processed noisy speech. The sEPSM concept has since been applied in various subsequent models, which have extended the predictive power of the original model to a broad range of conditions. This contribution presents the most recent developments within the sEPSM "family:" (i) A binaural extension, the B-sEPSM [Chabot-Leclerc *et al.* (2016). J. Acoust. Soc. Am. **140**(1), 192-205] which combines better-ear and binaural unmasking processes and accounts for a large variety of spatial phenomena in speech perception; (ii) a correlation-based version [Relaño-Iborra *et al.* (2016). J. Acoust. Soc. Am. **140**(4), 2670-2679] which extends the predictions of the early model to non-linear distortions, such as phase jitter and binary mask-processing; and (iii) a recent physiologically inspired extension, which allows to functionally account for effects of individual hearing impairment on speech perception.

9:40

5aPPa6. A model predicting the effect of audibility on speech reception thresholds and spatial release from masking. Mathieu Lavandier (ENTPE/LGCB, Univ. Lyon, Rue M. Audin, Vaulx-en-Velin 69518, France, mathieu.lavandier@entpe.fr), Jorg M. Buchholz (Linguist, Macquarie Univ., Chatswood, NSW, Australia), and Baljeet Rana (Linguist, Macquarie Univ., Sydney, NSW, Australia)

A binaural model is proposed to predict the effect of audibility on speech reception thresholds (SRTs) measured in the presence of two (unintelligible) vocoded-speech maskers which were either (artificially) spatially separated or co-located with the frontal speech target. Comparing these two configurations allowed to evaluate a spatial release from masking (SRM) which was based here primarily on better-ear glimpsing. Audibility was varied by testing four sound levels for the combined maskers (while the target level was varied relative to these reference levels to measure the SRTs). The proposed model is based on a short-term binaural speech intelligibility model

described by Collin & Lavandier [J. Acoust. Soc. Am. **134**, 1146-1159 (2013)] and takes the calibrated target and masker signals (independently) at each ear as inputs along with the listener hearing thresholds in order to calculate a binaural "effective" signal-to-noise ratio. Differences in ratio across conditions can be directly compared to differences in SRT. This model allows a good prediction of the decrease of SRT as well as the increase of SRM with increasing audibility/levels. The averaged absolute error between data and prediction was about 1dB across tested conditions.

10:00–10:20 Break

10:20

5aPPa7. Understanding effects of hearing loss on multitalker speech intelligibility in terms of glimpsing. Virginia Best, Christine Mason, Elin Roverud (Dept. Speech, Lang. and Hearing Sci., Boston Univ., 635 Commonwealth Ave., Boston, MA 02215, ginbest@bu.edu), Jayaganesh Swaminathan (Starkey Hearing Res. Ctr., Berkeley, CA), and Gerald Kidd (Dept. Speech, Lang. and Hearing Sci., Boston Univ., Boston, MA)

In multitalker mixtures, listeners with hearing loss demonstrate reduced spatial release from masking compared to listeners with normal hearing. However, it is not clear whether this problem reflects an inability to use spatial cues to segregate sounds, or a degraded representation of the target speech itself. In this work, a simple monaural glimpsing model was used to isolate the target information that is potentially available at each ear in spatialized speech mixtures, and intelligibility of these glimpsed stimuli was then measured directly. Performance in the glimpsed condition was strongly correlated with performance in the natural spatial condition, suggesting a common limit in both cases. Similar results were found for different kinds of speech mixtures in which the target and maskers were distinguished by cues other than spatial location (talker sex, or time-reversal). The results suggest that the primary detrimental effect of hearing loss might be on the representation of target glimpses, rather than on the segregation of competing talkers, in multitalker mixtures.

10:40

5aPPa8. The effect of listener head orientation on speech intelligibility in noise. Jacques A. Grange and John F. Culling (School of Psych., Cardiff Univ., Park Pl., Cardiff, Wales CF10 3AT, United Kingdom, CullingJ@cf.ac.uk)

The signal-to-noise ratio at one ear is progressively improved by orienting the head away from the target sound source by up to 65 degrees, so facing the speaker in a sidelong way may be an effective listening tactic in noisy listening situations. In a listening configuration that optimized this effect in a sound-treated room, speech reception thresholds improved by up to 10 dB with head orientation, but without instruction few listeners adopted the tactic. Audio-visual presentation of the target speech further suppressed its use [Grange and Culling, J. Acoust. Soc. Am. **139**, 703-712 (2016)]. Audio-visual presentation produced an additive lip-reading benefit that was unaffected by a head orientation 30 degrees away from the speaker. A head-orientation benefit was also observed in realistic listening conditions, simulated over headphones. Binaural room impulse responses from a real restaurant were used to simulate listening at six different tables with nine concurrent interferers. Head orientation of 30 degrees produced a mean benefit of 1 dB for speech interferers and 1.3 dB for noise interferers [Grange and Culling, J. Acoust. Soc. Am. **140**, 4061-4072 (2016)]. These results suggest that listeners would benefit from advice to orient away from the speaker while maintaining eye contact.

Contributed Papers

11:00

5aPPa9. Objective evaluation of binaural noise-reduction algorithms for the hearing-impaired in complex acoustic scenes. Marc René Schädler, Anna Warzybok, and Birger Kollmeier (Medizinische Physik and Cluster of Excellence Hearing4all, Universität Oldenburg, Oldenburg D-26111, Germany, a.warzybok@uni-oldenburg.de)

The simulation framework for auditory discrimination experiments (FADE) was used to predict the benefit in speech reception thresholds (SRT) with the German matrix sentence test when using a range of single- and multi-channel noise-reduction algorithms in complex acoustic conditions. FADE uses a simple robust automatic speech recognizer to predict SRTs from simulated speech recognition experiments in an objective way, independent from any empirical reference data. Here, it was extended with a simple binaural stage and individualized by taking into account the audiogram. Empirical data from the literature was used to evaluate the model in terms of predicted SRTs and benefits in SRT when using eight different noise-reduction algorithms. In a realistic binaural cafeteria condition, FADE explained about 90% of the variance of the empirical SRTs for normal-hearing listeners and predicted the corresponding benefits in SRT with a root-mean-square prediction error of 0.6 dB. In contrast to the surprisingly high performance of this simple approach for normal-hearing listeners, much less of the inter-individual variance can be explained for hearing-impaired listeners, where individual audiograms only sufficed for aided group performance predictions. In a single competing talker condition, the prediction even failed for the normal listeners, clearly demonstrating the limits of the current versatile approach and demanding further extensions.

11:20

5aPPa10. The interaction between reverberation and digital noise reduction in hearing aids: Acoustic and behavioral effects. Paul Reinhart (Commun. Sci. and Disord., Northwestern Univ., 2240 Campus Dr., Evanston, IL 60208, preinhart@u.northwestern.edu), Pavel Zahorik (Univ. of Louisville, Louisville, KY), and Pamela Souza (Commun. Sci. and Disord., Northwestern Univ., Evanston, IL)

Digital noise reduction (DNR) is widely implemented in hearing aids to improve the signal-to-noise ratio (SNR) of speech-in-noise. To accomplish this, the DNR processor must be able to accurately discriminate the speech versus noise signals. Acoustically, reverberation causes the blending of the speech and noise signals. The purpose of the present experiment was to examine whether reverberation impacts the benefits of DNR processing. Speech stimuli were combined with white noise at multiple SNRs. Speech-in-noise signals were processed using virtual auditory space techniques to simulate reverberation times and with a DNR simulation that mimicked hearing aid processing based on spectral subtraction. Signals were acoustically analyzed to quantify changes in SNR as a result of DNR processing. As reverberant degradation increased, the improvement in SNR decreased. Behaviorally, hearing-impaired individuals listened to low-context sentences in noise with varying reverberation either with or without DNR processing. Without reverberation, DNR had no or minimal impact on speech intelligibility, consistent with previous work. However, as reverberant degradation increased, the effects of DNR on speech intelligibility were variable. These results suggest that the benefit of DNR processing in hearing aids in noisy environments may depend on the amount of reverberation in the environment. [Work supported by NIH.]

11:40

5aPPa11. The effect of room acoustics on speech intelligibility and spatial release from masking. Thomas Biberger and Stephan D. Ewert (Medizinische Physik and Cluster of Excellence Hearing4All, Universität Oldenburg, Carl-von-Ossietzky-Straße 9-11, Oldenburg, Lower Saxony 26135, Germany, thomas.biberger@uni-oldenburg.de)

In daily life, verbal communication often takes place in indoor situations with interfering sounds, where speech intelligibility (SI) is affected by (i) masking and (ii) reverberation. Both introduce spectral and temporal changes to the signal. A critical spatial configuration to assess (binaural) SI is a frontal target speaker and two interfering sources symmetrically placed to either side (± 60°). Here a spatial release from masking (SRM) is observed in comparison to co-located frontal target and interferers, showing that the auditory system can make use of temporally fluctuating interaural differences. Room reverberation affects the temporal representation of the target and maskers and, moreover, the interaural differences depending on the spatial configuration and room acoustical properties. Here the effect of room acoustical properties (room size, T60, frequency dependency of T60), temporal structure of the interferers, and direct to reverberation ratio (DRR) on speech reception thresholds (SRT) and SRM were systematically assessed in a simulated room using headphone-based virtual acoustics. For constant T60 and DRR a different room size resulted in, e.g., significantly different SRTs but similar SRMs, implying the temporal structure of reverberation is less relevant for exploiting binaural cues. Data are discussed and compared to predictions of a binaural SI model.

12:00

5aPPa12. On the relationship between a short-term objective metric and listening efficiency data for different noise types. Nicola Prodi and Chiara Visentin (Dipartimento di Ingegneria, Università di Ferrara, via Saragat 1, Ferrara 44122, Italy, nicola.prodi@unife.it)

This study aims to compare the distinct effects of a steady-state (SSN) and a fluctuating (ICRA) masker on speech reception performance. SNR, reverberation and masker type were combined as to create several acoustic scenarios; matrixed-word listening tests in the Italian language were proposed to a panel of young adults with normal hearing, collecting data on intelligibility scores (IS) and response time (RT). The listening conditions were objectively qualified with the short-term metric STIr, defined as the average of the STI values calculated over short time-windows, whose duration reflects the typical phoneme length. The results showed that for a given STIr, both maskers yield the same IS, being the fluctuation benefit already accounted for by the objective metric. The slope of the STIr-IS function only depends on the speech material. Anyway, the fluctuating masker calls for an increased amount of cognitive resources to be deployed in the speech reception process, traced by a statistically significant higher response time. These results shade a new light on the fluctuating masker release (FMR) phenomenon.

THURSDAY MORNING, 29 JUNE 2017 — ROOM 300, 8:00 A.M. TO 12:20 P.M.

Session 5aPPb

Psychological and Physiological Acoustics: Sound Localization and Binaural Hearing

Griffin D. Romigh, Chair

Air Force Research Labs, 2610 Seventh Street, Area B, Building 441, Wright Patterson AFB, OH 45433

Contributed Papers

8:00

5aPPb1. Influence of interaural time differences on the loudness of low-frequency pure tones at varying signal and noise levels. Gauthier Berthomieu, Vincent Koehl, and Mathieu Paquier (Lab-STICC UMR 6285, Univ. of Brest, 6 Ave. Le Gorgeu, Brest 29200, France, gauthier.berthomieu@univ-brest.fr)

Directional loudness sensitivity, which is generally accounted for by at-ear pressure modifications because of the perturbation of the sound field by the head, has been reported to occur at 400 Hz where shadowing effects are usually considered small. Then, an effect of the interaural time difference (ITD) on loudness has been observed for pure tones below 500 Hz. The latter was rather small but still significant, contributing to directional loudness sensitivity. In addition, it has been shown that the effect of ITD on loudness was caused by the ITD itself and not by its related localization. As this effect appeared significant at low level only (40 phon), it was hypothesized that ITD could help separate the signal from the internal noise and enhance its loudness. The aim of the present study is to confirm this hypothesis by observing the effect of ITD on the loudness of low-frequency pure tones (100 and 200 Hz) for various signal-to-noise ratios. The signal level was varied from 30 to 90 phon and the noise could be internal only or external also. The effect of ITD appeared significant up to 40 or 50 phones depending on the frequency.

8:20

5aPPb2. Computational study of head geometry effects on sound pressure gradient with applications to head-related transfer function. Mahdi Farahikia and Quang T. Su (Mech. Eng., SUNY Binghamton, 13 Andrea Dr. Apt. A, Vestal, NY 13850, mfarahi1@binghamton.edu)

Effects of object geometry on sound scattering of far-field incident sound waves for two different head models (spherical and ellipsoidal) have been studied using an optimized Finite Element Method (FEM) based on frequency-dependent adaptive dimensions. This optimized FEM technique is proven both efficient and accurate when compared with analytical results for the spherical model, with maximum deviation of 0.6 dB. Comparisons between models have been made on the equivalent Head-Related Transfer Functions (HRTFs) for acoustic pressure, and for the first and second order pressure gradients on the surface. It is shown that while directionality cannot be achieved at lower frequencies using only pressure, pressure gradients provide sound cancellation for certain source orientations. Hence, it is possible to cancel incoming sound from the front (or behind) or sides depending on the direction (radial, azimuthal) and order of pressure gradients. While the pattern of pressure gradient directionality remains similar between the spherical and ellipsoidal models, the difference in dimensions affects the amplitude of the equivalent HRTFs for these parameters. This study provides insight into the placement of directional microphones on the head for hearing aids.

8:40

5aPPb3. My personal head related transfer function: High quality individualized computer models. Sebastian Fingerhuth, Danny Angles, and Juan Barraza (School of Elec. Eng., Pontificia Universidad Católica de Valparaíso, Av. Brasil 2147, Valparaíso 2362804, Chile, sebastian.fingerhuth@pucv.cl)

In this paper, we present a method to obtain individual 3D CAD-models of the head and pinna. The method is a hybrid method: A photogrammetric technique for the head is combined with a molding process for the pinna. For the 3D reconstruction of the head, a set of pictures from different perspectives of the person is taken. Therefore the person is stepwise rotated while seated and pictures are taken using semi-professional photographic equipment. Four different orbits are used: top level, upper level, eye level, and low level. These pictures, about 120, are then combined using a commercial photogrammetric software. Due to the complex and concave geometry of the pinna, an additional step has to be taken. An alginate mold is made for each pinna which then is molded again to obtain a positive plaster replica of the pinna. This replica is then converted to a CAD model. For this last step, two methods were compared: the same photogrammetric process as before and using a 3D scanner. Both CAD models, head and pinna, are then carefully combined into one CAD mesh. The CAD-models can then be used to compute HRTFs by means of the Boundary Element Method (BEM).

9:00

5aPPb4. Predicting sound-localization performance with hearing-protection devices using computational auditory models. Paul Calamia, Christopher Smalt, Shakti K. Davis, and Austin Hess (BioEng. Systems and Technologies Group, MIT Lincoln Lab., 244 Wood St., Lexington, MA 02420, pcalamia@ll.mit.edu)

Evaluation of the effect of hearing-protection devices (HPDs) on auditory tasks such as detection, localization, and speech intelligibility typically is done with human-subject testing. However, such data collections can be impractical due to the time-consuming processes of subject recruitment and the testing itself, particularly when multiple tasks and HPDs are included. An alternative, objective testing protocol involves the use of a binaural mannequin (a.k.a an acoustic test fixture) and computational models of the auditory system. For example, data collected at the eardrums of such a mannequin outfitted with an HPD can be fed into a binaural localization model. If the performance of the model with such input can be shown to be similar to that of human subjects, the model-based assessment may be sufficient to characterize the hearing protector and inform further design decisions. In this presentation we will describe the preliminary results of an effort to replicate human-subject localization performance for 5 HPDs and the open ear using an acoustic test fixture and three auditory localization models. The task involved localizing the direction of a gun-cocking sound from the center of a 24-loudspeaker ring. Variations among the models, as well as a comparison to the human-subject data will be discussed. [Work sponsored by US Army NSRDEC.]

9:20

5aPPb5. Localization of speech in noise with and without self-directed room exploration. Samuel W. Clapp and Bernhard U. Seeber (Audio Information Processing, Technische Universität München, Arcisstr. 21, München 80333, Germany, samuel.clapp@tum.de)

The auditory system adapts to reflections by suppressing them in favor of the direct sound, aiding greatly in sound localization and speech understanding in reverberation. Previous studies have shown an increase in the echo threshold over repeated exposure to a reflection pattern, and the breakdown of this effect when the pattern changes suddenly. These studies examined this effect as a low-level cognitive process. The current study gives listeners a more active role in adapting to reflections, to see if this can engage higher-level processes more durable over time. In a psychoacoustic test, listeners localized a short speech signal in the presence of either point-like or diffuse noise in a simulated room, using the Simulated Open Field Environment for simulation and playback over a loudspeaker array. In the "Block" condition, subjects heard stimuli from the same room in all trials. In the "Interrupt" condition, trials in the target room were interspersed with dummy trials from different spaces. In the "Learning" condition, listeners used a GUI to position a virtual source in a room and hear the resulting auralizations before each block of trials, conducted as in the "Interrupt" condition. Localization was most accurate in the "Learning" condition, particularly in the presence of the point-like noise source, and for target locations the furthest angular distance from the noise. These results suggest it is possible for actively engaged listeners to learn the acoustics of a room in a temporally durable manner.

9:40

5aPPb6. Binaural detection-based estimates of precision of coding of interaural temporal disparities across center frequency. Leslie R. Bernstein and Constantine Trahiotis (UConn Health Ctr., MC3401, 263 Farmington Ave., Farmington, CT 06030, Les@neuron.uchc.edu)

This presentation reports binaural detection data obtained using three masking configurations previously shown, in toto, to reveal: (1) whether listeners can compensate internally for external interaural temporal disparities (ITDs) and (2) the precision with which such compensation of ITD changes as a function of reference ITD [L. R. Bernstein and C. Trahiotis, J. Acoust. Soc. Am. 138, EL474-EL479 (2015)]. The current experiment extended that paradigm in order to obtain binaural detection data at center frequencies that ranged in octave steps from 250 Hz to 8 kHz. Results indicate that, at all center frequencies, listeners are able to compensate internally for externally imposed ITDs. Such compensation allows them to outperform what would be expected were they only utilizing changes in interaural correlation produced by the addition of the tonal signal to the noise masker. The data will be discussed in terms of quantitative analyses that show how detection performance is constrained by two distinct sources of interaurally and mutually uncorrelated internal noise: one having a power that increases with the power of the external masker and the other having a power that grows with the magnitude of the ITD.

10:00–10:20 Break

10:20

5aPPb7. A model of the ongoing precedence effect. Patrick Zurek (Sensimetrics Corp., 14 Summer St., Malden, MA 02148, pat@sens.com) and Richard L. Freyman (Commun. Disord., Univ. of Massachusetts, Amherst, MA)

The precedence effect is the observation that, when a sound is repeated after a delay of a few milliseconds, the interaural cues of the leading sound have greater influence on localization and lateralization judgments of the composite sound than do the cues of the lagging sound. Classic studies of the precedence effect with transient stimuli have suggested auditory mechanisms that emphasize earlier-arriving cues. More recent studies have shown, however, that non-transient sounds—steady, long-duration noises with slow onset—also give rise to the precedence effect. This *ongoing* precedence effect is difficult to explain with the abrupt-onset mechanisms invoked to explain the effect with transients. The present model of the ongoing effect assumes that listeners use interaural cues within auditory bands to form band-specific lateral positions, which are then integrated across bands. Within-band positions are estimated from the interaural level difference and the interaural phase during the rising side of bandpass envelope modulations. Cross-band integration includes a mechanism that favors cues that are consistent across wider frequency regions. Predictions from the model will be compared to published data. [Work supported by NIDCD R01 01625.]

10:40

5aPPb8. Improving interaural time difference sensitivity using short interpulse intervals with vowel-like stimuli in bilateral cochlear implants. Sridhar Srinivasan (Acoust. Res. Inst., Austrian Acad. of Sci., Wohllebengasse 12-14/1, Wien, Wien 1040, Austria, ssri.oeaw@gmail.com), Bernhard Laback (Acoust. Res. Inst., Austrian Acad. of Sci., Vienna, Austria), and Piotr Majdak (Acoust. Res. Inst., Austrian Acad. of Sci., Wien, Austria)

Interaural time differences (ITDs) in the signal are important for sound localization in the lateral dimension. However, even under laboratory stimulus control, ITD sensitivity of cochlear-implant (CI) listeners is poor at pulse rates commonly used for encoding speech. Recently, improvements in ITD sensitivity were shown for unmodulated high-rate pulse trains with extra pulses at short interpulse intervals (SIPIs). In this study, we extended this approach to more realistic stimuli, i.e., high-rate (1000 pulses-per-second) pulse trains with vowel-like temporal envelopes. Using fixed SIPI parameters derived from the preceding study, we independently varied the timing of the extra pulses across the fundamental frequency (F0) period, the modulation depth (0.1, 0.3, 0.5, 0.7, and 0.9), and the F0 frequency (125 and 250 Hz). Our results show largest improvements in ITD sensitivity for SIPIs at the rising and peak portions of the F0 period and for larger modulation depths. These findings may be useful for enhancing sound localization cues with bilateral CI strategies.

11:00

5aPPb9. Interactive simulation and free-field auralization of acoustic space with the rtSOFE. Bernhard U. Seeber and Samuel W. Clapp (Audio Information Processing, Technische Universität München, Arcisstrasse 21, Munich 80333, Germany, seeber@tum.de)

The Simulated Open Field Environment (SOFE), a loudspeaker setup in an anechoic chamber to render sound sources along with their simulated, spatialized reflections, has been used for more than two decades in free-field hearing research. In 2004, the concept was revised to incorporate room-acoustic simulation software that computes sound reflections in arbitrarily-shaped rooms and auralizes them via many loudspeakers—the principle of various systems used today (Hafter and Seeber, ICA 2004). For a complete redesign of the system, an anechoic chamber has been purpose-built at TUM and I will talk about its specifications. The anechoic chamber hosts the real-time SOFE (rtSOFE), a setup with 61 loudspeakers to create a spatial sound field in a 5 m x 5 m area along with 360° of visual 3D projection. New room-acoustic simulation software for interactive computation of reflections computes room impulse responses in sub-millisecond intervals and updates a convolution system capable of convolving seconds-long impulse responses for many independent loudspeaker channels with very short latency. I will present the general concept and capabilities of the new rtSOFE, give details about its implementation and first experimental results. The rtSOFE in the new anechoic chamber at TUM forms a cutting edge research facility for interactive psychoacoustic and audio-visual research in virtual acoustic space.

11:20

5aPPb10. Update on sound quality assessment with TWO!EARS. Alexander Raake, Janto Skowronek, Hagen Wierstorf (Inst. of Media Technol., Audiovisual Technol. Group, Tech. Univ. Ilmenau, Helmholtzplatz 2, Ilmenau 98693, Germany, alexander.raake@tu-ilmenau.de), and Christoph Hold (Assessment of IP-based Applications, Tech. Univ. Berlin, Berlin, Germany)

The paper summarizes the different test and modeling campaigns carried out in the EC-funded FET-Open project TWO!EARS (www.twoears.eu) for sound quality and Quality of Experience (QoE) evaluation of spatial audio reproduction technology like stereophony or Wave-field Synthesis (WFS). This work represents one of the two proof-of-concept application domains of the interactive listening model developed in TWO!EARS. One stream of our sound-quality-related work focused on listening tests and model development for the individual sound quality features *localization* and *coloration*. After briefly reviewing the modeling approaches for these individual features presented in more depth elsewhere, the paper presents data and modeling considerations for a set of pairwise preference listening tests, following a dedicated audio mixing and reproduction paradigm. For subsequent model development, the results are analyzed in different ways, for example in terms of the pairwise preference data directly, using the Bradley-Terry-Luce model and using multidimensional analysis techniques. Based on these analyses, different modeling approaches based on the TWO!EARS framework are presented. To conclude, considerations are provided on how multimodal interaction can affect preference selections, based on an additional test on the selection of the "sweet spot" in a spatial audio listening context.

11:40

5aPPb11. Specificity of adaptation to non-individualized head-related transfer functions. Griffin D. Romigh (Air Force Res. Labs, 2610 Seventh St., Area B, Bldg. 441, Wright Patterson AFB, OH 45433, griffin.romigh@us.af.mil), Brian Simpson (Air Force Res. Labs, Wright-Patterson AFB, OH), and Michelle Wang (Air Force Res. Labs, Dayton, OH)

Initial accuracy is poor when listeners are asked to localize virtual sound sources that have been low-pass filtered or rendered with non-individualized head-related transfer functions (HRTFs). However, Majdak *et al.* (2013) showed that, with training, localization performance improved when virtual sounds were rendered using HRTFs that were low-pass filtered at 8.5 kHz. This result suggests that previous outcomes showing training-induced improvement in localization performance with non-individualized HRTFs may merely be the result of listeners attending to low-frequency spectral information that is consistent with their own HRTFs, and not attending to the information from high frequencies, where HRTFs differ widely across individuals. That being the case, one would expect training to generalize to all non-individualized HRTFs, not just the non-individualized HRTF used during training. The current study investigated this hypothesis by performing tests of localization accuracy before and after three weeks of auditory localization training with a single non-individualized HRTF. For all subjects, localization performance with non-individualized HRTFs improved to a level at or near their performance with individualized HRTFs, and no generalization to the other non-trained HRTFs was found, suggesting subjects do learn to utilize an alternative set of high-frequency spectral information.

12:00

5aPPb12. Age-related cortical changes in spatial auditory attention. Erol J. Ozmeral, Madeleine Berg, David A. Eddins, and Ann C. Eddins (Commun. Sci. and Disord., Univ. of South Florida, 3802 Spectrum Blvd., Ste. 210, Tampa, FL 33612, eozmeral@usf.edu)

Along with established effects of age on hearing sensitivity, there is a growing body of evidence that the aging auditory system suffers from reduced temporal resolution as well. This, in combination with changes in attentional-resource allocation, could have profound effects on the ability for older listeners to selectively attend to spatial locations—a key component to successful listening and communication in challenging auditory environments. Because behavioral tasks rarely have an unattended comparison and electrophysiological tasks rarely have an attended comparison, it is difficult to ascertain the extent to which selective attention mediates or sharpens spatial tuning. To address this shortcoming, we measured cortical responses using electroencephalography for moving stimuli in the free field during both passive and active conditions. Active conditions required listeners to respond to the onset of a stimulus when it occurred at a specific location (either 30° to the left or right of center). Both younger and older normal-hearing listeners participated in the study. The event-related potentials as well as the source-localized activity in regions of interest associated with sensory processing (i.e., left and right auditory cortices) and top-down control (i.e., dorsal fronto-parietal areas) revealed considerable morphological differences between the age groups.

Session 5aSAa

Structural Acoustics and Vibration and Physical Acoustics: Numerical Methods and Benchmarking in Computational Acoustics I

Robert M. Koch, Cochair
Chief Technology Office, Naval Undersea Warfare Center, Code 1176 Howell Street, Bldg. 1346/4, Code 01CTO, Newport, RI 02841-1708

Micah R. Shepherd, Cochair
Applied Research Lab, Penn State University, PO Box 30, mailstop 3220B, State College, PA 16801

Manfred Kaltenbacher, Cochair
Mechanics and Mechatronics, TU Wien, Getreidemarkt 9, Wien 1060, Austria

Steffen Marburg, Cochair
Faculty of Mechanical Engineering, Technical University of Munich, Boltzmannstr. 15, Muenchen 85748, Germany

Invited Papers

8:00

5aSAa1. Benchmark problem identifying a pollution effect in boundary element method. Steffen Marburg (Faculty of Mech. Eng., Tech. Univ. of Munich, Boltzmannstr. 15, Muenchen 85748, Germany, steffen.marburg@tum.de)

In his former contributions on the boundary element method (BEM) in acoustics, the authors had not found any indication of a dispersion error similar to what is known as the pollution effect for the finite element method (FEM). However, in a recent paper, the author has demonstrated the effect of numerical damping in BEM. A consequence of the pollution effect of FEM and numerical damping in BEM is that the common rule of choosing a fixed number of elements per wavelength is not valid. By using one of the benchmark cases of the EAA Technical Committee for Computational Acoustics, this can be easily shown. Traveling waves in a long duct are decaying. In this presentation it will be shown that the numeric error depends on the length of the duct. For models with many waves over its surface, numerical damping adds an additional numeric error which can be understood as a pollution error because a local refinement in certain regions of the model will not significantly decrease the error. It will be discussed in which cases this problem may become relevant for practical use of BEM.

8:20

5aSAa2. Investigation of the flow- and sound-field of a low-pressure axial fan benchmark case using experimental and numerical methods. Florian Zenger (Inst. of Process Machinery and Systems Eng., Friedrich-Alexander Univ. Erlangen-Nürnberg, Cauerstr. 4, Erlangen 91058, Germany, ze@ipat.uni-erlangen.de), Clemens Junger (Inst. of Mech. and Mechatronics, TU Vienna, Vienna, Austria), Manfred Kaltenbacher (Inst. of Mech. and Mechatronics, TU Vienna, Wien, Austria), and Stefan Becker (Inst. of Process Machinery and Systems Eng., Friedrich-Alexander Univ. Erlangen-Nürnberg, Erlangen, Bavaria, Germany)

An extension of a benchmark case for a low-pressure axial fan is presented. The generic fan is a typical fan to be used in commercial applications. The fan design procedure, as well as the experimental setups are described in detail. The numerical approach is based on a forward coupling between a flow simulation with ANSYS Fluent and an aeroacoustic source term and wave propagation computation with multiphysics research software CFS++. Experimental and numerical data for aerodynamics and aeroacoustics are compared. This includes aerodynamic performance (volume flow rate, pressure rise and efficiency), fluid mechanical quantities on the fan suction and pressure side (velocity distribution and turbulent kinetic energy), wall pressure fluctuations in the gap region and acoustic spectra at various microphone positions. Finally, a comprehensive data base of an axial fan was generated. Flow field properties at the fan suction and pressure side from the CFD simulation are in good agreement and spectra from the wall pressure fluctuations are in excellent agreement with the experimental data. Spectra from the computed acoustic pressure tend to slightly overestimate the experimental results. Based on the good agreement of both aerodynamic and aeroacoustic data, a thorough study on the dominant sound generation mechanisms is made.

8:40

5aSAa3. Non-conforming finite element method for efficiently computing multilayered microperforated plate absorbers. Manfred Kaltenbacher, Sebastian Floss, and Jochen Metzger (Mech. and Mechatronics, TU Wien, Getreidemarkt 9, Wien 1060, Austria, manfred.kaltenbacher@tuwien.ac.at)

The concept of acoustic impedance is a very useful model approach to efficiently compute configurations for sound absorption. Thereby, the measurements are performed by an impedance tube and the obtained data is used for the first order Robin (impedance) boundary condition within the numerical simulation. However, this approach is only valid for sound incidence perpendicular to the boundary. A second approach is to resolve the volume of the absorber and use a rigid or even elastic frame model. Especially for multi-layered silencers based on microperforated plates (MPPs) the volume resolving approach is beneficial. Here, a main challenge is to cope with the quite different mesh sizes needed for accurately resolving the waves in the MPPs and the surrounding air regions. To efficiently simulate such designs, we apply a Nitsche-type mortaring within the Finite Element Method to allow for non-conforming meshes and thereby directly connect the different mesh sizes in the MPPs and surrounding air regions. We will discuss in detail our absorber design, the performed measurements and numerical simulations and plan to publish the complete setup and results as a benchmark case for computational acoustics.

Contributed Papers

9:00

5aSAa4. Quantification of numerical damping in the acoustic boundary element method for the example of a traveling wave in a duct. Suhaib K. Baydoun and Steffen Marburg (Chair of VibroAcoust. of Vehicles and Machines, Tech. Univ. of Munich, Boltzmannstraße 15, Garching bei München 85748, Germany, suhaib.baydoun@tum.de)

The boundary element method (BEM) is a popular numerical method for solving linear time-harmonic acoustic problems. Using the BEM, modeling is limited to the boundary of the fluid domain, which is particularly advantageous for exterior problems with unbounded domains. A widely unknown drawback of the acoustic BEM is numerical damping. This work is concerned with numerical damping encountered in the benchmark problem of an air-filled duct with rigid walls. A traveling wave, induced by a particle velocity at the inlet, is fully absorbed at the outlet of the duct by imposing an impedance boundary condition. The exact solution gives a constant pressure amplitude over the entire frequency range. However, the numerical solution exhibits decay of the pressure amplitude, which is clearly an indication for numerical damping. This phenomenon is studied for different frequencies and elements-per-wavelength ratios. The extent of numerical damping is quantified by relating the decay to the pressure distribution obtained for a fluid model considering damping. The gained knowledge enables more accurate estimations of real damping phenomena in the future.

9:20

5aSAa5. A brief review, benchmark, future development of structural acoustic tools. Kuangcheng Wu (Ship Signatures, Naval Surface Warfare Ctr. - Carderock, 9500 MacArthur Blvd., West Bethesda, MD 20817, kcwu@msn.com)

Numerical tools have been widely applied to real world applications for noise and vibration control. Each tool has its own advantages and limitations. To properly use the tool in the correct environment, it is imperative to understand its underlying physics. One of the main challenges in modeling a marine structure is to correctly simulate its unbounded surrounding domain. Analytic solutions are limited to certain geometries (Junger & Feit, Sound, Structure, and Their Interaction, ASA, 1993), semi-analytic solution (i.e., Surface Variational Principle), and different numerical techniques (i.e., boundary element, infinite element, and perfectly matching layer) has been used to model complex structures with unbounded fluid implicitly or explicitly while satisfying the Sommerfeld radiation BC (Pierce, Acoustics: An Introduction to its Physical Principles and Applications, ASA, 1989). In this paper, those numerical techniques will be briefly reviewed and several benchmark cases for simple structures will be presented. With the understanding of underlying physics, ideas in speeding up the numerical analysis by proper simplification or model reduction will be addressed.

9:40

5aSAa6. Comparison of three-dimensional acoustical Green's functions for a half-space BEM formulation. Martin A. Ochmann (FB II, Beuth Hochschule fuer Technik Berlin, Luxemburger Strasse 10, Berlin D-13353, Germany, ochmann@beuth-hochschule.de)

We consider three-dimensional acoustical scattering or radiation problems in frequency domain above an infinite flat plane equipped with a local impedance condition. For such half-space problems, a BEM formulation is of advantage, where the Green's function used satisfies not only the Helmholtz equation, but also the boundary condition at the impedance plane. When using such a tailored Green's function, only the surface of the sound radiating or scattering object has to be discretized, since the influence of the impedance ground is automatically taken into account. Simple formulas for such half-space Green's function exist only for rigid or soft infinite planes. For a ground with arbitrary surface impedances, many different expressions of the Green's function are given in the literature. For incorporating these functions into a BEM formulation, the first and second normal derivatives must be calculated in such a way that they do not possess strong singularities. In the present work, four representations of half-space Green's functions are investigated and compared for three different kinds of surface impedances with respect to accuracy and computing time: (1) for a pure absorbing ground, i.e., with real impedance, (2) for a masslike, and (3) for a springlike ground corresponding to pure imaginary impedances.

10:00–20 Break

10:20

5aSAa7. Atmospheric acoustic modes between a complex ground impedance and an artificial absorber. Richard B. Evans (Retired, 99F Hugo Rd., N. Stonington, CT 06359, richard.evans.01@snet.net), Xiao Di, and Kenneth E. Gilbert (National Ctr. for Physical Acoust., University, MS)

Atmospheric acoustic normal mode computer codes are faced with finding the complex modal eigenvalues. Searching in the complex plane is difficult and requires special numerical techniques and custom software. A Legendre-Galerkin technique that reduces the problem to a complex matrix eigenvalue problem can be solved by commercially available software. This proposed Legendre-Galerkin method is described as the projection of the acoustic normal mode problem onto a recombined basis of Legendre polynomials. The modal approach is best suited for providing benchmark quality results in cases when guided modes dominate the problem. Such results are useful in establishing the validity and interpreting the characteristics of atmospheric acoustic fields computed with the parabolic equation method, for the same problem. The Legendre-Galerkin method is applied to cases with a ground based duct and an elevated duct. Measured wind speeds, from a costal experiment, provide the effective downwind and upwind sound speed profiles with these ducted characteristics.

Invited Paper

10:40

5aSAa8. Comparison of finite element and analytical modeling of scattering of an acoustic wave by particles in a fluid. Valerie J. Pinfield, Derek M. Forrester (Chemical Eng. Dept., Loughborough Univ.,Loughborough LE11 3TU, United Kingdom, v.pinfield@lboro.ac.uk), Artur L. Gower, William J. Parnell, and Ian D. Abrahams (School of Mathematics, Univ. of Manchester, Manchester, United Kingdom)

Ultrasonic wave propagation through dispersions of particles in liquids is of interest for particle characterization and process monitoring applications. Interpretation of the measurements relies on a theoretical model; we typically use a multiple scattering model which builds on models of scattering by independent particles. We report finite element modeling of an acoustic wave propagating through a liquid and interacting with a particle, using the linearized thermo-acoustic equations for propagation in a viscous liquid. We demonstrate that the interaction of the acoustic field with the particle leads to decaying thermal and shear wave fields in the region very close to the particle. Since the length scale of the thermal and shear decay is orders of magnitude smaller than the propagational mode acoustic wavelength, fine meshing is necessary in the region of the particle/fluid boundary. The simulation results are compared with analytical solutions for scattering of a plane wave by a single spherical particle, provided by Epstein and Carhart [JASA, 25, 533, (1953)] and Allegra and Hawley [(JASA, 51, 1546 (1972)].

Contributed Paper

11:00

5aSAa9. A coupled isogeometric finite element and boundary element method with subdivision surfaces for structural -acoustic analysis of shell structures. Zhaowei Liu, Robert Simpson (School of Eng., Univ. of Glasgow, Glasgow G12 8QQ, United Kingdom, z.liu.2@research.gla.ac.uk), Fehmi Cirak, and Musabbir Majeed (Dept. of Eng., Univ. of Cambridge, Cambridge, United Kingdom)

We demonstrate a method for simulating medium-wave acoustic scattering over elastic thin shell structures. We propose a coupled approach whereby the finite element formulation is used to describe the dynamic structural response of the shell and the boundary element method models the acoustic pressure within the infinite acoustic domain. The two methods are coupled through the relationship between acoustic velocities on the structural-fluid interface. In our approach, a conforming subdivision discretization is generated in Computer Aided Design (CAD) software which can be used directly for analysis in keeping with the idea of isogeometric analysis whereby a common geometry and analysis model is adopted. The subdivision discretization provides C1 surface continuity which satisfies the challenging continuity requirements of Kirchhoff-Love shell theory. The new method can significantly reduce the number of elements required per wavelength to gain same accuracy as an equivalent Lagrangian discretization, but the main benefit of our approach is the ability to handle arbitrarily complex geometries with smooth limit surfaces directly from CAD software. Our implementation make use of H-matrices to accelerate dense matrix computations and through this approach, we demonstrate the ability of our method to handle high-fidelity models with smooth surfaces for structural-acoustic analysis.

Invited Papers

11:20

5aSAa10. Computing head related impulse responses and transfer functions using time domain equivalent sources. John B. Fahnline (ARL / Penn State, P.O. Box 30, State College, PA 16804-0030, jbf103@arl.psu.edu)

In the past, head-related impulse responses (HRIR) and head-related transfer functions (HRTF) have primarily been computed using frequency domain boundary element methods or finite-difference time domain methods. The possibility of computing HRIRs and HRTFs using transient equivalent sources is examined using a lumped parameter technique for enforcing the specified boundary condition. It is demonstrated that performing the computations in the time domain is advantageous because only a few thousand time steps are need to fully define the HRIRs and nonuniform meshes can be used to reduce the number of acoustic variables drastically without significantly degrading the solution accuracy. It is also shown that the computations adapt well to parallel processing environments and the times associated with the equivalent source calculations are proportional to the number of processors.

11:40

5aSAa11. A new infinite element paradigm in computational structural acoustics? David S. Burnett (Naval Surface Warfare Ctr., 110 Vernon Ave., Panama City, FL 32407, david.s.burnett@navy.mil) and Les H. Wigdor (Syslink Consulting LLC, Beacon, NY)

In the 1990s, the lead author developed a radical new formulation for infinite elements for modeling scattering and radiation from structures in unbounded domains. It was shown to be faster than the popular boundary element method (BEM), for the same physics to the same accuracy, by several orders of magnitude; the speedup is unbounded as problem size increases. Academia and industry called it a “revolution” in computational acoustics that would probably bring an end to the BEM. But then Bell Labs patented and licensed the elements, effectively ending the “revolution” and removing the technology from the public domain for the next 20 years. Now, in 2017, some patents have expired and the rest will expire soon, thus restoring the technology to the public domain. The talk will review the original technology and then describe new R&D since 2015: (i) speeding up a commercial acoustic scattering code by 1400x and (ii) extending the technology by developing a new hybrid version that computes the external field over 12,000x faster than the traditional, expensive Helmholtz integral. Now that this “revolutionary” technology is back in the public domain, the market place can finally decide whether it constitutes a new paradigm in computational structural acoustics.

Contributed Paper

12:00

5aSAa12. Acoustic radiation modes and normal modes in exterior acoustic problems. Lennart Moheit and Steffen Marburg (Chair of VibroAcoust. of Vehicles and Machines, Tech. Univ. of Munich, Boltzmannstr. 15, Garching b. München, Bavaria 85748, Germany, lennart.moheit@tum.de)

Acoustic radiation modes are eigenvectors of the real and symmetric acoustic impedance matrix Z, which is usually computed by the boundary element method (BEM) matrices G and H at the surfaces of inner obstacles in an unbounded fluid-filled domain. Application of the finite element method (FEM) and the infinite element method (IFEM) allows the computation of the acoustic radiation modes as well, but also normal modes can be computed as right eigenvectors of a state-space eigenvalue problem. Modal superposition of both radiation modes and normal modes leads to accurate results of the radiated sound power. However, normal modes additionally provide modal sound pressure distributions in the whole computational domain and can therefore be used to calculate frequency response functions. In this work, modal superposition and reduction in exterior acoustics are presented and discussed.

THURSDAY MORNING, 29 JUNE 2017 — ROOM 204, 10:40 A.M. TO 12:00 NOON

Session 5aSAb

Structural Acoustics and Vibration, Noise, Physical Acoustics, and Architectural Acoustics: Acoustics and Vibration of Sports and Sports Equipment

Daniel A. Russell, Chair

Graduate Program in Acoustics, Pennsylvania State University, 201 Applied Science Bldg., University Park, PA 16802

Contributed Paper

10:40

5aSAb1. How runners deal with the shock induced vibrations propagating through their body? Delphine Chadefaux, Eric Berton, and Guillaume Rao (Aix Marseille Univ, CNRS, ISM, Inst Movement Sci., 163 Ave. de Luminy, Marseille 13009, France, delphine.chadefaux@univ-amu.fr)

Runners experience numerous shocks leading to vibrations propagating from the foot toward the entire body. These repetitive shocks are related to musculoskeletal injuries. Consequently, runners tend to adapt their running patterns according to the ground surface to cushion the impact. However, the way runners manage precisely the three-dimensional components of the vibrations, especially in the frequency domain, is not well understood. The present study investigated which biomechanical parameters runners adapt to tune the shock induced vibrations according to different running conditions. A specific experimental procedure was designed, based on simultaneously collecting kinematic, dynamic, vibration, and electromyographic data during running barefoot or shod and at various velocities. Using 10 non-specialist runners, energetic and spectral analyses of the three-dimensional foot impact induced vibrations occurring at the third metatarsal bone, the tibial plateau, the knee joint, the hip joint, and the 7th cervical were performed. Results outlined the transfer function of each investigated segment. A significant outcome is the strategy set up by the neuro-musculoskeletal system to protect upper areas of the human body. This contribution opens up new perspectives in running analyses by underlining the significance of the three-dimensional and the spectral contents in the shock induced vibrations.

Invited Paper

11:00

5aSAb2. Hitting the ball on the meat—Finding the sweet spot of a hurley. Eoin A. King and Robert Celmer (Acoust. Program and Lab, Univ. of Hartford, 200 Bloomfield Ave., West Hartford, CT 06117, eoking@hartford.edu)

Hurling is one of Ireland's most popular indigenous sports. It is a Gaelic stick-and-ball sport that combines elements of field hockey, lacrosse, and handball. Key to the game is a player's mastery of the hurling stick (hurley) which is used to the strike the ball (sliotar). The hurley is similar to a hockey stick but with a shorter, wider and more circular head (bás), which is the area of the hurley that strikes the sliotar. The sweet spot is a general term used amongst players to indicate the correct position on the hurley to strike the ball ("hitting the ball on the meat"). By measuring the moment of inertia and center of percussion of a hurley, combined with experimental modal analysis, this paper attempts to define the location of a sweet spot on a 34 inch ash huley. Measurements are based on the ASTM standard test method for measuring the moment of inertia and center of percussion of a baseball bat; however, we also propose an alternative method to this standard for determining the center of percussion of a bat.

Contributed Papers

11:20

5aSAb3. Dynamical analysis of stroke induced vibrations in tennis racket. Delphine Chadefaux, Guillaume Rao (Aix Marseille Univ, CNRS, ISM, Inst. Movement Sci., 163 Ave. de Luminy, Marseille 13009, France, delphine.chadefaux@univ-amu.fr), Jean-Loïc LE CARROU (Sorbonne Universités, UPMC Univ Paris 06, CNRS, UMR 7190, LAM - Institut Jean le Rond d'Alembert, Paris, France), Eric Berton, and Laurent Vigouroux (Aix Marseille Univ, CNRS, ISM, Inst Movement Sci, Marseille, France)

Tennis rackets are mostly designed disregarding the boundary condition managed by the player's hand on the handle. This process leads to a lack of accuracy in the mechanical parameters the manufacturers provide to their rackets in order for them to be reliable and comfortable to the player. Our work aimed at providing a better understanding of the effect of the tennis player's hand on the racket's dynamical behavior. For this purpose, a dedicated experimental procedure involving 14 tennis players and 5 tennis rackets has been carried out. Vibrations propagated from the racket toward the upper-limb have been collected synchronously with kinematic and electromyographic data during forehands of various intensities. Additionally, an analytical model of the hand/racket interaction has been designed based on operational modal analyses. This model provides a straightforward tool to predict changes in the dynamical behavior of a tennis racket under playing conditions. Results indicated that tennis players adjust their grip-force to tune the vibrational content entering into his upper-limb. Besides, a noteworthy outcome is that grip-force induces modifications in the racket's dynamical behavior that are at least as important as the differences observed under free boundary conditions due to the rackets' own mechanical parameters.

11:40

5aSAb4. Vibroacoustic analysis of table tennis rackets and balls: The acoustics of ping pong. Daniel A. Russell (Graduate Program in Acoust., Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, drussell@engr.psu.edu)

Table Tennis rackets (ping-pong paddles) exhibit a large number of structural vibration modes which follow patterns first observed by Ernst Chladni and Mary Waller for elliptical plates. Vibrational mode shapes and frequencies obtained through experimental modal analysis will be shown. Acoustic analysis reveals that one structural mode of the paddle, in particular, dominates the sound produced by the ball-paddle impact. The rubber padding provides some damping, and a significant mass loading to the paddle vibrations. The hollow cellulose nitrate balls exhibit a number of vibrational mode shapes typical of a hollow spherical shell, starting at frequencies around 5900 Hz; these will be demonstrated from experimental and computational results. However, the contact time between ball and paddle is such that the lowest acoustic modes of the ball do not contribute to the radiated sound. Instead, the ball appears to radiate sound at a much higher frequency sound (10-12 kHz) most likely due to snap-through buckling common to spherical shells undergoing deformation while impacting a rigid surface at high speeds.

THURSDAY MORNING, 29 JUNE 2017 BALLROOM A, 8:00 A.M. TO 12:20 P.M.

Session 5aSC

Speech Communication: Variation: Age, Gender, Dialect, and Style (Poster Session)

Elizabeth D. Casserly, Chair
Dept. of Psychology, Trinity College, 300 Summit St., Hartford, CT 06106

All posters will be on display from 8:00 a.m. to 12:20 p.m. To allow contributors in this session to see the other posters, authors of odd-numbered papers will be at their posters from 8:00 a.m. to 10:10 a.m. and authors of even-numbered papers will be at their posters from 10:10 a.m. to 12:20 p.m.

Contributed Papers

5aSC1. Acoustic cues and linguistic experience as factors in regional dialect classification. Steven Alcorn, Kirsten Meemann (The Univ. of Texas at Austin, 150 W. 21st St., Stop B3700, Austin, TX 78712, steven.alcorn@utexas.edu), Erin Walpole, Cynthia G. Clopper (The Ohio State Univ., Columbus, OH), and Rajka Smiljanic (The Univ. of Texas at Austin, Austin, TX)

Listeners rely on a variety of acoustic cues when identifying regional dialects, including segmental, prosodic, and temporal features of speech. The purpose of this study was to examine how native speakers of American English (AE) classify AE talkers by regional dialect when segmental and prosodic features are manipulated in the stimuli they hear; it also considered experience with different regional dialects as an additional factor affecting classification. Native AE listeners residing in Ohio and Texas completed a free classification task in which they heard the same sentence read by 60 monolingual AE talkers and grouped talkers together based on perceived regional similarities. Three versions of the stimuli were presented in a between-subjects design: unaltered, monotonized (f0 flattened to remove intonation cues), or low-pass filtered (to remove segmental cues). Preliminary analyses indicate that performance in the unaltered and monotone conditions was more accurate overall than in the low-pass filtered condition, suggesting that listeners rely on segmental information more than prosodic information for classification. Overall performance was similar across the Ohio and Texas listener groups for all three conditions, but Ohioans outperformed Texans at grouping talkers from the local Midland dialect together, providing preliminary evidence for an effect of experience on classification.

5aSC2. Applying pattern recognition to formant trajectories: A useful tool for understanding African American English (AAE) dialect variation. Meisam K. Arjmandi, Laura Dilley, and Zachary Ireland (Dept. of Communicative Sci. and Disord., Michigan State Univ., 1026 Red Cedar Rd., East Lansing, MI, khalilar@msu.edu)

Few studies have focused on the acoustic-phonetic characteristics of African American English (AAE) which distinguish this dialect from Standard American English (SAE), particularly for vowels and sonorant consonants. This study investigated whether formant dynamics from short, sonorant portions of speech are sufficient to distinguish AAE and SAE dialects. Seven female speakers, four SAE and three AAE, from the Lansing, Michigan area, were selected from a corpus of 30-45 minute sociolinguistic interviews. Target portions of speech consisting of a V or VC sequence (C = /n/, /m/, /l/, /r/) were identified from contexts selected to control for coarticulation. First (F1) and second (F2) formant values were extracted from randomly selected tokens at points 19%, 56%, and 81% of the duration through the demarcated speech portions. Pattern recognition techniques were examined to differentiate tokens of the two dialects based on formant trajectories as feature vectors. The results revealed that formant dynamics of the selected contexts are acoustically informative enough to differentiate groups of SAE from AAE speakers. A near-perfect classification of some contexts was also achieved by applying support vector machines to the formant trajectories. These findings highlight the usefulness of incorporating pattern recognition techniques for understanding acoustic variation due to dialect.

5aSC3. Articulatory kinematics during connected speech across dialects and dysarthria. Jeffrey J. Berry (Speech Pathol. & Audiol., Marquette Univ., P.O. Box 1881, Milwaukee, WI 53201-1881, jeffrey.berry@marquette.edu), Yunjung Kim (Commun. Sci. and Disord., Louisiana State Univ., Baton Rouge, LA), and James Schroeder (Elec. and Comput. Eng., Marquette Univ., Milwaukee, WI)

The current work presents an analysis of articulatory kinematics during connected speech in typical talkers and talkers with dysarthria from two different dialects of American English. Instrumental methods for obtaining articulatory kinematic data during speech (particularly electromagnetic articulography) are becoming increasingly viable within the clinical setting. Yet almost no existing clinical standards for collecting and interpreting articulatory kinematic data have been established. Moreover, there is little basis for differentiating the impact of dialect from dysarthria on articulatory kinematics. We examine articulatory kinematics obtained via electromagnetic articulography during a standard connected speech passage read by typical talkers ($n = 30$) and talkers with dysarthria ($n = 15$). Participants are divided among upper Midwestern and Southern American English dialects. Analyses focus on kinematic measures of articulatory movement (range-of-motion, speed, acceleration, and jerk) within and across dialect groups and between typical talkers and individuals with dysarthria. The goal of the current work is to provide a preliminary evaluation of whether different kinematic measures of articulatory movement during connected speech may be differentially sensitive to the impact of dialect and dysarthria. The results of this work are germane to establishing clinically relevant measures of articulatory kinematics to improve the clinical assessment of dysarthria.

5aSC4. Dialect classification reveals mismatch between speech processing and dialect perception. Megan Dailey and Cynthia G. Clopper (Ohio State Univ., 1961 Tuttle Park Pl., 108A Ohio Stadium East, Columbus, OH 43210, clopper.1@osu.edu)

Familiar dialects can facilitate speech processing. However, recent investigations of speech processing of the Northern and Midland dialects of American English reveal a different pattern: in noise-masked speech, listeners from both dialect regions identify Midland words and phrases with higher accuracy than Northern words and phrases. This preference may be explained by inconsistencies between Northern talkers' production and perception of their own dialect. The goal of the current study was to determine whether cross-dialect processing differences between the Northern and Midland dialects reflect listeners' explicit dialect identification ability. Participants completed a speech intelligibility in noise task followed by a forced-choice dialect categorization task. Speech stimuli in both tasks were short phrases taken from passages read by eight Northern and eight Midland talkers. Responses in both tasks were scored for accuracy. Results revealed higher accuracy in intelligibility for Midland phrases than Northern phrases, as in previous work, but poor dialect categorization performance across all listeners. The inability of listeners to explicitly categorize talkers by dialect while showing an intelligibility benefit for Midland forms indicates that the observed cross-dialect processing differences emerge even in the absence of explicit dialect categorization, revealing a perceptual mismatch between speech processing and dialect perception.

5aSC5. Sociophonetic variation in Mississippi: Gender, ethnicity, and prevoiced plosives. Wendy Herd (MS State Univ., 2004 Lee Hall, Drawer E, MS State, MS 39762, wherd@english.msstate.edu)

While native English speakers are traditionally reported to produce word-initial voiced plosives with short positive VOTs, recent studies suggest sociophonetic variation exists in the production of these sounds. In separate studies, more prevoicing has been reported for men than women, for African American speakers than Caucasian American speakers, and for southern American English speakers than speakers from other regions. The current study investigates the effects of gender, ethnicity, and context on voicing variation in Mississippi by analyzing word-initial /b, d, g/ as read in sentences by forty native speakers of English grouped according to self-reported gender and ethnicity. A significant effect of ethnicity and an interaction between gender and ethnicity were found. African American speakers produced voiced stops with a larger proportion of closure voicing and produced more fully voiced closures than Caucasian American speakers. While African American men and women produced similarly voiced closures, Caucasian American men voiced closures more than women. Similarly, Caucasian American speakers' closure voicing was affected by context (e.g., following an approximant vs. following a plosive), but African American speakers consistently produced voiced closures regardless of context. These findings strongly suggest that dialectal differences play a role in the voicing variation of word-initial voiced stops.

5aSC6. Assessing vowel space area metrics in the context of dialect variation. Ewa Jacewicz and Robert A. Fox (Dept. and Speech and Hearing Sci., The Ohio State Univ., 1070 Carmack Rd., 110 Pressey Hall, Columbus, OH 43210, jacewicz.1@osu.edu)

There has been a growing interest in the development of a sensitive methodology to define a working vowel space area (VSA) as a metric in basic and clinical research applications. In this study, three approaches to the assessment of VSA were tested to evaluate their efficacy in characterizing cross-dialectal and cross-generational variation: The traditional vowel quadrilateral, the traditional convex hull, and a more liberal convex hull. In the two traditional approaches, VSA was computed as a planar convex polygon shaped by phonologically distinct vowel categories as its corners. The mean F1/F2 values at vowel midpoint were used to define the respective areas. The liberal convex hull utilized vowel dynamics and variable F1/F2 temporal locations to refine the outer boundaries and maximize the VSA. This approach used an unrestricted number of vowels and measurement locations to define the optimal VSA. All computations were based on a common speech material produced by 135 female speakers representing three American English dialects and four generations ranging from 8 to 91-year olds. The three metrics yielded inconsistent and contradictory estimates of VSA. Discussion will focus on the limited utility of polygon geometry in characterizing a working VSA in American English dialects.

5aSC7. "He maybe did" or "He may be dead"? The use of acoustic and social cues in applying perceptual learning of a new dialect. Rachael Tatman (Linguist, Univ. of Washington, Guggenheim Hall, 3940-2425 Benton Ln, Seattle, WA 98195, rctatman@uw.edu)

When learning to recognize words a new dialect, listeners rely on both acoustics (Norris, McQueen, & Cutler 2003) and extra-linguistic social cues (Kraljic, Brennan, & Samuel 2008). This study investigates how listeners use both acoustic and social information after exposure to a new dialect. American English (AmE) speaking listeners were trained to correctly identify the front vowels of New Zealand English (NZE). To an AmE speaker, these are highly confusable: "head" is often heard as "hid". Listeners were

then played 500 ms vowels produced by both AmE and NZE speakers. Half of the listeners were given correct information on the speakers' dialect, and half incorrect. Listeners' classifications of vowels were affected by what they were told about the speakers' dialect. Vowels labeled as a given dialect, correctly or not, were more likely to be classified as if they were from that dialect. There was also an effect of speakers' actual dialect. Overall, the AmE-speaking listeners were more accurate when identifying vowels from AmE than those from NZE; even with the very limited acoustic information available listeners are still sensitive to inter-dialectal differences. Any model of cross-dialect perception, then, must account for listener's use of both social and acoustic cues.

5aSC8. Transcription and forced alignment of the digital archive of southern speech. Margaret E. Renwick, Michael Olsen, Rachel M. Olsen, and Joseph A. Stanley (Linguist Program, Univ. of Georgia, 240 Gilbert Hall, Athens, GA 30602, mrenwick@uga.edu)

We describe transcription and forced alignment of the Digital Archive of Southern Speech (DASS), a project that will provide a large corpus of historical, semi-spontaneous Southern speech for acoustic analysis. 372 hours of recordings (64 interviews) comprise a subset of the Linguistic Atlas of the Gulf States, an extensive dialect study of 1121 speakers conducted across eight southern U.S. states from 1968 to 1983. Manual orthographic transcription of full DASS interviews is carried out according to in-house guidelines that ensure consistency across files and transcribers. Separate codes are used for the interviewee, interviewer, non-speech, overlapping, and unintelligible speech. Transcriber output is converted to Praat TextGrids using LaBB-CAT, a tool for maintaining large speech corpora. TextGrids containing only the interviewee's speech are generated, and subjected to forced alignment by DARLA, which accommodates the levels of variation and noise in the DASS files with a high degree of success. Toward acoustic analysis, we evaluate three methods for vowel formant extraction: the native output of DARLA, a local implementation of FAVE-Extract, and a Praat-based extractor that incorporates separate formant tracks for different regions of the vowel space. We present this workflow of transcription and analysis to benefit other projects of similar size and scope.

5aSC9. An acoustic perspective on legacy data: Vowels in the digital archive of Southern speech. Margaret E. Renwick and Joseph A. Stanley (Linguist Program, Univ. of Georgia, University of Georgia, 240 Gilbert Hall, Athens, GA 30602, mrenwick@uga.edu)

Speech varies widely in the American South, but the region is argued to share the Southern Vowel Shift, whose various characteristics include monophthongization of upgliding diphthongs, convergence of certain front vowels via raising and lowering, and back-vowel fronting. We investigate the influence of social factors on shift participation using vowel formant and duration data from the Digital Archive of Southern Speech (recorded 1968−1983), which is newly transcribed and segmented by forced alignment. With this corpus of 64 linguistic interviews (372 hours), we study how shifting varies in geographic space, across states from Texas to Florida. The interviews offer large amounts of data from individual speakers, and their semi-spontaneous nature reveals a more realistic portrait of phonetic variability than is typically available. Interviews of European- and African American speakers permit comparison of the Southern Vowel Shift with the African American Vowel Shift. The impacts of other factors on the vowel space are evaluated including generation, gender, socioeconomic status, and education level. Acoustic analysis of historical speech corpora offers perspective for modern sociophonetic studies, by providing a point of comparison to illuminate the development of modern regional variation, which will inform and enhance models of language change over time.

5aSC10. Implications of covert articulatory variation for several phonetic variables in Raleigh, North Carolina English. Jeff Mielke, Bridget Smith (English, North Carolina State Univ., 221 Tompkins Hall, Campus Box 8105, Raleigh, NC 27695-8105, jimielke@ncsu.edu), and Michael J. Fox (Sociology and Anthropology, North Carolina State Univ., Altoona, WI)

We examine several phonological variables in a spontaneous speech corpus and a lab-collected acoustic/articulatory dataset, in order to pursue the hypothesis that covert inter-speaker differences in speech production are instrumental to solving the actuation problem in language change. We examine two known cases of covert articulatory variation and their impact on a range of low-level and high-level sound patterns active in Raleigh, NC. The covert articulatory variables are /ɹ/ tongue shape and /l/ posterior constriction location. The overt variables are the retraction of /s/ and /z/ and the affrication of /t/ and /d/ in various contexts near /ɹ/ (within words and across word boundaries), flapping and deletion of /ɹ/ after /θ/, intrusive [ɫ] next to /θ/ and nasals, and the quality of vowels before and after /l/. We also examine affrication of /t/ and /d/ before /w/. The spontaneous speech comes from 132 hour-long interviews from the Raleigh Corpus (Dodsworth and Kohn 2012), and the lab speech is a set of 29 wordlist recordings exhibiting the overt variables under investigation. We report the observed relationships between the production of the covert and overt variables in the lab speech, and relate this to the distribution of variants in spontaneous speech.

5aSC11. Splitting of Arabic communal dialects at childhood: The case of consonant acquisition in Kfar Kanna Israel. Judith K. Rosenhouse (Linguist and Humanities and Arts, SWANTECH Ltd. and Technion I.I.T., 9 Kidron St., Haifa 3446310, Israel, judith@swantech.co.il) and Jomana Abu Dahoud (Commun. Disord., Tel-Aviv Univ., Tel-Aviv, Israel)

Phonetic studies of the acquisition of Arabic dialects are few, both in Israel and in Arab countries where native speakers use Arabic dialects as their daily communication means. The need to know when and how dialects split in the numerous Arabic-speaking communities is important both for the linguistic aspect and for some practical clinical goals. The current study focuses on the acquisition of consonants in Kfar Kanna (near Nazareth), a village in the north of Israel, which has a mixed population of Christian and Moslem inhabitants. Participants were altogether 127 girls and boys, in six age groups (3:00 to 7:00 years old) of those faith communities. The children had normal development and no hearing or speech problems. Our findings show that the pronunciation of the colloquial Arabic consonantal inventory develops with age, as expected. It also assimilates gradually to the adult faith groups' phonetic systems. This is evident, especially in a few distinctive consonants, e.g., /q, D^3/. Due to schooling, some effects of the Modern Standard Arabic phonetic system are also evident in the older children's data. This is a first study of communal Arabic dialects development in Israel, as far as we know.

5aSC12. Detection of creaky voice as a function of speaker pitch and gender. Lisa Davidson (Linguist, New York Univ., 10 Washington Pl., New York, NY 10003, lisa.davidson@nyu.edu)

Creaky voice in American English speakers (especially women) has been flagged as a negative characteristic, such as in business and radio (Anderson *et al.* 2014, Glass 2015). However, it is unclear how accurately naïve listeners can identify creaky voice, and what factors facilitate or hinder its identification. American listeners (N = 55) are presented with stimuli from four podcast hosts: a high- and low-pitched male speaker, and a high- and low-pitched female speaker. Other manipulated factors include whether or not the utterance is a full sentence, and whether the utterance is completely modal, completely creaky, or partially creaky (begins modal and ends creaky). After a short familiarization, listeners identify whether 1.5sec utterances contain creaky voice. Results show that listeners are significantly less accurate in identifying creak in both male speakers than in both female speakers, and less accurate when the utterance is partially creaky. For male speakers, listeners are more accurate on fragments than on full sentences. Lower accuracy on male speech may be a combination of smaller, less noticeable differences between average modal and creaky pitch, if listeners heavily rely on low F0 as a cue to creaky voice (Khan *et al* 2016), and bias toward attributing creak to female voices.

5aSC13. The role of voice onset time in the perception of English voicing contrasts in children's speech. Elaine R. Hitchcock (Commun. Sci. and Disord., Montclair State Univ., 116 West End Ave., Pompton Plains, NJ 07444, hitchcocke@mail.montclair.edu) and Laura L. Koenig (Haskins Labs., New Haven, CT)

The perception of phonemic voicing distinctions is typically attributed mainly to voice onset time (VOT). Most previous research focusing on voicing discrimination used synthetic speech stimuli varying in VOT. Results of this work suggest that adult listeners show stable crossover boundaries in the 20-35 ms range. However, no research has evaluated how VOT values correspond to adult labeling regardless of whether the intended target is voiced or voiceless. The present study obtained adult labeling data for natural productions of bilabial and alveolar pairs produced by 2-3-year-old monolingual, English-speaking children. Randomized stimuli were presented twice to 20 listeners resulting in 5,760 rated stimuli. Stimuli were categorized as short VOT (<20 ms), ambiguous VOT (20-35 ms) and long VOT (>35 ms). The findings show that listeners demonstrated the greatest accuracy for bilabials (>99%) and alveolars (>92%) when the target matched the expected VOT duration (i.e., *doe→short lag* and *toe→long lag).* As expected, ambiguous tokens showed generally lower levels of accuracy across all stimuli although listeners were able to identify the target phoneme with greater than chance accuracy. These findings suggest that other variables such as burst intensity, fundamental frequency, and first formant transition duration contribute to adults' perception of children's stops.

5aSC14. Cross-register speaker identification: The case of infant and adult directed speech. Thayabaran Kathiresan, Volker Dellwo (Phonet. Lab., Univ. of Zurich, Plattenstrasse 54, Zurich, Zurich 8032, Switzerland, thayabaran.kathiresan@uzh.ch), Moritz Daum (Development Psych., Univ. of Zurich, Zurich, Zurich, Switzerland), and Rushen Shi (Université du Québec, Montreal, QC, Canada)

The performance of automatic speaker recognition (ASR) systems decreases when training and test data are produced in different social situations (speech registers). The present research tested ASR performance across adult- and infant-directed speech registers (ADS and IDS respectively). IDS compared to ADS is generally characterized by higher and more variable F0, hyper-articulated vowels and higher segment duration and variability. Our dataset consisted of 12 sentences read by 10 Swiss-German mothers to their infants (IDS register) and to an adult experimenter (ADS register). ASR was performed when training and test registers were the same (within register) and when they varied (between register) in 3 experiments. Experiment I used segmental features such as MFCCs and their deltas. Results revealed considerable recognition rate within register (87%) that dropped to about half between registers (44%). This suggests that the variability between IDS and ADS poses challenges on ASR. Experiment II (in progress) uses prosodic features such as F0 statistics, local and long term variations of F0, intensity variations and energy of the frame for the identification. In experiment III, segmental and prosodic features are combined to model the classifier for the identification done in the previous experiments.

5aSC15. Acoustic features and gender differences in clear and conversational speech produced in simulated environments. Shae D. Morgan, Sarah H. Ferguson (Commun. Sci. and Disord., Univ. of Utah, 390 South 1530 East, Ste. 1201, Salt Lake City, UT 84112, shae.morgan@utah.edu), and Eric J. Hunter (Communicative Sci. and Disord., Michigan State Univ., East Lansing, MI)

In adverse listening environments or when barriers to communication are present (such as hearing loss), talkers often modify their speech to facilitate communication. Such environments and demands for effective communication are often present in professions that require extensive use of the voice (e.g., teachers, call center workers, etc.). Women are known to suffer a higher incidence of voice disorders among those in these professions, possibly due to their accommodation strategies they employ when in adverse environments. The present study assessed gender differences in speech acoustic changes made in simulated environments (quiet, low-level noise, high-level noise, and reverberation) for two different speaking style instructions (clear and conversational). Ten talkers (five male, five female) performed three speech production tasks (a passage, a list of sentences, and a picture description) in each simulated environment. The two speaking styles were recorded in separate test sessions. Several acoustic features relevant to clear speech will be compared between simulated environments, speaking styles, and talker genders to identify differences that may begin to address the higher incidence of voice disorders among women in professions with a high vocal use.

5aSC16. Perceptual learning and gender normalization in fricative perception. Benjamin Munson and Mara Logerquist (Univ. of Minnesota, 164 Pillsbury Dr., SE, Minneapolis, MN 55455, munso005@umn.edu)

Listeners identify fricatives ambiguous between /s/ and /ʃ/ differently depending on whether they believe that the talker is a woman or a man (Strand & Johnson, 1996). The current experiment examined how robust this effect is across experimental manipulations in a relatively large group ($n = 99$) of listeners. Listeners identified a sack-shack continuum created by pairing a fricative continuum with a VC token with gender neutral acoustic characteristics. Listeners who participated in an experiment in which the stimuli were paired with a male face did not label fricatives differently from listeners for whom the stimuli were paired with a female face. Listeners who participated in an experiment in which the stimuli were paired with both male and female faces identified fricatives differently in the two conditions, but only for participants who were flexible in their assignment of talker gender from speech stimuli alone in a second experiment. Most strikingly, fricative identification for listeners in all three experiments changed significantly over the course of the experiment, as compared to listeners in a baseline experiment in which the stimuli were not paired with a face. This suggests that the presence of a face delays listeners' perceptual learning of ambiguous sounds.

5aSC17. Acoustic analysis of whispery voice disguise in Chinese. Cuiling Zhang (Southwest Univ. of Political Sci. & Law, Tawan St., NO.83, Huanggu District, Shenyang, Liaoning 110854, China, cuiling-zhang@forensic-voice-comparison.net) and Bin Lin (The City Univ. of Hong Kong, Kowloon, Tong, Hong Kong)

This paper investigates the auditory and acoustical features of disguised whispery voices, the acoustic difference between normal (non-disguised) voices and whispery voices, and the effect of whispery disguise on forensic speaker recognition. Recordings of eleven male college students' normal voices and whispery disguised voices were collected. All their normal and whispery speech was acoustically analyzed and compared. The parameters average syllable duration, intensity, vowel formant frequencies, and long term average spectrum (LTAS) were measured and statistically analyzed. The effect of whispery disguise on forensic speaker recognition by auditory and phonetic-acoustic approach were evaluated. Correlation and regression analyses were made on the parameters of whispery voice and normal voice. To some extent, these simple regression models can be used for parameter compensation in forensic casework.

5aSC18. Phonetic properties of the non-modal phonation in Shanghainese register contrast. Jia Tian and Jianjing Kuang (Linguist, Univ. of Pennsylvania, Apt B3, 4105 Spruce St., Philadelphia, PA 19104, jiatian@sas.upenn.edu)

Earlier literature documented that the tonal registers of Shanghainese were distinguished by both pitch and breathy voice, although recent studies suggested that young speakers might have lost the non-modal phonation. The goal of our present study is two-fold: (1) with the development of analytic tools, we extensively investigated the acoustic and articulatory properties of the breathy voice in Shanghainese; (2) with the ongoing sound change in mind, we recruited a large pool of speakers, born from 1930s to 2000s, so that we are able to discuss the different strategies in producing the register contrast among speakers. Simultaneous audio and EGG signals were collected. Statistical models suggested that speakers born after 1980 use different strategies in producing the register contrast. For older speakers, overall, the lower register has a breathy phonation type: periodicity and noise measures (HNR and CPP) are the most important acoustic correlates, followed by spectral tilts (H1-A1, H1-A2, and H1-A3); articulatorily, Contact Quotient and Speed Quotient reliably distinguish the two registers. For

younger speakers, spectral measures are generally no longer a reliable cue, but noise measures still differ significantly between the two registers, since some younger speakers produce the lower register tones with creaky voice.

5aSC19. Power priming and speech perception. Ian C. Calloway (Linguist, Univ. of Michigan, 2947 Roundtree Blvd., Ypsilanti, MI 48197, icalllow@umich.edu)

English listeners use information about speaker gender in categorizing a sibilant as /s/ or /ʃ/. This study investigates whether sibilant categorization is influenced by whether cues to speaker gender are congruous and whether self-perceived power in turn influences one's ability to process incongruous gender cues. One's self-perceived power, the perceived capacity to control another individual's resources, can inhibit how one attends to information associated with a social category. Participants were primed experimentally for a high or low degree of self-perceived power. They then performed a forced-choice identification task—for each trial, participants were presented with an auditory stimulus—a word ranging from "sigh" to "shy"—and a visual stimulus—a male or female face—and they indicated whether they heard "sigh" or "shy." Participant likelihood to respond "sigh" was significantly influenced by speaker gender, whether the gender of the face matched that of the speaker, and the power prime the participant received. For the female voice, both participants responded "sigh" more often when presented with a male face. For the male voice, however, low-power individuals were less likely to respond "sigh" when presented with a female face, while high-power individuals responded similarly regardless of the face presented.

5aSC20. Apparent-time study of interdental-stopping among English-monolingual Finnish- and Italian-heritage Michiganders. Paige Cornillie, Julianne Fosgard, Samantha Gibbs, Delani Griffin, Olivia Lawson, and Wil A. Rankinen (Commun. Sci. and Disord., Grand Valley State Univ., 515 Michigan St. NE, Ste. 300, Office 309, Grand Rapids, MI 49503, wil.rankinen@gvsu.edu)

The production of coronal oral stops in place of interdental fricatives, referred to as interdental-stopping, has been documented in Michigan's Upper Peninsula (UP) [3, 2], as well as, in other ethnic-heritage influenced English varieties. However, there is a lack of quantitative inquiry into the degree to which this salient feature is present among Michigan UP's now predominantly monolingual English-speaking communities; recent studies have focused primarily on the last remaining older-aged bilinguals [1]. Michigan's UP is in an ideal position to examine to what extent this feature is present among a rural and predominantly monolingual English-speaking community. The present study examines 40 Finnish-Americans and 44 Italian-Americans, whom are all monolingual speakers from Michigan's Marquette County. Both samples are stratified by age, sex, and socioeconomic status. All data are obtained from a passage task. To what degree, if any, does stopping occur among the Finnish- and Italian-heritage monolingual-English speaking communities? This study reveals interdental-stopping occurring most often among working-class males but least among Italian middle- and younger-aged groups. The study's apparent-time construct highlights a potential change in the covert prestige that has been typically associated with this feature among the older generation [1]. The decrease of interdental-stopping among the younger generation indicates a shift in the feature's prestige within the community.

5aSC21. Generalization of cross-category phonetic imitation of Mandarin regional variants. Qingyang Yan (Linguist, The Ohio State Univ., 591 Harley Dr. Apt. 10, Columbus, OH 43212, yan@ling.ohio-state.edu)

The current study investigated cross-category phonetic imitation of Jianshi Mandarin regional variants by Laifeng participants. An auditory shadowing task and a post-exposure reading task were used to examine how participants imitated vowel variant [i] and coda variant [n] during shadowing and how imitation generalized to various types of novel words. During the post-exposure reading task, participants were instructed to say the words like the talker they heard, and to simply read the words, in a between-subjects design. Laifeng participants consistently imitated Jianshi [i] and [n] variants during shadowing, and imitation generalization was observed, but only when participants were instructed to imitate the previously heard talker. Imitation of both variants generalized to novel words with the same syllables (onset + rime + tone) but different orthography, novel words with the same onset and rime but different tones, and novel words with the same rime and tone but different onsets from the shadowed words, with similar degrees of imitation across these three types of novel words. These results suggest that generalization of cross-category imitation is a controlled, as opposed to automatic, process, and that cross-category imitation can operate and generalize at syllable, onset + rime, and phoneme levels in Mandarin.

5aSC22. Effects of age, sex, context, and lexicality on hyperarticulation of Korean fricatives. Charles B. Chang (Linguist Program, Boston Univ., 621 Commonwealth Ave., Boston, MA 02215, cc@bu.edu) and Hae-Sung Jeon (School of Lang. and Global Studies, Univ. of Central Lancashire, Preston, Lancashire, United Kingdom)

Seoul Korean is known for a rare three-way laryngeal contrast among *lenis*, *fortis*, and *aspirated* voiceless stops, which has recently undergone a change in phonetic implementation: whereas older speakers rely more on voice onset time (VOT) to distinguish lenis and aspirated stops, younger speakers rely more on onset fundamental frequency (f_0) in the following vowel. This production difference is reflected in disparate strategies for enhancing the contrast in clear speech, supporting the view that younger and older speakers represent the three laryngeal categories differently in terms of VOT and f_0 targets (Kang & Guion, 2008). In the current study, we used the clear speech paradigm to test for change in the representation of the two-way contrast between fortis (/s*/) and non-fortis (/s/) fricatives. Native Seoul Korean speakers (n = 32), representing two generations and both sexes, were recorded producing the coronal stops and fricatives in different vowel contexts, item types (real vs. nonce words), and speech registers (plain citation vs. clear). We report acoustic data on how the above factors influence production of the fricative contrast and discuss implications for the phonological categorization of non-fortis /s/ as lenis, aspirated, or a hybrid lenis-aspirated category.

Session 5aSPa

Signal Processing in Acoustics: Audio and Array Signal Processing I

Gary W. Elko, Cochair
mh Acoustics, 25A Summit Ave., Summit, NJ 07901

Janina Fels, Cochair
Institute of Technical Acoustics, RWTH Aachen University, Neustr. 50, Aachen 52074, Germany

David C. Swanson, Cochair
Penn State ARL, 222E ARL Bldg., PO Box 30, State College, PA 16804

Contributed Papers

8:20

5aSPa1. Comparison of beamforming methods to reconstruct extended, partially-correlated sources. Blaine M. Harker, Kent L. Gee, Tracianne B. Neilsen (Dept. Phys. & Astronomy, Brigham Young Univ., N283 ESC, Provo, UT 84602, blaineharker@gmail.com), and Alan T. Wall (Battlespace Acoust. Branch, Air Force Res. Lab, Wright-Patterson AFB, OH)

Advanced cross-beamforming methods improve upon traditional beamforming to reconstruct complex source information and to estimate their respective acoustic radiation. Regularization of the cross-beamforming matrix as part of the calculation procedure helps improve method robustness, but differences in implementation impact volume velocity source results and subsequent field predictions. This paper compares the abilities of four regularization-based, cross-beamforming methods: hybrid method, functional beamforming, generalized inverse beamforming, and mapping of acoustic sources (MACS), along with ordinary cross beamforming, in their ability to reproduce source and field characteristics for an extended, partially correlated numerical source that mimics key characteristics of supersonic jet noise radiation. The four methods that rely on regularization significantly outperform cross-beamforming results, yet the effectiveness of each of four methods is dependent on the individual regularization schemes, which depend on frequency as well as the signal-to-noise ratio. Within those methods, generalized inverse method shows the greatest resistance to regularization variability when comparing results to benchmark cases, although in many cases the hybrid method can give slight improvements. The successful application of the methods demonstrates the utility of cross-beamforming in formulating equivalent source models for accurate field prediction of complex sources, including jet noise. [Work supported by ONR and USAFRL through ORISE.]

8:40

5aSPa2. Single-layer array method to reconstruct extended sound sources facing a parallel reflector. Elias Zea (KTH Royal Inst. of Technol., Teknikringen 8, Stockholm 100 44, Sweden, zea@kth.se) and Ines Lopez-Arteaga (KTH Royal Inst. of Technol., Eindhoven, Netherlands)

The accuracy of sound field reconstruction methods with single-layer microphone arrays is subject to the room or enclosure in which the measurements take place. Thus, the authors recently introduced a single-layer method that can be employed to reconstruct compact sources in the presence of a reflecting surface that is parallel to the array. Now the authors propose a method conceived for extended planar sources such as baffled plates facing a parallel reflector. The method is based on a wavenumber-domain function describing the propagation paths between the source, the reflector and the array. The free-space sound field radiated by the source is then recovered by means of a regularized inversion of the propagation function. Numerical simulations are performed in order to assess the method's performance and potential for source reconstructions. The results are promising and point towards future experimental validation.

9:00

5aSPa3. Measurement extension limits of patch nearfield acoustic holography. Kanthasamy Chelliah and Ralph T. Muehleisen (Argonne National Lab., 9700 S Cass Ave., Bldg. 362, Lemont, IL 60439, kchelliah@anl.gov)

Patch nearfield acoustic holography is widely used when the measurement aperture is smaller than the area of the sound source. This paper discusses the factors affecting the limit of aperture extension. One step patch nearfield acoustic holography based on equivalent sources model was considered in this study. Available literature provide constant estimations on the ratio of measurement area to the area of the source. However, the present study shows that this extension limit is a variable depending on a few parameters. Wave number, hologram distance and choice of regularization technique were found to affect the extended area of reconstruction significantly. By choosing the right combination of these parameters, accurate reconstructions were achieved even when the measurement area was only ten percent of the source surface in size. This paper provides a systematic comparison of reconstruction errors for cases with various parameter combinations. A model relating the parameters and the extension limit is under development.

9:20

5aSPa4. The nearfield acoustic holography using a concentric rigid microphone array to identify cylindrical sources disturbed by environment noise. Weikang Jiang and Shang Xiang (Shanghai Jiao Tong Univ., 800 Dong Chuan Rd., Shanghai 200240, China, wkjiang@sjtu.edu.cn)

Many nearfield acoustic holography (NAH) techniques were proposed to identify acoustical noise sources, which were usually available in the free sound field. To extend NAH to more applications in noisy environments such as factories, an NAH based on a rigid rectangle microphone array was proposed to reconstruct normal velocities of vibrating plates in the environment with disturbing noise sources recently. In this presentation, an NAH using rigid microphone array with the profiles matching the vibrating surface is proposed to identify the cylindrical surface sources. The microphone array is flush-mounted at the bottom plane, and two concentric side surfaces and two rectangular side surfaces are designed to satisfy the Neumann's boundary condition. The transfer matrix between the normal velocities of sources and measured pressure is expressed as the summation of acoustic modes of the volume within the rigid array. The disturbing waves from environments can be decomposed by the inverse patch transfer functions

method. The normal velocities of a cylindrical surface source are correctly reconstructed in the numerical simulation as well as a motor operating in a workshop. The results indicate that the proposed procedure is valid for reconstructing velocities of the cylindrical sound source in a noisy environment.

9:40

5aSPa5. Model-matching for impulse sound localization in urban areas. Sylvain Cheinet and Loic Ehrhardt (ISL, 5 Rue General Cassagnou, Saint-Louis 68300, France, sylvain.cheinet@isl.eu)

The presentation addresses the localization of a punctual, impulse sound source with distributed sensors in an urban area of typical size 150mx150m. In the considered approach, the source is localized by matching some observed characteristics of the signals (here, the first times-of-arrival) to those obtained from a pre-defined database of simulations with known source positions. The localization performance is analyzed on the basis of small-scale model and real-scale measurements, with various building heights, source positions, combinations from 15 down to 5 microphones. The time matching localization attains an accuracy of 10 m in the vast majority of the configurations. A confidence level for each localization is satisfactorily tested. The robustness to the buildings height and to the urban map is evaluated. Adaptations of the approach to real-time constraints and to shot and shooter localizations are demonstrated.

10:00–10:20 Break

10:20

5aSPa6. Localization and source assignment of blast noises from a military training installation. Michael J. White, Edward T. Nykaza (US Army ERDC/CERL, PO Box 9005, Champaign, IL 61826, michael.j.white@usace.army.mil), and Andrew Hulva (US Army ERDC/CERL, Blacksburg, VA)

Time differences of arrival (TDOA) are often sufficient data for localization of a sound source with a sensor array. We consider the problem of localizing multiple impulsive sound sources that occur on a military installation having live-fire training exercises, using a dozen or more noise monitors as a large array. In this setup, though, sounds from multiple events can arrive in different order at different monitors. When multiple sources are operating, ambiguous source-detection assignments degrade the localization estimates. By minimizing a global cost-function on the entire detection catalog, we resolve source-detection assignments and improve the localizations. We outline an estimator and show results with simulated data and field measurements.

10:40

5aSPa7. DOA (direction of arrival) estimation of incident sound waves on a spherical microphone array: Comparison of some correction methods proposed to solve the DOA bias. Jean-Jacques Embrechts (Elec. Eng. and Comput. Sci., Univ. of Liege, Campus du Sart-Tilman B28, Quartier Polytech 1, 10 Allée de la découverte, Liege 4000, Belgium, jjembrechts@ulg.ac.be)

A spherical 16-microphones array has been designed and built for room acoustics applications (measurement of directional room impulse responses or DRIR). The identification of strong specular reflections in a DRIR requires an accurate determination of their direction of incidence on the sphere. Beamforming methods are therefore applied for their DOA (direction of arrival) estimation. However, the crude applications of these methods revealed significant deviations between the measured and expected DOA. In-depth experiments in anechoic conditions have been carried out to analyze this problem and they revealed that the origin of these deviations could be related to the non-rigidity of the spherical surface of the array. In these experiments, incident waves were generated by a pseudo—point source at several frequencies in the anechoic chamber. The microphone array is rotated and oriented at several azimuthal positions. The sound pressures measured on the sphere by the 16 microphones are then compared with their theoretical values obtained with a rigid sphere assumption, which revealed some differences. Some methods are then proposed to correct this problem. These methods are finally presented and their results in DOA estimation tasks are compared.

11:00

5aSPa8. Deconvolving the conventional beamformed outputs. Tsih C. Yang (College of Information Sci. and Electron. Eng., Zhejiang Univ., Rm. 412, Bldg. of Information Sci. and Electron. Eng., 38 Zhe Da Rd., Xihu District, Hangzhou, Zhejiang, China, Hangzhou 310058, China, tsihyang@gmail.com)

Horizontal line arrays are often used to detect/separate a weak signal and estimate its direction of arrival among many loud interfering sources and ambient noise. Conventional beamforming (CBF) is robust but suffers from fat beams and high level sidelobes. High resolution beamforming such as minimum-variance distortionless-response (MVDR) yields narrow beam widths and low sidelobe levels but is sensitive to signal mismatch and requires many snapshots of data. This paper applies deconvolution algorithm used in image de-blurring to the conventional beam power of a uniform line array (spaced at half-wavelength) to avoid the instability problems of common deconvolution methods. The deconvolved beam power yields narrow beams, and low sidelobe levels similar to, or better than MVDR and at the same time retains the robustness of CBF. It yields a higher output signal-to-noise ratio than MVDR for isotropic noise. Performance is evaluated with simulated and real data. Deconvolution is also applied to a circular array to compare with that obtained using superdirective beamforming (SDB) at small ka, where a is the radius and k is the wavenumber. The deconvolved beam output achieves similar performance as the SDF, and offers the same robustness as CBF at small ka.

11:20

5aSPa9. Separating silent sources. Richard Goldhor (Speech Technol. & Appl. Res., 54 Middlesex Turnpike, Entrance D, Bedford, MA 01730, rgoldhor@sprynet.com), Keith Gilbert (Speech Technol. & Appl. Res., Berlin, MA), and Joel MacAuslan (Speech Technol. & Appl. Res., Bedford, MA)

In the presence of multiple continuously active acoustic sources, microphone response signals are additive mixtures of acoustic images: that is, filtered, delayed, and scaled versions of the source signals. By adaptively minimizing a contrast function such as the mutual information between outputs, Blind Source Separation (BSS) algorithms can "demix" sets of such microphone responses into output signals, each one of which is coherent with a single hidden source signal. The time required to converge on such a "separation solution" is typically at least several tens of seconds. However, important real-world acoustic signals of interest (such as speech) are intermittent, not continuous. As a result, the response mixtures resulting from such signals include "mute episodes" during which one or more of the sources are silent. These episodes can be employed to advantageously partition the adaptive separation process, and even to generate permanently valid separation solutions for the momentarily silent sources. We present a method of rapid optimal isolation (ROI) of intermittent sources, and explore the utility of enhancing an adaptive BSS algorithm using this new method.

11:40

5aSPa10. Single channel speech enhancement based on harmonic estimation combined with statistical based method to improve speech intelligibility for cochlear implant recipients. Dongmei Wang and John H. L. Hansen (Elec. Eng., Univ. of Texas at Dallas, 800 West Campbell Rd., ECSN 4.414, Richardson, TX 75080, dongmei.wang@utdallas.edu)

In this study, we propose a single microphone speech enhancement algorithm by combining harmonic structure estimation and traditional MMSE speech enhancement for a leveraged overall solution. Traditional single channel speech enhancement methods are usually based on the statistic characteristics of noise signals which are effective only for stationary noise, but not for non-stationary noise. In our study, we attempt to estimate noise by exploring the harmonic structure of the target speech combined with temporal noise tracking. In voiced segments, since speech energy is sparsely carried by harmonic partials, the spectrum located between adjacent harmonic partials are considered as noise. We assume that the speech spectrum distributes continuously along the frequency-dimension. Thus, the noise overlapped with speech harmonics can be estimated with an interpolation technique. Next, the estimated noise is incorporated into a traditional MMSE framework for speech enhancement. A listening test is carried out

with 6 cochlear implant recipients to evaluate the proposed speech enhancement algorithm. The experimental results show that the proposed algorithm is able to improve the speech intelligibility in terms of word recognition rate for CI listeners.

12:00

5aSPa11. On the influence of continuous subject rotation during HRTF measurements. Jan-Gerrit Richter and Janina Fels (Inst. of Tech. Acoust., RWTH Aachen Univ., Kopernikusstr. 5, Aachen 52074, Germany, jri@akustik.rwth-aachen.de)

In recent years, the measurement time of individual Head-Related Transfer Function (HRTF) measurements has been reduced by the use of loudspeaker arrays. The time reduction is achieved by some kind of parallelization of measurement signals. One such fast system was developed at the Institute of Technical Acoustics, RWTH Aachen University and is evaluated in this paper. When measuring HRTFs, the subject is usually rotated by some angle, and stops and waits for the measurement signal to complete before moving to the next measurement angle. It was shown that with this static approach a comparable results to a traditional measurement using a single speaker could be achieved. To further reduce the measurement time, a slow continuous subject rotation can be used instead. While this rotation will violate LTI (linear, time-invariant) requirements of the commonly used signal processing, the influence is assumed to be negligible. As the subject is rotating during the measurement sweep, different azimuth angles are measured per frequency. This frequency dependent offset in the measurement positions has to be corrected during the post processing. To this end, a spherical harmonic decomposition and reconstruction is applied as an interpolation method. To quantify the influence of the rotation and the subsequent post processing, a subjective and objective comparison between statically and continuously measured objects is shown in this paper.

THURSDAY MORNING, 29 JUNE 2017 — ROOM 304, 8:20 A.M. TO 12:20 P.M.

Session 5aSPb

Signal Processing in Acoustics and Underwater Acoustics: Underwater Acoustic Communications

Milica Stojanovic, Chair
ECE, Northeastern Univ., 360 Huntington Ave., 413 Dana Bldg., Boston, MA 02115

Contributed Paper

8:20

5aSPb1. Power delay profiles and temporal vs. spatial degrees of freedom: Predicting and optimizing performance in underwater acoustic communications systems. James Preisig (JPAnalytics LLC, 638 Brick Kiln Rd., Falmouth, MA 02540, jpreisig@jpanalytics.com)

The performance of underwater acoustic communications systems is often characterized as a function of source to receiver range or the received in-band SNR. However, these measures are incomplete with respect to predicting performance. There is ample field data for which performance improved with increasing range or for which data sets with similar SNRs exhibited markedly different performance. Using field and simulation data, the Primary to Intersymbol Interference (ISI) and Noise Ratio (PINR) is shown to be a more complete predictor of performance than SNR. The results also demonstrate the significant adverse effect of ISI on system performance. With this in mind, the optimal configuration in terms of mitigating the impact of ISI is considered for the receive array and processor structure of equalizers. The role of total degrees of freedom (DOFs), equalizer filter adaptation averaging interval, and the relative stability of the channel's spatial and temporal structures is evaluated. The performance gains attainable with array spatial aperture vs the gains attainable with filter temporal depth is analyzed using closed form expressions as well as simulation and field data.

Invited Papers

8:40

5aSPb2. Acoustic Communications: Through soils, sands, water, and tissue. Andrew Singer (Elec. and Comput. Eng., Univ. of Illinois at Urbana Champaign, 110 CSL, 1308 W. Main St., Champaign, IL 61801, acsinger@illinois.edu), Sijung Yang, and Michael Oelze (Elec. and Comput. Eng., Univ. of Illinois at Urbana Champaign, Urbana, IL)

This talk will cover several experimental results and the challenges that arise in applying the basic concept of acoustic communications to vastly different media. Specifically, results obtained from state-of-the-art ultrasonic communications over distances of 100 m at data rates in excess of 1Mbps will be described, along with the unique challenges that such high data rates impose of any subsea applications, for which traditional Doppler compensation techniques are wholly inadequate. A sample-by-sample time-scale projection method was developed for streaming video applications for subsea operations. Over much shorter distance scales, even more acoustic bandwidth is available, and data races in excess of 300 MB/s have been achieved. Difficulties in such applications arise due to the nonlinearities excited in both the medium and potentially the transducers owing to the extremely wide bandwidths of operation. For potential

biomedical applications, acoustic communication methods have been successfully used to transmit data in excess of 30 Mb/s through animal tissue. Challenges in such applications include no only nonlinearity and Doppler compensation, but also reverberation and scattering off bone and other materials. A final application that will be discussed is the use of acoustic communications methods for through-soils and through-sands experiments with application to various geotechnical projects. Attenuation, coupling, and excitation pose substantive challenges for such applications.

9:00

5aSPb3. Reliable data delivery using packet coding over acoustic links. Rameez Ahmed (Cambridge, MA) and Milica Stojanovic (ECE, Northeastern Univ., 360 Huntngton Ave., 413 Dana Bldg., Boston, MA 02115, millitsa@ece.neu.edu)

Acoustic links are challenged by high bit error rates, which cause data packets to be declared as erroneous. To prevent the packet loss and achieve reliable data transmission over such links, some form of feedback must be implemented to deliver acknowledgments from the receiver to the transmitter, thus initiating re-transmission of erroneous packets. Conventionally, data packets are grouped and a selective acknowledgment procedure of the stop-and-wait type is employed as a suitable solution for half-duplex acoustic links. We revisit the issue of reliable data transmission and investigate the use of random packet coding in conjunction with a selective acknowledgment procedure. Specifically, we introduce random packet coding, and regard a block of coded packets as an equivalent of a packet—now termed the super-packet. We then form groups of super-packets and apply a selective acknowledgment procedure on the so-obtained units. Analytical results, obtained with experimentally verified channel models, demonstrate the power of grouped packet coding which offers a much improved throughput-delay performance on randomly time-varying acoustic channels with long delays.

Contributed Papers

9:20

5aSPb4. Estimating the acoustic channel to improve underwater communication. Ballard J. Blair (Electron. Systems and Technol. Div., MITRE Corp., 202 Burlington Rd., Bedford, MA 01730, bblair@mitre.org)

With the increasing number of unmanned vehicles and sensors being deployed to support scientific research, oil and gas exploration, and military operations, research and development in underwater acoustic communication is vital. The costs for testing new ideas for underwater acoustic communication is often prohibitively high due to expensive ship time, personnel, and equipment. It is also challenging to accurately model the physics and dynamics of the underwater communication channel because it is spread in both Doppler and delay. The goal of the presented research is to create a library of channels extracted from available experimental data sets. The extracted channels are compared with physical models to see which characteristics, if any, are predictable from the observed data sets. This library of channels would be used both for comparison of existing underwater communication systems and to aid in the development of communication techniques. [Work supported by the MITRE Innovation Program.]

9:40

5aSPb5. Combining sparse recovery approaches with underwater acoustic channel models for robust communications in the shallow water paradigm. Ananya Sen Gupta (Elec. and Comput. Eng., Univ. of Iowa, 4016 Seamans Ctr. for the Eng. Arts and Sci., Iowa City, IA 52242, ananya-sengupta@uiowa.edu)

Shallow water acoustic communication techniques are fundamentally challenged by the rapidly fluctuating multipath effects due to oceanic phenomena such as surface wave focusing, specular reflections from the moving sea surface, Doppler effects due to fluid motion as well as sediment-dependent absorption from the sea bottom. Several signal processing techniques have been recently proposed that specialize in recovering the shallow water acoustic channel components using compressing sampling and mixed norm optimization theory. However, these novel techniques are typically agnostic of underlying underwater acoustic propagation phenomena. Furthermore, state-of-the-art in shallow water channel estimation typically does not account for the three-way uncertainty principles governing the localization of sparsity, time, and frequency for the time-varying shallow water acoustic channel. This talk will focus on recent techniques proposed in this domain, their relative benefits and shortcomings; as well as offer new insights into how we can combine knowledge of basic underwater acoustic propagation physics with state-of-the-art in sparse sensing and related techniques to achieve robust shallow water channel estimation. The talk will also provide an overview of equalization techniques that can be harnessed with channel estimation techniques proposed.

10:00

5aSPb6. A multiple-input multiple-output orthogonal frequency division multiplexing underwater communication system using vector transducers. Yuewen Wang, Erjian Zhang, and Ali Abdi (Elec. Comput. Eng., New Jersey Inst. of Technol., University Heights, Newark, NJ 07102, ez7@njit.edu)

Vector sensors and transducers are compact multichannel devices that can be used for underwater communication via acoustic particle velocity channels (A. Abdi and H. Guo, "A new compact multichannel receiver for underwater wireless communication networks," IEEE Transactions on Wireless Communications, vol. 8, pp. 3326-3329, 2009). In this paper, a multiple-input multiple-output (MIMO) underwater acoustic communication system is presented using orthogonal frequency division multiplexing (OFDM) modulation. Upon transmitting multiple independent data streams simultaneously over several channels, this MIMO system can increase the transmission rate, whereas the OFDM modulation mitigates the highly frequency selective underwater channels. Various components of the system including vector transducers and algorithms for synchronization, channel estimation, MIMO detection, channel coding, etc., are designed and implemented. Using this system, experiments are conducted to measure and study acoustic particle velocity channels in the MIMO setup. Additionally, system performance parameters such as bit error rate and spectral efficiency are measured and discussed for various conditions and configurations, to understand the performance of the developed vector MIMO-OFDM system. [The work was supported in part by the National Science Foundation (NSF), Grant IIP-1500123.]

10:20–10:40 Break

10:40

5aSPb7. Very low signal to noise ratios coherent communications with an M-ary orthogonal spread spectrum signaling scheme. Jacob L. Silva and Paul J. Gendron (ECE Dept., Univ. of Massachusetts Dartmouth, 285 Old Westport Rd., Darmouth, MA 02747, jsilva13@umassd.edu)

Considered here is the use of an M-ary orthogonal spread spectrum signal set for coherent symbol detection at very low signal to noise ratios (SNR). We consider symbols that are convolutionally orthogonal over the entire multipath delay spread of the channel and allow for the minimum mean square error (MMSE) estimation of an observed acoustic response function with minimal computational effort at the receiver. We employ these signals with an assumed sparsity prior to implement a joint symbol and broadband channel estimation scheme for coherent detection without the use of intra-packet training symbols of any kind. The approach allows us to compensate for the shared time varying Doppler process of the various coherent arrivals. The approach is demonstrated with at-sea recordings at extremely low received SNR.

11:00

5aSPb8. Combating impulsive noise for multicarrier underwater acoustic communications. Zhiqiang Liu (US Naval Res. Lab., 4555 Overlook Ave., Washington, DC 20375, zhiqiang@ieee.org)

This paper presents a novel multicarrier communication scheme that is specially designed for underwater acoustic channels with dominant impulsive noise. A novel multicarrier signaling structure is introduced at the transmitter. Thanks to the signaling structure, the received signal is shown satisfying some special properties even after various channel distortions. By exploiting these properties, we propose two receiver processing algorithms, one for signaling parameter estimation and the other for symbol recovery. The two algorithms take fully into account the presence of strong impulsive noise, and both are shown capable of offering inherent robustness against it. The proposed design is first evaluated via extensive simulations and then tested in a recent sea-going experiment. Both simulated and experimental results validate the proposed design and confirm its merits. [This work was supported by US Office of Naval Research.]

11:20

5aSPb9. Broadband underwater acoustic communications subsystem design. Corey Bachand, Tyler Turcotte, and David A. Brown (BTech Acoust. LLC, 151 Martine St., Fall River, MA 02723, corey.bachand@cox.net)

Increased bandwidth for acoustic communication places demanding requirements on the electroacoustic transducer, tuning and matching circuits, and amplifier design. This presentation investigates various aspects of the transmit channel with a focus on increasing bandwidth to the fullest extent possible. This includes using alternative transducer designs, polarization orientations, transduction materials, broadband tuning networks, and gain/phase control in high efficiency Class-D amplifiers. Comparisons between modeled and measured performance for systems that achieve 10, 20, and 30 kHz are considered.

11:40

5aSPb10. Blind adaptive correlation-based decision feedback equalizer (DFE) for underwater acoustic communications. Xiaoxia Yang, Jun Wang, and Haibin Wang (Chinese Academic of Socience, Inst. of Acoust., No. 21 North 4th Ring Rd., Haidian District, Beijing 100190, China, yangxiaoxia@mail.ioa.ac.cn)

Passive time reversal uses a set of matched filters corresponding to individual channel impulse response to reduce intersymbol interferences (ISIs) in underwater acoustic communications. Because of residual ISIs after time reversal, a coupled single decision feedback equalizer (DFE) is necessary. The coupled DFE uses a fixed tap number applicable to most shallow oceans with less user supervision. However, this correlation-based DFE needs the training sequence to estimate the individual channel response and initialize DFE, which decreases the effective bit rate and increases user supervision. We therefore in this paper propose a new blind structure of time reversal coupling with DFE, and it can complete multichannel identification and equalization at the same time without the training sequence. This new receiver has a reversible structure. In the first mode, it is linear and adapted by blind algorithms, e.g. constant modulus algorithm (CMA), to estimate the multichannel impulse responses and initialize the equalizer. In the second mode, the receiver becomes correlation-based DFE. From the viewpoint of user supervision and spectral efficiency, the new structure with no training sequence is more attractive. The blind correlation-based DFE exhibits the same steady-state mean square error (MSE) with the trained structure, and has been validated on real underwater communication data.

12:00

5aSPb11. Cyclic-feature based Doppler scale estimation for orthogonal frequency-division multiplexing (OFDM) signals over doubly selective underwater acoustic channels. Bingbing Zhang (College of Electron. Sci. and Eng., National Univ. of Defence Technol., Kaifu district, Changsha, Hunan 410073, China, zbbzb@nudt.edu.cn), Yiyin Wang (Dept. of Automation, Shanghai Jiao Tong Univ., Shanghai, China), Hongyi Wang, Liming Zheng, Zhaowen Zhuang (College of Electron. Sci. and Eng., National Univ. of Defence Technol., Changsha, China), and Kele Xu (College of Electron. Sci. and Eng., National Univ. of Defence Technol., Paris, France)

Underwater acoustic (UWA) communications enable underwater wireless networks to be applied in various applications, such as oceanographic research, pollution early-warning, disaster prevention, and military systems. However, a major challenge in UWA communications is to combat the Doppler distortion caused by doubly selective (time and frequency selective) channels. Most Doppler scale estimators rely on training data or specially designed packet structures. These methods have fundamental limitations in transmission rate and spectral efficiency. Different from these methods, this paper presents a Doppler scale estimation approach exploiting the redundant information contained within the cyclic prefix (CP) or cyclic suffix (CS) of orthogonal frequency-division multiplexing (OFDM) signals. We analyze the cyclic features of OFDM signals over doubly selective underwater channels in order to demonstrate the relationship between the cyclic features and the Doppler scale. Based on the theoretical analyses, we find that the Doppler scale can be estimated from the extremums of the cyclic autocorrelation function (CAF) of the received signal. Simulations validate our theoretical analyzes and the performance of the proposed Doppler scale estimator. Apart from the high estimation performance, we also highlight the utility of our method when only few OFDM blocks are available.

Session 5aUWa

Underwater Acoustics: Acoustical Localization, Navigation, Inversion, and Communication

Aijun Song, Chair
School of Marine Science and Policy, University of Delaware, 114 Robinson Hall, Newark, DE 19716

Contributed Papers

8:00

5aUWa1. Over-sampling improvement for acoustic triangulation using Barker code audio signals. Romuald Boucheron (DGA HydroDynam., Chaussée du Vexin, Val-de-Reuil 27105, France, romuald.boucheron@intradef.gouv.fr)

Acoustic triangulation applications are widely used in submarine domain to estimate 3D-coordinates of a free model or buoy localization, for example. Accuracy of positioning is dominated by the "perfect" knowledge of sound celerity in the domain and also by temporal synchronization. This short communication deals with the use of Barker audio code signals in order to improve temporal detection. Indeed, properties of such signals allow performing correlation with over-sampling detection like spatial subpixel algorithms in image processing domain. The first part of the communication presents this kind of signal and theoretical expected performances with very low signal to noise ratio. The second part is dedicated to the results obtained with an academic experiment devoted to measure directivity pattern of a sub-marine acoustic source. In order to decrease uncertainty of this measurement, over-sampling algorithm has been used successfully to estimate the position of the source during its displacement. This experimental set up allows in return a good angular resolution and an accurate estimate of the directivity pattern.

8:20

5aUWa2. Underwater source localization using unsynchronized passive acoustic arrays. Daniel P. Zitterbart and Ying-Tsong Lin (Appl. Ocean Phys. & Eng., Woods Hole Oceanographic Inst., 213 Bigelow Lab., MS#11, Woods Hole, MA 02543, dpz@whoi.edu)

Localizing sources of underwater sound is a well studied field that is utilized by several scientific and naval communities. The scope of localization might differ dramatically, from the necessity to localize targets with a submeter accuracy to estimation of the position of an object on a kilometer scale. Advances in data storing capabilities during the last decade now allow multi-year deployments of autonomous passive acoustic monitoring arrays for which past-recovery time synchronization cannot be guaranteed. For localization of transient signals, like marine mammal vocalization, arrival time based localization schemes are currently the prevalent method. Applying arrival time based methods to non-synchronized multi station arrays eventually leads to large localization uncertainties. Here, we present a backpropagation based localization scheme that overcomes the necessity to synchronize between array stations for localization purposes. It utilizes waveguide dispersion measured within distributed arrays for simultaneous source localization and time synchronization. Numerical examples are presented to demonstrate that localization uncertainty significantly improves compared to arrival time based methods.

8:40

5aUWa3. Geoacoustic inversion using distributed sensors. Jingcheng Zhang (College of Information Sci. and Electron. Eng., Zhejiang Univ., Bldg. of Information Sci. and Electron. Eng., 38 Zhe Da Rd., Hangzhou, Zhejiang 310058, China, 597572827@qq.com) and Tsih C. Yang (College of Information Sci. and Electron. Eng., Zhejiang Univ., Kaohsiung, Taiwan)

Geoacoustic inversion is a method using sound to invert for the sediment/bottom properties. The method has been researched extensively using sound received on a vertical or horizontal line array (VLA/HLA). It requires usually a dedicated ship to deploy the array and source to collect the data. Conducting such experiments are costly and its' use is limited in practical applications. It is anticipated in the future that many low-cost acoustic transceivers will be deployed in the ocean in the form of a distributed sensor network. It seems logical to use these distributed sensors as the receivers to conduct geoacoustic inversion using a ship-mounted source, thus covering a wide area wherever the distributed nodes are deployed. The question is how well can one estimate the bottom parameters using distributed sensors, compared with that using the traditional arrays. Note that the VLA/HLA data offers spatial coherence, thus one usually carries out geoacoustic inversion using the so called matched-field inversion. The signal received on the distributed data may not be coherent and one will conduct geoacoustic inversion using frequency-coherent inversion methods. The advantage of distributed receiver is spatial diversity, particularly when the source moves. Simulation results will be presented to compare their performance.

9:00

5aUWa4. Source depth discrimination using the array invariant. Rémi Emmetiere, Julien Bonnel (ENSTA Bretagne, 2 rue Francois Verny, Brest cedex 9 29806, France, remi.emmetiere@ensta-bretagne.org), Marie Gehant Pernot (THALES, Sophia Antipolis, France), and Thierry Chonavel (Télécom Bretagne, Plouzané, France)

In this study, low frequency (0-500 Hz) source depth discrimination in deep water is investigated using a horizontal line array (HLA). In this context, propagation is dispersive and can be described by modal theory. Array invariant theory is known to allow source ranging via the modal beam-time migration pattern. This pattern is defined by the evolution of a conventional beamformer output over time. Recently, it has been shown that the array invariant and the waveguide invariant are intrinsically related. In other words, the beam-time migration pattern can be derived using waveguide invariant theory. Utilizing this dependence, we pursue the link between the two invariants to perform source depth discrimination using a HLA. Since the waveguide invariant distribution is strongly related to source depth, the beam-time migration pattern also depends on source depth and would allow for source depth discrimination to be achieved. The proposed method is successfully applied on simulated data. As the classical array invariant method, it is restricted to short signals, away from the array broadside, but it could be used with minimal environmental knowledge in a multisource context.

9:20

5aUWa5. Estimation of clock drift in underwater acoustic instruments using propagation channel analysis. Ilya A. Udovydchenkov, Ballard J. Blair (The MITRE Corporation, 202 Burlington Rd., Bedford, MA 01730, ilya@whoi.edu), Ralph A. Stephen (Geology and Geophys., Woods Hole Oceanographic Inst., Woods Hole, MA), Peter F. Worcester, and Matthew Dzieciuch (Scripps Inst. of Oceanogr., La Jolla, CA)

All underwater acoustic sensors require accurate on-board clocks for subsequent data analysis and interpretation. Unfortunately, most clocks suffer from a phenomenon called "clock drift" (loss of accuracy), which occurs due to environmental changes, aging, and other factors. Typically, the clock drift is accounted for by calibrating the clock in the instrument before and after the deployment, and applying a correction during data post-processing. This method, however, does not allow accurate estimation of clock errors during a particular experimental event. In this presentation a small subset of data collected on the last day of the Bottom Seismometer Augmentation in the North Pacific (OBSANP) Experiment in June-July 2013 is analyzed. It is shown that advanced signal processing techniques can be used to accurately reconstruct the motion of the ship-suspended acoustic source, which, in turn, can improve the accuracy of the acoustic receivers deployed on the seafloor in the deep ocean. [Work supported by the MITRE Innovation Program and ONR.]

9:40

5aUWa6. Covet underwater acoustic communication based on spread spectrum orthogonal frequency division multiplexing (OFDM). Shaofan Yang, Zhongyuan Guo, Shengming Guo, Ning Jia, Dong Xiao, and Jianchun Huang (Key Lab. of Underwater Acoust. Environment, Inst. of Acoust., Chinese Acad. of Sci., No. 21 North Fourth Ring Rd., Beijing 100190, China, 1172966054@qq.com)

A covert underwater acoustic communication (UAC) method is realized with a dolphin whistle as the information carrier. The proposed method jointly utilizes the spread spectrum and orthogonal frequency division multiplexing (OFDM) techniques. The original dolphin whistle is represented as the complex baseband OFDM transmitted signal and the phase of its discrete Fourier transform (DFT) coefficients are modulated by a spread spectrum code to carry information. The audio quality and waveform similarity are used as two covert effect criterion. In simulations, the influences of the code length and modulation parameters are investigated. It is verified that the communication performance and camouflage effect are a pair of contradictions. Therefore, the best compromise formula should be chosen according to the actual demands.

10:00–10:20 Break

10:20

5aUWa7. High frequency underwater acoustic communication channel characteristics in the Gulf of Mexico. Aijun Song (Elec. and Comput. Eng., Univ. of Alabama, 114 Robinson Hall, Newark, DE 19716, ajsong@udel.edu)

In the applications of underwater acoustic communications, higher carrier frequencies support wider bandwidth, which is preferable for achieving higher data rates. At the same time, due to stronger attenuation, higher frequencies lead to shorter communication ranges. During past decades, a large number of efforts were devoted to investigate acoustic communications for the band of 8-50 kHz, over the medium range of several kilometers and beyond. Several efforts focused on the very high frequencies, greater than 200 kHz, for short communication ranges, tens or hundreds of meters. Here we consider a frequency band that falls between the well-studied high frequencies 8-50 kHz, for example from the Kauai Island experiment series, and the "unknown" very high frequencies (200 kHz or higher). A high frequency acoustic experiment was conducted in the northern Gulf of Mexico in August, 2016 to examine the operating ranges, data rates, and performance of acoustic communication systems at the carrier frequencies of 85 and 160 kHz. The received signal-to-noise ratios, channel coherence, and impulse responses will be reported between multiple transducers and a five-element hydrophone array. Communication performance will be reported as well.

10:40

5aUWa8. Directional received medium access control protocol for underwater sensor networks. Maochun Zheng (Harbin Eng. Univ., Harbin 150001, China, zmc2015@hrbeu.edu.cn)

Aiming at handshake protocols in the presence of hidden terminal and exposed terminal problems in underwater sensor networks of single-channel, this paper has presented the communication protocol of directional received slotted floor acquisition multiple access based on single vector sensor orientation. This protocol uses the characteristics of single vector sensor to directional receive, solves the problem of the hidden and exposed terminal problem in the single channel condition through the directional handshake mechanism. For the problem of near-far effect, each node maintains the neighbor node information table in one-hop range, calculates the power of transmitting RTS and CTS packets according to the current neighbor information, and then calculates the transmission power that satisfies the signal-to-noise ratio requirement of the destination node according to the obtained information. Compared with the single channel protocol based on omnidirectional antenna, this protocol can make full use of the resources of idle channel and the degree of network spatial reuse. Simulation experiments show that under the two different business, compared with the classical Slotted FAMA protocol and MACAW, the throughput of network has improved by 40%~60% and 30%~40%, respectively, which proves that the directional protocol can improve the network throughput effectively.

11:00

5aUWa9. Vertical multiuser communication using adaptive time reversal. Takuya SHIMURA, Yukihiro Kida, and Mitsuyasu Deguchi (Maine Tech. Development Dept., JAMSTEC, 2-15 Natsushima-cho, Yokosuka, Kanagawa 237-0061, Japan, shimurat@jamstec.go.jp)

Time reversal is an attractive solution to achieve channel equalization in a rich multipath environment such as underwater acoustic communication (UAC) channel because of its spatial and temporal focusing effect. Recently, demands for multiuser communication have increased in the field of UAC. In Japan Agency Marine-Earth Science and Technology (JAMSTEC), multiple autonomous underwater vehicles (AUVs) operation is planned to widen observation area. Adaptive time reversal is a method for space division multiplexing (SDM) based on its spatial focusing and nulling effect, and a promising solution for multiuser communication. In previous studies, it has been demonstrated that adaptive time reversal is very effective for "horizontal" multiuser communication, in which signals from multiple sources are received vertical receiver array. In this study, adaptive time reversal is applied for "vertical" multiuser communication, which is between a support vessel and multiple vehicles below the vessel. At-sea experiments were carried out in which signals from two sources near the seabed were measured at the receivers on the bottom of the research vessel. As results, adaptive time reversal is also effective for such vertical multiuser communication and has better performance than orthogonal frequency-division multiplexing (OFDM) with SDM combiners.

11:20

5aUWa10. Source localization in shallow waters with horizontal and vertical arrays. Yuqing Jia (Key Lab. of Underwater Acoust. Environment, Inst. of Acoust.,Chinese Acad. of Sci., No.21 North Fourth Ring Rd., Beijing 100190, China, 469120473@qq.com), Shengming Guo, and Lin Su (Key Lab. of Underwater Acoust. Environment,Inst. of Acoust.,Chinese Acad. of Sci., Beijing, China)

In order to develop the ability of localization of the underwater target in shallow sea, a method of localization with multi-array is proposed in this paper. This method employ matched field processing (MFP) combined with vertical line arrays (VLA) and horizontal line array (HLA) to sample the sound field in large scale to get the 3-D information of the target. The multi-array is built by take a full account the affect of waveguide in shallow sea comparing with single array, which can overcome the twin-line arrays port/starbord blur problem and achieve the estimation of range and azimuth effectively. The localization is realized by a MFP and the results from simulations and data processing is more reliable in complex shallow sea environment than with single array MFP.

11:40

5aUWa11. Specifics of DEMON acoustic signatures for large and small boats. Alexander S. Pollara (Maritime Security Ctr., Stevens Inst. of Technol., 940 Bloomfield St., Apt. 3, Hoboken, NJ 07030, apollara@stevens.edu), Gregoire Lignan (École Navale, Paris, France), Louis Boulange (École Navale, Lanvéoc, France), Alexander Sutin, and Hady Salloum (Maritime Security Ctr., Stevens Inst. of Technol., Hoboken, NJ)

Marine vessel propellers produce noise by the formation and shedding of cavitation bubbles. This process creates both narrow-band tones and broad-band amplitude modulated noise. The Detection of Envelope Modulation on Noise (DEMON) is an algorithm to determine the frequencies that modulate this noise. Results of DEMON processing depend on the selection of a ship noise frequency band to analyze. It is well known that the best passband to use may vary dramatically between vessels. Despite this, there has been no systematic investigation how the DEMON spectra depend of the carrier noise frequencies, and the modulation indices of vessel noise have not been investigated. We use a modification of the Cyclic Modulation Spectrum (CMS) to determine the modulation index of cavitation noise across the entire spectrum of carrier frequencies. We investigated how speed and vessel size affect the modulation index and carrier frequency of vessel noise. Several phenomena in the distribution of modulation indices for large and small boats were observed. These can be used for vessel classification. For small boats, the DEMON spectra have a different set of frequency peaks at various carrier frequencies. This is explained by the engine exhaust which produces amplitude modulated noise much like a propeller.

12:00

5aUWa12. Iterative source-range estimation in a sloping-bottom shallow-water waveguide using the generalized array invariant. Chomgun Cho, Hee-Chun Song (Scripps Inst. of Oceanogr., Univ. of California, San Diego, 9500 Gilman Dr., La Jolla, CA 92093-0238, hcsong@mpl.ucsd.edu), Paul Hursky (HLS Res. Inc., San Diego, CA), and Sergio Jesus (Univ. of Algarve, Faro, Portugal)

The array invariant proposed for robust source-range estimation in shallow water is based on the dispersion characteristics in ideal waveguides, utilizing multiple arrivals separated in beam angle and travel time for broadband signals. Recently, the array invariant was extended to general waveguides by incorporating the waveguide invariant β, referred to as a generalized array invariant. In range-dependent environments with a sloping bottom, the waveguide invariant β is approximately proportional to the source range via the water depth. Assuming knowledge of the bottom slope, the array invariant can be applied iteratively starting with $\beta=1$ in shallow water, which converges toward the correct source range in a zigzag fashion. The iterative array invariant approach is demonstrated in a sloping-bottom shallow-water waveguide using a short-aperture vertical array (2.8 m) from the Random Array of Drifting Acoustic Receivers 2007 experiment (RADAR07), where a high-frequency source (2-3.5 kHz) close to the surface (6-m) was towed between 0.5 to 5 km in range with the corresponding water depth being 80 and 50 m, respectively.

THURSDAY MORNING, 29 JUNE 2017 ROOM 309, 8:40 A.M. TO 12:20 P.M.

Session 5aUWb

Underwater Acoustics, Acoustical Oceanography, and ASA Committee on Standards: Underwater Noise From Marine Construction and Energy Production III

James H Miller, Cochair
Ocean Engineering, University of Rhode Island, 215 South Ferry Road, Narragansett Bay Campus URI, Narragansett, RI 02882

Paul A. Lepper, Cochair
EESE, Loughborough University, Loughborough LE113TU, United Kingdom

Invited Papers

8:40

5aUWb1. The underwater sound field from impact pile driving: Observations and modeling of key features in the time domain. Peter H. Dahl and David R. Dall'Osto (Appl. Phys. Lab. and Mech. Eng., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, dahl@apl.washington.edu)

Observations of the pressure time series made from a vertical line array at range of order 10 waveguide depths from the pile source are presented along with interpretive modeling. Modeling is based on conceptualizing underwater sound from impact pile driving as originating from a vertical distribution of harmonic point sources uniformly distributed along the wetted-length of the pile, each source phased prior to summation in the frequency domain. This leads to credible representations of pressure time series as function of range and depth that are linearly related to field measurements. In particular, precursor arrivals which arrive prior to any water borne arrival as result of having propagated through a higher-speed sediment, and deterministic manifestations of the Mach cone effect within the main pressure time series, are shown in comparison with modeling results. Both phenomena are influenced by the increasing sediment excursion of the pile, and the importance of the sound speed gradient and sediment attenuation to the precursor will be shown. Finally, the synthetic results are used to generate acoustic streamlines that trace the pathway of (short-time-averaged) active acoustic intensity associated with the precursor. Results from this work suggest the precursor amplitude sets the performance bound of any sound mitigation strategy.

9:00

5aUWb2. Experimental validation of models for prediction of marine pile driving sound. Roel A. Müller, Marten Nijhof, Bas Binnerts (TNO, Oude Waalsdorperweg 63, Den Haag 2597 AK, Netherlands, roel.muller@tno.nl), Christ A. de Jong, Michael A. Ainslie (TNO, The Hague, Netherlands), and Erwin Jansen (TNO, Den Haag, Netherlands)

Various models for the underwater noise radiation due to marine pile driving are being developed worldwide, to predict the sound exposure of marine life during pile driving activities. However, experimental validation of these models is scarce, especially for larger distances. Recently, TNO has been provided with data from underwater noise measurements up to 65 km from the piling location, gathered during the construction of two wind farms in the Dutch North Sea. These measurement data have been compared with different modeling approaches, in which the sound source is either formulated as an equivalent point source, or as a axially symmetric finite element model of the pile including the surrounding water and sediment. Propagation over larger distances, with varying bathymetry, is modeled efficiently by either an incoherent adiabatic normal mode sum or a flux integral approach. Differences between simulation and measurement data are discussed in terms of sound exposure level and spectral content, which leads to more insight into the mechanisms of sound radiation and propagation that are relevant during marine piling activities. An overview is given of the merits, shortcomings, and possibilities for improvement of the models.

Contributed Papers

9:20

5aUWb3. Hydroacoustic measurements and modeling of pile driving operations in Ketchikan, Alaska. Graham A. Warner (JASCO Appl. Sci., 2305-4464 Markham St., Victoria, BC V8Z 7X8, Canada, graham.warner@jasco.com), Melanie Austin (JASCO Appl. Sci., Anchorage, AK), and Alexander MacGillivray (JASCO Appl. Sci., Victoria, BC, Canada)

Underwater acoustic measurements of pile driving operations were made at the Ketchikan ferry terminal in July 2016. At the time of the measurements, marine mammal injury and disturbance criteria developed by the National Marine Fisheries Service (NMFS) were based on sound pressure level (SPL) thresholds. Shortly after the measurements, NMFS changed the injury thresholds to a dual criteria involving peak pressure and sound exposure levels specific to marine mammal functional hearing groups. This paper presents distances to both injury criteria and the (unchanged) SPL-based disturbance criteria for vibratory driving and impact hammering 30-inch diameter cylindrical piles. Threshold distances were obtained using empirical regressions of sound levels measured by seabed-mounted recorders at 10 and 1000 m nominal range. A finite-difference method pile driving source model was used with a parabolic equation propagation model to compare measurements with simulations and to estimate received levels at all ensonified locations in the complex bathymetric environment of the Tongass Narrows. Measured and modeled results show the importance of hydrophone placement with respect to the Mach cone and near-pile bathymetry.

9:40

5aUWb4. On the airborne contribution to the underwater sound-field from marine pile installation. David R. Dall'Osto (Appl. Phys. Lab., Univ. of Washington, 1013 N 40th St., Seattle, WA 98105, dallosto@apl.washington.edu) and Peter H. Dahl (Appl. Phys. Lab. and Dept. of Mech. Eng., Univ. of Washington, Seattle, WA)

Airborne sound generated by the impact hammer used for marine pile installation contributes to the total underwater sound field. This contribution is distinct from the sound generated underwater, e.g., by the up-and-down going Mach waves generated by the pile underwater. Airborne sound transmission into the water occurs within a 26 deg. cone directly below the pile-hammer. As the pile is driven deeper, the distance between the hammer and the water surface decreases producing a time-dependent signature in the observed underwater sound pressure that correlates directly to the observed pile-depth monitored during installation. Additionally, the area subtended by the cone of transmission decreases as the pile is driven deeper, which reduces the duration of the air-borne contribution which is longest at the beginning of the pile installation. In this presentation, a model for the airborne contribution is examined alongside data that was measured at ranges of approximately 2 and 10 times the water depth. In addition to basic interpretation of the time signature from impact pile driving, the observed transmission of high-intensity airborne sound into the water during pile installation has implications for the contributions of airborne noise associated with wind-turbines.

Invited Papers

10:00

5aUWb5. Depth dependence of pile driving noise measured at the research platform FINO3. Frank Gerdes (WTD 71, Berliner Straße 115, Eckernförde 24340, Germany, frankgerdes@bundeswehr.org)

Impact pile driving is a source of high-amplitude underwater sound that can propagate large distances and may have a negative impact on marine fauna. In Germany, wind farm operators are required to monitor underwater sound levels and to ensure that the levels do not exceed certain values. Mainly because of practical reasons sound measurements are usually performed with hydrophones about 2 to 3 m above the sea-floor. It is of some interest to know whether these sound values are representative for the entire water column. We investigated this with an underwater measurement system that was designed to provide simultaneous sound measurements at up to eight different heights above the sea-floor. It consisted of a bottom mounted tripod to which a vertical chain of five to six hydrophones was attached with the top-most hydrophone being located about four meters below the sea-surface. The system was cable-connected to the research platform FINO3. This paper analyses the depth dependence of pile driving noise that was emitted from piling operations in the nearby offshore wind-farm DanTysk with propagating distances between pile and FINO3 ranging between 3km and 15km. Observed sound exposure levels (SEL) were usually somewhat larger at 2 m above the sea-floor than at hydrophones higher in the water column. Details about the variability and frequency dependence of the observed differences are presented.

10:20–10:40 Break

10:40

5aUWb6. Overview of underwater acoustic and seismic measurements of the construction and operation of the Block Island Wind Farm. James H Miller, Gopu R. Potty (Ocean Eng., Univ. of Rhode Island, 215 South Ferry Rd., Narragansett Bay Campus URI, Narragansett, RI 02882, miller@uri.edu), Ying-Tsong Lin, Arthur Newhall (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., Woods Hole, MA), Kathleen J. Vigness-Raposa (Marine Acoust., Inc., Middletown, RI), Jennifer Giard (Marine Acoust., Inc., Narragansett, RI), and Tim Mason (Subacoustech, Ltd., Southampton, United Kingdom)

The Block Island Wind Farm (BIWF), the first offshore wind farm in the United States, consists of five 6-MW turbines 3 miles southeast of Block Island, Rhode Island in water depths of approximately 30 m. Construction began in the summer of 2015 and power production began in late 2016. Underwater acoustic and geophysical measurement systems were deployed to acquire real-time observations of the construction and initial operation of a wind facility to aid the evaluation of environmental effects of future facilities. The substructure for these BIWF turbines consists of jacket type construction with piles driven to the bottom to pin the structure to the seabed. The equipment used to monitor construction and initial operation consisted of a towed array consisting of eight hydrophones, two fixed moorings with four hydrophones each and a fixed sensor package for measuring particle velocity. This sensor package consists of a three-axis geophone on the seabed and a tetrahedral array of four low sensitivity hydrophones at 1 m from the bottom. Additionally, an acoustic vector sensor was deployed in mid-water. Data collected on these sensor systems during construction and initial operation will be summarized. [Work supported by Bureau of Ocean Energy Management (BOEM).]

11:00

5aUWb7. Measurements of particle motion near the seafloor during construction and operation of the Block Island Wind Farm. Gopu R. Potty, Makio Tazawa, Jennifer Giard, James H Miller (Dept. of Ocean Eng., Univ. of Rhode Island, 115 Middleton Bldg., Narragansett, RI 02882, potty@egr.uri.edu), Ying-Tsong Lin, Arthur Newhall (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., Woods Hole, MA), and Kathleen J. Vigness-Raposa (2 Corporate Pl., Ste. 105, Marine Acoust., Inc., Middletown, RI)

Noise radiation and particle motion from pile driving activities were monitored using multiple sensors during the construction of the first offshore wind farm off Block Island, RI, USA in 2016. The Block Island Wind Farm (BIWF) consists of five turbines in water depths of approximately 30 m. The substructure for these turbines consists of jacket type construction with piles driven to the bottom to pin the structure to the seabed. Pile driving operations generate intense sound, impulsive in nature, which radiates into the surrounding air, water and sediment producing particle motion that may affect marine animals. The particle velocity sensor package consists of a three-axis geophone on the seabed and a tetrahedral array of four low sensitivity hydrophones at 1 m from the bottom. The acoustic pressure acquired by the hydrophones will be processed to calculate particle motion. Data from the BIWF site will be compared with model predictions and published data from other locations. Recent measurements from the same wind farm location during the operational phase also will be presented. [Work supported by Bureau of Ocean Energy Management (BOEM).]

11:20

5aUWb8. A preliminary numerical model of three-dimensional underwater sound propagation in the Block Island Wind Farm area. Ying-Tsong Lin, Arthur Newhall (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., Bigelow 213, MS#11, WHOI, Woods Hole, MA 02543, ytlin@whoi.edu), Gopu R. Potty, and James H Miller (Dept. of Ocean Eng., Univ. of Rhode Island, Narragansett, RI)

The Block Island Wind Farm, consisting of five 6-MW turbines, is the first U.S. commercial offshore wind farm harvesting wind energy to generate electricity, located 3.8 miles southeast of Block Island, Rhode Island. In-situ underwater and airborne noise measurements were made during the construction and the first two months of the operational period for the purpose of environmental impact assessment. To better interpret the noise measurements and extend the noise propagation prediction beyond the coverage of listening stations, a three-dimensional underwater sound propagation model is created with a high resolution bathymetric map and a data-assimilated ocean dynamic model. The bathymetric map is made using the 3 arc-second U.S. Coastal Relief Model (CRM) with a 100-m horizontal resolution provided by the National Centers for Environmental Information (NCEI). The ocean model is extracted from the Regional Ocean Modeling System (ROMS) ESPreSSO (Experimental System for Predicting Shelf and Slope Optics) model covering the Mid-Atlantic Bight with a 5 km horizontal resolution and 36 terrain-following vertical levels. Temporal and spatial variability of noise propagation conditions is identified in the integrated acoustic and oceanographic model. Future model development incorporating surface wind waves and sub-bottom sediment layer structure will be discussed. [Work supported by BOEM.]

11:40

5aUWb9. Variations in the acoustic field recorded during pile-driving construction of the Block Island Wind Farm. Kathleen J. Vigness-Raposa (Marine Acoust., Inc., 2 Corporate Pl., Ste. 105, Middletown, RI 02842, kathleen.vigness@marineacoustics.com), Jennifer Giard (Marine Acoust., Inc., Narragansett, RI), Adam S. Frankel (Marine Acoust., Inc., Arlington, VA), James H Miller, Gopu R. Potty (Dept. of Ocean Eng., Univ. of Rhode Island, Narragansett, RI), Ying-Tsong Lin, Arthur Newhall (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., Woods Hole, MA), and Tim Mason (Subacoustech, Ltd., Southampton, United Kingdom)

The Block Island Wind Farm, the first offshore wind farm in the United States, consists of five 6-MW turbines three miles southeast of Block Island, Rhode Island in water depths of approximately 30 m. The turbines include a jacket-type substructure with four piles driven at an angle of approximately 13 deg to the vertical to pin the structure to the seabed. The acoustic field was measured during pile driving of two turbines in September 2015 with an 8-element towed horizontal line array. Measurements began at a range of 1 km from the turbine on which piling was occurring and extended to a range of 8 km from the construction. The peak-to-peak received level, sound exposure level, and kurtosis from each pile strike were determined as a function of range from the pile. The ambient noise just prior to each signal was also measured to calculate signal-to-noise ratio values. Results provide insight into the transition from fast-rise-time impulsive signals at close range to slow-rise-time non-impulsive signals at longer ranges. In addition, the variability among signals at the same range is being characterized as a function of pile and hammer strike characteristics. [Work supported by Bureau of Ocean Energy Management (BOEM).]

Contributed Paper

12:00

5aUWb10. Hydroacoustic measurements during construction of the first US offshore windfarm—Methodologies to address regulatory requirements. Erik J. Kalapinski and Kristjan Varnik (Energy Programs, Tetra Tech, Inc., 160 Federal St., Fl. 3, Boston, MA 02210, erik.kalapinski@tetra-tech.com)

The regulations governing underwater noise from offshore wind farm development in the United States have not been as explicit as in other countries. The Block Island Wind Farm represents the first case study. In this context, it is important to disseminate information about the relevant noise sources, address evolving guideline criteria, and develop noise measurement and analysis procedures to address regulatory reporting requirements. Tetra Tech led the hydroacoustic monitoring program which occurred in two distinct stages. The first involved short-term monitoring of the installation of the initial wind turbine generator foundation using both mobile real-time and static monitoring techniques used for daily reporting. Long-term monitoring of the remaining four foundations with static recorders documented the inherent variability in the data set. Received sound levels measured at pre-determined distances were used to assess site-specific propagation characteristics and to verify ranges to the relevant sound exposure thresholds. This involved the evaluation of multiple metrics including the apparent sound source level of pile-driving activities and the confirmation of the Exclusion and Monitoring Zone established to ensure the protection of marine life. All of the monitoring objectives were met, including the field verification of modeling results established during the environmental permitting process.

THURSDAY AFTERNOON, 29 JUNE 2017 ROOM 207, 1:15 P.M. TO 4:40 P.M.

Session 5pAAa

Architectural Acoustics: Architectural Acoustics and Audio: Even Better Than the Real Thing III

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

Alexander U. Case, Cochair
Sound Recording Technology, University of Massachusetts Lowell, 35 Wilder St., Suite 3, Lowell, MA 01854

Wolfgang Ahnert, Cochair
Ahnert Feistel Media Group, Arkonastr. 45-49, Berlin D-13189, Germany

Chair's Introduction—1:15

Contributed Paper

1:20

5pAAa1. Multidimensional scaling approach in room acoustic evaluation. Pawel Malecki and Jerzy Wiciak (Dept. of Mech. and VibroAcoust., AGH - Univ. of Sci. and Technol., al. Mickiewicza 30, Kraków 30-059, Poland, pawel.malecki@agh.edu.pl)

In the event of natural acoustics recordings, it is essential to decide on the room and, consequently, on the reverb. The decision is very important, as the consequences are not possible to be changed in the process of mixing. Therefore, proper selection of the parameter is often essential to achieve the effect desired. The aim of the study is an attempt to answer the question what is the optimal or preferred by an average listener reverberation time and other room acoustic parameters in choral music. As part of the study, sound samples of choral music that differed only by the reverb were made. The samples were tested using the procedures of multidimensional scaling. Ambisonics, altogether with VBAP, was used in the listening test.

Invited Papers

1:40

5pAAa2. Enhancing home recording spaces through architectural acoustics design. Sebastian Otero (Architectural Acoust., Acustic-O, Laurel 14, San Pedro Martir, Tlalpan, Mexico, D.F. 14650, Mexico, sebastian@acustic-o.com)

The demand for home recording spaces has increased in the past years. They vary in terms of privacy, comfort, size, audio quality, budget, type of materials, acoustic treatments, types of usage and equipment. Although it is hard to unify the concept of "home studio," there are certain architectural acoustics criteria that should be considered in order to guarantee the use of the space for creative and technical purposes. This paper analyzes different cases to demonstrate how home recording is enhanced applying these architectural acoustic principals.

2:00

5pAAa3. When the difference between "good" and "great" is 0.5dB—Ponderings of a "Tuning Conductor." Christopher Blair (Akustiks, LLC, 93 North Main St., Norwalk, CT 06854, cblair@akustiks.com)

For many years the author has been privileged to work closely with music directors, their orchestras, and soloists in optimizing the acoustic environment in numerous rooms employing acoustic enhancement. During the process he has learned that even very small changes in energy levels and timing of virtual reflections can make a profound difference in perceived acoustic quality, largely due to masking effects, both in the audience and on the podium. The quick adjustments that can be made utilizing such systems not only make A/B comparisons useful in assessing relative quality, but also as an educational tool informing the designer as to what specific attributes in an impulse response are helpful. This presentation contains a number of illustrative vignettes from his experience as both an acoustician and conductor, including the notion that sometimes the most effective changes to acoustical perception in a concert hall can come from changing how the orchestra plays.

Contributed Paper

2:20

5pAAa4. An integrated passive and active room acoustics + sound reinforcement system design solution for a large worship space. David Kahn (Acoust. Disctinctions, 400 Main St., Ste. 600, Stamford, CT 06901, dkahn@ad-ny.com)

The United Methodist Church of the Resurrection, with more than 16,000 adult members and an average weekly worship attendance of more than 8,600, recently completed a new 3,500-seat worship space. The new sanctuary building has an ellipsoidal plan shape and a very tall ceiling to address an important programmatic goal to achieve "A Sense of Majesty". Acoustical goals included warm speech throughout all seating areas with acoustics inspiring to create a high level of energy therefore encourage high level of participation. Furthermore, the church wants to support both traditional (orchestra and choir) and contemporary (Praise band with vocalists) styles of music in this new worship space. These programming goals, to some extent, are in conflict with one another. Particularly the architectural goal of having a soaring space is in conflict with the goals for a space that provides acoustical support of corporate worship. Since the walls and ceiling are too far away from most of the seating area to provide beneficial sound reflections to support corporate worship and to support traditional music, an electronic enhancement system was provided to electronically create the early sound reflections that could be created by a lower ceiling.

Invited Papers

2:40

5pAAa5. So you wanna be a rock "n" roll star (there's an app for that). Sam Ortallono (Visual Performing Arts, Lee College, 711 W. Texas Ave., Baytown, TX 77522, sortallono@lee.edu)

With the increase of mobile recording technology more recordings are made in unconventional, acoustically uncontrolled environments. We designed an experiment to compare some of these spaces. At Lee College, we recorded the same song three times. The same musicians were recorded in three different spaces using different technology; in a large, controlled studio using ProTools, in a home studio setting using a laptop and in a living space using a mobile telephone application. Once the songs were mixed and mastered, college students were asked to rate the quality of the mixes.

3:00–3:20 Break

3:20

5pAAa6. Nuance is dead (or What You Will). Scott D. Pfeiffer (Threshold Acoust. LLC, 53 West Jackson Blvd., Ste. 815, Chicago, IL 60604, spfeiffer@thresholdacoustics.com)

Live performance is about connection. Traditional amplification at its least effective introduces microphones and loudspeakers in between the artist with a message and the audience intending to show appreciation for the effort. The authenticity of performance in the right setting where the mircophones and loudspeakers are unnecessary can be deeply moving, and provides for an unambiguous exchange between performer and audience when the quality of the environment for the purpose is up to the challenge. The "Unplugged" movement in popular music of the early 90s attempted to bridge this gap, though clearly none of the acts was truly "Unplugged." For all of acoustic design history, architectural or electronic technology to enhance the natural acoustic connection between performer and audience. Thanks to the efforts, we can replicate aspects of natural acoustics of a human scale—and the connection created by the shared environment—to make an under-performing acoustic environment better, a great acoustic environment more flexible, or simply create an interior acoustic where none exists. Applications and limitations of the available technologies for fictionalizing acoustic traits are explored to help recognize when the technology, architectural or electronic, becomes a distraction from the purpose.

3:40

5pAAa7. Sound systems in reverberant spaces: Approaches in practice. David S. Woolworth (Roland, Woolworth, & Assoc., LLC, 356 CR 102, Oxford, MS 38655, dave@oxfordacoustics.com)

This paper presents three separate case studies of the pairing of sound systems and reverberant spaces, including existing and retrofitted spaces. The design or troubleshooting utilizes various approaches to meet the challenges of reverberant spaces in regard to acoustics, programming of the space, and the use of loudspeakers and processing.

4:00

5pAAa8. Using the extended techniques on the trombone to demonstrate many of the basic principles of music acoustics. Thomas J. Plsek (Brass/Liberal Arts, Berklee College of Music, MS 1140 Brass, 1140 Boylston St., Boston, MA 02215, tplsek@berklee.edu)

The author has realized that the trombone can be used as a low tech device to demonstrate many of the principles of acoustics. This is especially true when one considers all the extended techniques that have been developed for the instrument. A sampling of them could include (but not be limited to) multiphonics creating beats, playing lip buzzes through the various parts of the instrument giving audible information about the signal processing path, mouthpieces slaps to illustrate resonance, inhaling while playing to put an interesting take on brass pedagogy, showing how the water key can indicate the presence of nodes and antinodes and how the instrument provides very little feedback in the extreme high register. Live demonstrations will be presented.

4:20

5pAAa9. Even weirder than the real thing—Gated reverb history and aesthetics. Alexander U. Case (Sound Recording Technol., Univ. of Massachusetts Lowell, 35 Wilder St., Ste. 3, Lowell, MA 01854, alex@fermata.biz)

Sound presented via loudspeaker may take advantage of signal processing to create sounds not possible in an all-acoustic production chain. Breaking free of the acoustic constraints for music-making in the concert hall to take advantage of an analog, digital and electro-acoustic production chain has been a major attraction for many popular recording artists. New aesthetics evolved. Among the most absurd sonic concoctions to come from this, gated reverb is part discovery, and part invention, motivated by misunderstandings, and driven by plain old rock and roll rebellion. This paper tours the development of gated reverb, with audio illustrations, and makes the case for its continued use today.

THURSDAY AFTERNOON, 29 JUNE 2017 ROOM 208, 1:20 P.M. TO 4:20 P.M.

Session 5pAAb

Architectural Acoustics: Simulation and Evaluation of Acoustic Environments IV

Michael Vorländer, Cochair
ITA, RWTH Aachen University, Kopernikusstr. 5, Aachen 52056, Germany

Stefan Weinzierl, Cochair
Audio Communication Group, TU Berlin, Strelitzer Str. 19, Berlin 10115, Germany

Ning Xiang, Cochair
School of Architecture, Rensselaer Polytechnic Institute, Greene Building, 110 8th Street, Troy, NY 12180

Invited Paper

1:20

5pAAb1. First international round robin on auralization: Results of the perceptual evaluation. Fabian Brinkmann, David Ackermann (Audio Commun. Group, Tech. Univ. Berlin, Einsteinufer 17c, Berlin 10787, Germany, fabian.brinkmann@tu-berlin.de), Lukas Aspöck (Inst. of Tech. Acoust., RWTH Aachen Univ., Aachen, Germany), and Stefan Weinzierl (Audio Commun. Group, Tech. Univ. Berlin, Berlin, Germany)

The round robin on auralization aimed at a systematic evaluation of room acoustic modeling software by means of comparing simulated and measured impulse responses. While a physical evaluation by means room acoustical parameters and spectro-temporal comparisons is addressed in an accompanying talk, here, we focus on an evaluation of perceptual differences arising in complex room acoustical scenarios. In these cases, a mere physical evaluation might not be able to predict the perceptual impact of the manifold interaction of different sound propagation phenomena in enclosed spaces such as reflection, scattering, diffraction, or modal behavior. For this purpose, dynamic auralizations of binaural room impulse responses that were simulated with different room acoustical modeling softwares were evaluated against their measured counterparts. For this purpose, listening tests were conducted using "plausibility" and "authenticity" as overall quality criteria and the Spatial Audio and Quality Inventory (SAQI) for a differential diagnosis of the remaining differences.

Contributed Paper

1:40

5pAAb2. Evaluation of a Loudspeaker-Based Virtual Acoustic Environment for Investigating sound-field auditory steady-state responses. Valentina Zapata-Rodriguez (InterAcoust. Res. Unit, InterAcoust. A/S, c/o: DTU, Ørsteds Plads B352, R027, Kgs. Lyngby 2800, Denmark, valr@iru.interacoustics.com), Gerd Marbjerg (Acoust. Technol., Tech. Univ. of Denmark, Kongens Lyngby, Denmark), Jonas Brunskog, Cheol Ho Jeong (Acoust. Technol., Tech. Univ. of Denmark , Kgs. Lyngby, Denmark), Søren Laugesen, and James M. Harte (InterAcoust. Res. Unit, InterAcoust. A/S, Kgs. Lyngby, Denmark)

Measuring sound-field auditory steady-state responses (ASSR) is a promising new objective clinical procedure for hearing aid fitting validation, particularly for infants who cannot respond to behavioral tests. In practice, room acoustics of non-anechoic test rooms can heavily influence the auditory stimulus used for eliciting the ASSR. To systematically investigate the effect of the room acoustics conditions on sound-field ASSR, a loudspeaker-based auralization system was implemented using a mixed order Ambisonics approach. The present study investigates the performance of the auralization system in terms of objective room acoustic measurements and sound-field ASSR measurements, both in the actual room and in the simulated and auralized room. The evaluation is conducted for a small room with well-defined acoustic properties. The room is carefully modeled using the novel room acoustic simulation tool PARISM (Phased Acoustical Radiosity and Image Source Method) and validated through measurements. This study discusses the limitations of the system and the potential improvements needed for a more realistic sound-field ASSR simulation.

Invited Papers

2:00

5pAAb3. Auditory Illusion over Headphones Revisited. Karlheinz Brandenburg (Fraunhofer IDMT, Ehrenbergstr. 31, Ilmenau 98693, Germany, bdg@idmt.fraunhofer.de), Florian Klein, Annika Neidhardt, and Stephan Werner (Technische Universität Ilmenau, Ilmenau, Germany)

Plausibility and immersion are two of the keywords which describe quality features of virtual and augmented reality systems. There is a plethora of research results in this area, but current headphone based systems still do not enable auditory illusion for everybody and all types of signals. To address the open questions, a series of studies have been conducted to study the quality of spatial audio reproduction using binaural synthesis. This contribution gives a revisited insight in the creation of a perfect auditory illusion via headphone. First, a summary on technical parameters to realize correct auditory cues is given, which includes requirements like headphone equalization or interpolation methods. Second, we will point out that beyond reproducing the physically correct sound pressure at the ear drums, more effects play a significant role in the quality of the auditory illusion *(and can be dominating in some cases and overcome physical deviation).* Perceptual effects like the room divergence effect, additional visual influences, personalization, pose and position tracking as well as adaptation processes are discussed. The single effects are described and the interconnections between them are highlighted.

2:20

5pAAb4. Interactive reproduction of virtual acoustic environments for the evaluation of hearing devices—Methods and validation. Giso Grimm and Volker Hohmann (Medizinische Physik and Cluster of Excellence Hearing4all, Universität Oldenburg, Germany, Carl-von-Ossientzky Universitaet, Oldenburg 26111, Germany, volker.hohmann@uni-oldenburg.de)

Virtual acoustic environments are increasingly used for evaluating hearing devices in complex acoustic conditions. In this talk we propose an interactive simulation method via multi-channel loudspeaker systems or headphones. The method focuses on the time-domain simulation of the direct path and a geometric image source model, which simulates air absorption and in case of motion the Doppler effect of all primary and image sources. To establish the feasibility of this approach, the interaction between reproduction method and technical and perceptual hearing aid performance measures was investigated using computer simulations. Three spatial audio reproduction methods were compared in regular circular loudspeaker arrays with 4 to 72 channels. The influence of reproduction method and array size on performance measures of multi-microphone hearing aid algorithms was analyzed. In addition to the analysis of reproduction methods, algorithm performance was tested in a number of different virtual acoustic environments in order to assess the underlying factors of decreased hearing aid performance in complex environments. The results confirm previous findings that spatial complexity has a major impact on hearing aid benefit, and demonstrate the potential of virtual acoustic environments for hearing aid evaluation.
[Funded by DFG FOR1732 "Individualized hearing acoustics."]

2:40

5pAAb5. Perceptually motivated sound field synthesis for music presentation. Tim Ziemer (Inst. of Systematic Musicology, Univ. of Hamburg, Neue Rabenstr. 13, Hamburg 20354, Germany, tim.ziemer@uni-hamburg.de)

The conceptualization and implementation of a psychoacoustic sound field synthesis system for music is presented. Critical bands, the precedence effect, and integration times of the auditory system as well as the radiation characteristics of musical instruments are implemented in the signal processing. Interaural coherence, masking and auditory scene analysis principles are considered as well. The sound field synthesis system creates a natural, spatial sound impression and precise source localization for listeners in extended listening area, even with a low number of loudspeakers. Simulations and a listening test provide a proof of concept. The method is particularly robust for signals with impulsive attacks and quasi-stationary phases as in the case of many instrumental sounds. It is compatible with many loudspeaker setups, such as 5.1, ambisonics systems and loudspeaker arrays for wave front synthesis. The psychoacoustic sound field synthesis approach is an alternative to physically centered wave field synthesis concepts and conventional stereophonic sound and can benefit from both paradigms. Additional psychoacoustic quantities that have the potential to be implemented in the presented and other audio systems are discussed.

3:00–3:20 Break

Contributed Paper

3:20

5pAAb6. On the numerical simulation of natural acoustic sound sources. David Ackermann, Christoph Böhm, and Stefan Weinzierl (Audio Commun. Group, TU Berlin, TU Berlin, EN-8, Einsteinufer 17c, Berlin 10587, Germany, david.ackermann@tu-berlin.de)

A convincing auralization of acoustical scenes does not only require a proper modeling of the sound propagation between source and receiver, but also an appropriate representation of the acoustical source itself. While the properties of electro-acoustic sources can be well represented by advanced loudspeaker formats with high resolution, the complex, time-variant behavior of natural acoustic sources such as speakers, singers or musical instruments is not in any way considered by current techniques for acoustical simulation and auralization. In the talk, we will present measurement results of the sound power and directivity of natural acoustic sound sources and their dependence on pitch and dynamic level, as well as measurements of typical movements of the source during musical performances. We will demonstrate the physical and perceptual relevance of these effects both in the direct field and in room acoustical environments based on a technical and perceptual evaluation, and discuss new approaches to include these effects in numerical simulations.

Invited Papers

3:40

5pAAb7. A loudspeaker orchestra for opera houses studies. Dario D'Orazio, Luca Barbaresi, and Massimo Garai (DIN, Univ. of Bologna, Viale Risorgimento, 2, Bologna 40128, Italy, dario.dorazio@unibo.it)

A "Loudspeaker Orchestra" is an array of loudspeakers with a well-defined setup and layout. Initially proposed by contemporary composers for innovative performances (e.g., the "Acousmonium" used by the Groupe de Recherches Musicales in the 1970s), in the last years, the Loudspeaker Orchestra has been used for MIMO acoustic measurements in concert halls. In the present work, a Loudspeaker Orchestra for measurements in opera houses is proposed, taking into account the acoustic differences between a concert hall and an opera house. In fact, in a concert hall, the orchestra plays on the stage, while in an opera house, the orchestra plays in the pit, with a different layout, a smaller instrumental ensemble, etc. In a concert hall, the soloists are placed near the conductor, while in an opera house they move all over the stage, due to dramatic reasons. Basing on previous studies on room criteria and experiences with real orchestras, the authors proposed a Loudspeaker Orchestra layout for opera houses. In this work, tests on various configurations are presented, comparing measurements and numerical simulations.

4:00

5pAAb8. Acoustical evaluation of the Teatro Colón of Buenos Aires. Gustavo J. Basso (Facultad de Bellas Artes, Universidad Nacional de La Plata, Argentina, CALLE 5 N° 84, LA PLATA, Buenos Aires 1900, Argentina, gustavobasso2004@yahoo.com.ar)

It is been said that the Teatro Colón of Buenos Aires is one of the best halls for opera and symphonic music of the world, and this statement is made mainly for its wonderful acoustics. During the restoration works carried out between 2006 and 2010, we made a lot of studies in order to preserve the acoustical quality of the theater. Among them, we can mentioned acoustical measurements, architectonical examination, statistical analysis, aural descriptions and the development of a digital model of the space. None of the tools used for evaluating the acoustical quality of the hall, mainly based on the parameters of the ISO-3382 Standard, like the acoustical systems proposed by Leo Beranek and Yoichi Ando, was enough to explain the true quality of the room, which was deduced from opinion polls about the perceived sound by the audience. In order to find the causes of its acoustical behavior, we developed an architectural and acoustical study of the hall with the aid of a digital model that can explain, if not all, some of the main characteristics of its particular acoustical field. A synthesis of the results of this evaluation is described in this paper.

Session 5pAAc

Architectural Acoustics: Recent Developments and Advances in Archeo-Acoustics and Historical Soundscapes IV

David Lubman, Cochair
DL Acoustics, 14301 Middletown Ln., Westminster, CA 92683-4514

Miriam A. Kolar, Cochair
Architectural Studies; Music, Amherst College, School for Advanced Research, 660 Garcia St., Santa Fe, NM 87505

Elena Bo, Cochair
DAD, Polytechnic Univ. of Turin, Bologna 40128, Italy

Chair's Introduction—1:35

Invited Papers

1:40

5pAAc1. Rock art and prehistoric soundscapes: Some results from Italy, France, and Spain. Margarita Díaz-Andreu (Història i Arqueologia, ICREA, Universitat de Barcelona, Història i Arqueologia, Facultat de G. i Història, Carrer de Montalegre 6, Barcelona 08001, Spain, m.diaz-andreu@ub.edu) and Tommaso Mattioli (Història i Arqueologia, Universitat de Barcelona, Barcelona, Spain)

Under the axis of the SONART project—Sounds of Rock Art. Archaeoacoustics and post-palaeolithic Schematic art in the Western Mediterranean—a series of acoustic tests have been undertaken in seven rock art areas of Italy, France and Spain. The early chronology of this art—Neolithic and Chalcolithic—makes that no information exists about the reasons prehistoric peoples had to produce the art and the beliefs surrounding its creation. This means that in order to check whether sound is related to this cultural manifestation, formal methods need to be used. This also affects the analysis of sound for, in contrast to other areas of the world such as the circumpolar area, no legends or myths can be found to explain the link between art and special reverberation or echoes. Related to the latter, the results of our experiments to assess the direction of arrival (DOA) of echoes will be explained for the rock art landscapes of Valle d'Idivoro (Italy) and Baume Brune (France). The location of rock art in sites where the audibility of the landscape is exceptionally optimal will also be analyzed for the case of the Arroyo de San Serván landscape area in Extremadura (Spain).

2:00

5pAAc2. Were palaeolithic cave paintings placed because of acoustic resonances? Bruno M. Fazenda (Acoust. Res. Ctr., Univ. of Salford, The Crescent, Salford, Manchester M5 4WT, United Kingdom, b.m.fazenda@salford.ac.uk)

Previous archaeoacoustics work published from the 1980s to the 2000s has suggested that the location of palaeolithic paintings in French caves, such as Le Portel, Niaux, Isturitz, and Arcy-sur-Cureis, are associated with the acoustic response of those locations, particularly with strong low frequency resonances. Recent work done in caves in the Asturian and Cantabrian regions of Northern Spain has shown some evidence of statistical association between paintings dated from the Aurignacian/Gravettian period (cf. 42,000-25,000 BP) and the existence of acoustic responses which exhibit resonant artifacts. The work presented in this paper reports on a further analysis of the data that explores the association in more detail. A number of metrics focused specifically on low frequency response are used as factors to form statistical models that explain the position of paintings within the caves studied. The results of this study further our understanding on how perception of acoustic response might have played a part in modulating the expressive behavior of our ancestors.

2:20

5pAAc3. Did Paleolithic cave artists intentionally paint at resonant cave locations? David Lubman (DL Acoust., 14301 Middletown Ln., Westminster, CA 92683-4514, dlubman@dlacoustics.com)

In 1988, two investigators (Iegor Reznikoff and Michel Dauvois) reported a connection between the local density of cave paintings and local sonic "resonance" in three French Paleolithic painted caves. Archaeologist Chris Scarre summarized their findings in a brief article that drew much attention (Painting by Resonance, Nature 338 [1989]: 382). Scarre wrote "Reznikoff- Dauvois theory is consistent with the likely importance of music and singing in the rituals of our early ancestors." Reznikoff-Dauvois believed cave artists intentionally chose painting locations for their sonic resonance. They further conjectured it was the artists admiration for resonant sound that inspired their choice. This writer suggests the associations found were merely correlative, and not necessarily causal. (Crowing roosters do not cause the sun to rise.) How then can the correlation be explained? This writer hypothesizes that initially, "painterly" needs rather

than sonic preferences may have influenced choice of painting location (large expanses of non-porous rock). Since non-porous rock is highly sound reflective the best cave locations for long-lasting cave paintings are "resonant." Paintings on porous (non-resonant) rock would not persist. This is a testable hypothesis. Moreover, combining impressive art and sound could inspire singing and dance. Such sites might plausibly become ritual spaces.

2:40

5pAAc4. The acoustic of Cumaean Sibyl. Gino Iannace (Dept. of Architecture and Industrial Design, Università della Campania, Borgo San Lorenzo, Aversa 83016, Italy, gino.iannace@unina2.it) and Umberto Berardi (Architectural Sci. Dept., Ryerson Univ., Toronto, ON, Canada)

The Cumaean Sibyl cave is a place in the north of Naples, Italy. The Sibyl was a priestess presiding over the Apollonian oracle; she received the travelers to whom she predicted their future. The cave is length about 140 m, with trapezoidal section excavated in the rock of tuff; the height is about 4.5 m and large about 2.4 m. There is a little room, in the final part of the cave, where according to the legend, the Sibyl received the travelers. The acoustic measurements were done with an omnidirectional sound source in the in the little room and the microphones along the cave. The software for architectural acoustic were used to better understand the sound propagation in the cave. From acoustic measurements and numerical results it emerges that the voice emitted by an orator positioned in the room inside the Sybil cave, is understood in every tunnel point. This paper shows that the legend that this site was an oracle has true acoustical basis.

3:00–3:20 Break

3:20

5pAAc5. Acoustic measurements at the sacred sites in Finland. Riitta Rainio, Kai Lassfolk (Musicology, Univ. of Helsinki, Unioninkatu 38 C 213, FI-00014, Finland, riitta.rainio@helsinki.fi), Antti Lahelma (Archaeology, Univ. of Helsinki, Helsinki, Finland), and Tiina Äikäs (Archaeology, Univ. of Oulu, Oulu, Finland)

In Finland, near the canyon lakes of Julma-Ölkky, Somerjärvi and Taatsijärvi, steep rock cliffs produce distinctive acoustic spaces. On these cliffs, prehistoric rock paintings (5200—500 BC) as well as an ancient Sami offering site (cf. 1100—AD) can be found. Ethnographic sources describe that the Sami used to sing and listen to echoes while making offerings there. This paper presents the results of an archaeoacoustic project that seeks to explore the role of sound in the development and use of these archaeological sites. The applied methods include multichannel impulse response recording, angle of arrival estimation of early reflections, spectrum analysis, digital image processing, and 3D laser scanning. On the basis of the analyses, we have concluded that the cliffs that have been painted or held as sacred are efficient sound reflectors. They create discreet echoes and, accordingly, phantom sound sources. Especially at the Värikallio cliff, the sound appears to emanate directly from the painted figures. These results, together with previously unnoticed drumming figures in the Värikallio painting, provide a clue to the significance of the sound rituals at these sacred sites.

3:40

5pAAc6. A theoretical framework for archaeoacoustics and case studies. Steven J. Waller (Rock Art Acoust., 5415 Lake Murray Blvd. #8, La Mesa, CA 91942, wallersj@yahoo.com)

Application of black box theory is proposed as a theoretical framework for archaeoacoustic studies. First, input/output analysis can be applied to archaeological sites, in which the physical characteristics of a site serve as a black box to physically transform initial sonic inputs directly into various acoustic output phenomena that can be quantitatively measured, such as reflected repeats, resonance, etc. In turn, acoustic output from the first (archaeological) black box serves as input for a second black box: the human mind. Examples are presented to support the argument that the psychoacoustics of sound perception is/was a crucial aspect of archaeoacoustics. Cognitive response is heavily influenced by culture, expectations, and subjective values attributed to various sounds. Outputs of the second (cognitive) black box that can be analyzed include tangible archaeological manifestations such as rock art, megaliths, etc., as well as intangible responses such as orally recorded myths, traditions, rituals, and beliefs. Analysis of each of the input/output components from these two coupled black boxes can reveal important interrelationships between initial sonic inputs, sound transforming characteristics of archaeological sites, and the cognitive responses of ancient artists and architects to those transformed sounds. Case studies are presented to illustrate application of this theoretical approach.

4:00

5pAAc7. Archaeoacoustic guidelines. Preparation, execution, and documentation. David N. Thomas (Archaeology, Univ. of Highlands and Islands, 1/l,1 Sibbald St., Dundee, Tayside dd37ja, United Kingdom, sibbald1@blueyonder.co.uk)

Preparation A discussion of Archaeological theory and archaeoacoustics in a post processual era. The author considers the goals and objectives of archaeoacoustic research in the light of current trends; the nature of the human perception of sound is discussed in the light of the philosophical discussions of "Intentionality." The importance of a preliminary model displaying accurate reproduction of the resonant space, and use of light as indicator of sound wave propagation is emphasized (using examples from previously published Mousa broch and Minehowe papers). Execution Notes on practical considerations in the field. Equipment, sound makers, recording audio and image photos, tools, and markers. Mic and source recording locations. Wind noise and soundscapes. Notes on theoretical considerations in the field. Sound wave length and the word frequency, the myth of infrasound, and discrete echoes. Documentation Notes on types of documentation. Photo/video record, model record, audio record, and sound analysis. Conclusions, explanations, and possibilities. The implications of applying additional audio recording to illustrate convolution reverbs. Listing and sharing failed approaches. Sharing and Popularization of the discipline of archaeoacoustics.

4:20–5:00 Panel Discussion

Session 5pAB

Animal Bioacoustics: Ecosystem Acoustics II

Susan Parks, Cochair
Biology, Syracuse University, Biology, 107 College Place, RM 114, Syracuse, NY 13244

Jennifer L. Miksis-Olds, Cochair
Center for Coastal and Ocean Mapping, Univ. of New Hampshire, 24 Colovos Rd., Durham, NC 03824

Denise Risch, Cochair
Ecology, Scottish Association for Marine Science (SAMS), SAMS, Oban PA371QA, United Kingdom

Contributed Papers

1:20

5pAB1. Long-term monitoring of cetacean bioacoustics using cabled observatories in deep-sea off East Sicily. Francesco Caruso (IAMC, National Res. Council, Via del Mare 3, Torretta Granitola, Trapani 91021, Italy, fcaruso@unime.it), Virginia Sciacca (Univ. of Messina, Catania, Italy), Giuseppe Alonge (Observations and Analyses of Earth and Climate, ENEA, Palermo, Italy), Giorgio Bellia (Univ. of Catania, Catania, Italy), Giuseppa Buscaino (IAMC, National Res. Council, Capo Granitola (TP), Italy), Emilio De Domenico (Univ. of Messina, Messina, Italy), Rosario Grammauta (IAMC, National Res. Council, Capo Granitola (TP), Italy), Giuseppina Larosa (INFN, Catania, Italy), Salvatore Mazzola (IAMC, National Res. Council, Capo Granitola (TP), Italy), Gianni Pavan (CIBRA, Univ. of Pavia, Pavia, Italy), Elena Papale (IAMC, National Res. Council, Capo Granitola (TP), Italy), Carmelo Pellegrino (INFN, Bologna, Italy), Sara Pulvirenti (INFN, Catania, Italy), Francesco Simeone (INFN, Roma, Italy), Fabrizio Speziale (INFN, Catania, Italy), Salvatore Viola (INFN, Catanoa, Italy), and Giorgio Riccobene (INFN, Catania, Italy)

The EMSO Research Infrastructure operates multidisciplinary seafloor-cabled observatories in a deep-sea area offshore Eastern Sicily (2100 m of depth). In a data-lacking zone, Passive Acoustic Monitoring activities revealed new information on cetacean bioacoustics over multiple ecological scales. Expert operators investigated the presence of cetacean vocalizations within the large acoustic datasets acquired. Then, algorithms were developed to provide information on the behavior and ecology of the recorded species. In 2005-2006, the acoustic activity of toothed whales was investigated through the OvDE antenna (100 Hz to 48 kHz). The assessment of the size distribution of sperm whales was acoustically possible and the tracking of the animals showed the direction of movement and the diving profile. The biosonar activity of dolphins resulted mostly confined in the nighttime, linked to seasonal variation in daylight time and prey-field variability known for these deep-pelagic waters. Furthermore, in 2012-2013, we monitored the annual acoustic presence of fin whales thanks to the NEMO-SN1 station (1 Hz to 1 kHz). The results showed that the species was present throughout all seasons, with peaks in call detection rate during spring and summer months, and that the fin whale calls were mostly detected in low background noise conditions.

1:40

5pAB2. Polar coastal soundscapes: Tridimensional mapping of benthic biophony and ice geophony with a compact sensor array. Julie Lossent (Res. Inst. Chorus, 46, Ave. Felix Viallet, Grenoble cedex 1 38031, France, julie.lossent@chorusacoustics.com), Cedric Gervaise (Chair CHORUS, Saint Egreve, France), Laurent Chauvaud (Université de Bretagne Occidentale, Institut Universitaire Européen de la Mer, LIA BeBEST, Laboratoire des Sci. de l'Environnement Marin, Plouzané, France), Aurélie Jolivet (TBM Environnement, Auray, France), Delphine Mathias (Société d'Observation Multi-Modale de l'Environnement, Plouzané, France), and Jérôme Mars (Univ. Grenoble Alpes, CNRS, GIPSA-Lab, Grenoble, France)

Polar areas show fast changes linked to global warming. The reduction of the ice pack and the melting of the ice sheet modify the conditions of living of marine fauna. We propose the simultaneous monitoring of the ice and benthic fauna using passive acoustics. Thanks to a compact sensor array of 4 hydrophones (2m*2m*2m), we detected, localized and mapped in three dimensions ({azimuth, elevation} or {x, y, z}) the biophonic and geophonic contributions made up of short and wideband pulses. Tridimensional maps of benthic biophony and ice geophony of Antarctic and Arctic 7 days-long recording sessions (2015, 2016) are built and analyzed over a surface of the order of 1 km^2. Benthic invertebrates emit high energetic pulses with peak frequencies ranging from 2 to 55 kHz, most of them below 15 kHz. Geophony is structured into two parts. The ice sheet, located several kilometers or tens of kilometers away, creates a stable spatial distribution of low energetic pulses (representing the majority of pulses in the soundscape) modulated by the temporal variability. The movements of isolated icebergs or pack ice produce localized acoustic events identifiable by the high sound levels and the stable peak frequencies of the emitted pulses.

2:00

5pAB3. Acoustic habitat utilized by ice-living seals: Hearing and masking in natural noise environments. Jillian Sills, Colleen Reichmuth (Inst. of Marine Sci., Long Marine Lab., Univ. of California, Long Marine Lab., 115 McAllister Way, Santa Cruz, CA 95060, jmsills@ucsc.edu), and Alex Whiting (Kotzebue IRA, Native Village of Kotzebue, Kotzebue, AK)

Acoustic habitat is a fundamental but poorly understood resource for marine mammals, including seals. To evaluate the soundscapes experienced by seals in dynamic Arctic environments, two DSG-Ocean Acoustic Dataloggers were deployed in Kotzebue Sound, Alaska from September 2014 through September 2015, providing a full year of acoustic coverage for this region of the Chukchi Sea. The recorders were placed in an area of seasonal fast ice where spotted, ringed, and bearded seals are all found at various times of year. The data describe the acoustic conditions typically experienced by these ecologically and culturally important seal species, including variations in noise up to 48 kHz within and across scales of hours, days, months, and seasons. The noise profiles provide an ecological framework for laboratory studies of hearing with trained seals, allowing for improved understanding of their sensory biology in the context of their acoustic habitat. The integration of these noise measurements with hearing and auditory masking data enables a quantitative assessment of the effects of varying ambient noise conditions on the communication ranges of seals living in Arctic waters. [Work supported by the Northwest Arctic Borough Science Committee.]

2:20

5pAB4. Acoustic and biological trends on coral reefs off Maui, Hawaii. Maxwell B. Kaplan (Biology, Woods Hole Oceanographic Inst., 266 Woods Hole Rd., MS50, Woods Hole, MA 02543, mkaplan@whoi.edu), Marc Lammers (Hawaii Inst. of Marine Biology, Kaneohe, HI), T Aran Mooney (Biology, Woods Hole Oceanographic Inst., Woods Hole, MA), and Eden Zang (Oceanwide Sci. Inst., Makawao, HI)

Coral reef soundscapes comprise a range of biological sounds. To investigate how the sounds produced on a given reef relate to the species present, 7 Hawaiian reefs that varied in their species assemblages were equipped with acoustic recorders operating on a 10% duty cycle for 16 months, starting in September 2014. Benthic and fish visual surveys were conducted 4 times over the course of the study. Acoustic analyses were carried out in 2 frequency bands (50-1200 Hz and 1.8-20.5 kHz) that corresponded with the spectral features of the major sound-producing taxa on these reefs, fish and snapping shrimp, respectively. In the low-frequency band, the presence of humpback whales (December-May) was the major driver of sound level, whereas in the high-frequency band sound level closely tracked water temperature. On shorter timescales, the magnitude of the diel trend varied in strength among reefs, which may reflect differences in the species assemblages present. Regression trees indicated that, at low frequencies, the relationship between species assemblages and acoustic parameters varied by season; however, at high frequencies, a given reef was generally most like itself over time. Thus, long-term acoustic recordings can capture and distill the substantial acoustic variability present in coral reef ecosystems.

2:40

5pAB5. Temporal soundscape dynamics in a Magellanic penguin colony in Tierra del Fuego. Dante Francomano, Ben Gottesman, Taylor Broadhead, and Bryan C. Pijanowski (Dept. of Forestry and Natural Resources, Purdue Univ., Ctr. for Global Soundscapes, B066 Mann Hall, 203 South Martin Jischke Dr., West Lafayette, IN 47907, dfrancomano@gmail.com)

On Isla Martillo in Tierra del Fuego, we continuously recorded in a colony of Magellanic penguins (*Spheniscus magellanicus*) in the beginning of the 2016 molting season. Here we describe the daily soundscape dynamics within this colony using existing soundscape metrics, which were originally developed to facilitate acoustic-based ecological inferences from multi-source soundscapes. While these indices have exhibited great successes, little research has explored the utility of soundscape metrics for characterizing ecological patterns and processes when a single soniferous species dominates the soundscape. Bioacoustics offers tools for such applications, but soundscape metrics may be favorable in situations where sounds of chorusing animals temporally overlap or when sounds are non-stereotypical. Some of the diel behavior patterns of this species have been previously documented by studies focusing on foraging behavior, but these studies relied on human observation and dive trackers mounted on individual birds. Instead, we consider the potential utility of terrestrial acoustic recording to monitor populations and behavior of this near-threatened species. We interpret our acoustic data in the context of known Magellanic penguin behavior and the few non-penguin sounds in this habitat, and through this interpretation we evaluate how soundscape metrics can be used to assess nearly monospecific assemblages.

3:00

5pAB6. Noise affects black-tufted marmoset (*Callithrix penicillata*; GEOFFROY, 1812) *phee* call frequency. Sara Santos, Marina H. Duarte, Isabela F. Cardoso (Pontifícia Universidade Católica de Minas Gerais, Belo Horizonte, MG, Brazil), Renata S. Sousa-Lima (Physiol. and Behavior, UFRN, Lab. of BioAcoust., Centro de Biociencias, Campus Universitario, Caixa Postal 1511, Natal, Rio Grande do Norte 59078-970, Brazil, sousa-lima.renata@gmail.com), and Robert J. Young (Univ. of Salford Manchester, Salford, United Kingdom)

Anthropogenic noise is very different from natural sounds, and could cause organisms living in noisy areas to modify their vocal communication. We assessed the influence of noise on black tufted-ear marmoset acoustic communication. Spontaneously produced *phee* vocalizations were recorded in two areas: a noisy urban park, located in Belo Horizonte, Minas Gerais state, Brazil and a quiet natural forest, located at Cauaia in Matozinhos, in the same state. We also recorded bus brakes sounds (BBS) in the noisy park because we noticed that the sounds produced by these vehicles were similar to the *phee* vocalizations. Frequencies and duration of *phees* and the BBS were measured and used to compare: 1- *phees*from the urban and natural areas and 2- urban *phee* vocalizations and BBS. The duration of the *phee* calls was longer in the urban area. The low, high and dominant frequencies were significantly higher in the natural area. The low frequency extracted from BBS was similar to those of the *phee* calls of marmosets in the urban area. We suggest that the difference between the marmoset calls from urban and natural areas is influenced by noise and BBS compete with marmosets calls and may disturb the communication of these primates.

3:20–3:40 Break

3:40

5pAB7. Vocal behavior and ontogeny of Northern right whales in the southeast critical habitat. Edmund R. Gerstein (Charles E. Schmidt College of Sci., Florida Atlantic Univ., 777 Glades Rd., Boca Raton, FL 33486, gerstein2@aol.com), Vasilis Trygonis (Univ. of the Aegean, Lesvos, Greece), and James B. Moir (Marine Resources Council , Stuart, FL)

North Atlantic right whales are one of the most endangered of the great whales. A remnant population of ~500 inhabits the eastern seaboard of North America. A small fraction (2%) travels south to their critical calving habitat along the Florida and Georgia coast. By late November and through the winter, right whales give birth and nurse their calves in these shallow waters before departing in early spring to their northern habitats. In the southeast critical habitat mother-calf pairs remain generally isolated from other whales, presenting a unique platform to study vocal development and learning in large whales. Using small boats, GPS-instrumented, free-drifting autonomous acoustic buoys were deployed in close proximity to 44 photo-identified mother-calf pairs over 7 calving seasons. Surface video and synchronized underwater recordings documented their social and vocal behavior. With the exception of some low-energy gunshot sounds, mothers, and their calves, remained predominantly silent during the first 4 weeks. This might be due to calf maturation, and or a strategy to avoid harassment by other whales or potential predators. Over 100 calls have been analyzed from 15 different calves. Some of these calves were resampled at different stages at <1 week up to 12 weeks of age. Evidence of individual and age-related variance and changes in call structure, complexity, power, rates, as well as vocal mimicry are presented. [Funding: HBOI Florida PFW License Plate Fund, The Harry Richter Foundation and IBM, NOAA Permit #14233.]

4:00

5pAB8. Fish sound production in freshwater habitats of New England: Widespread occurrence of air movements sounds. Rodney A. Rountree (23 Joshua Ln., Waquoit, MA 02536, rrountree@fishecology.org), Francis Juanes (Biology, Univ. of Victoria, Victoria, BC, Canada), and Marta Bolgan (Univ. of Liège, Liége, Belgium)

We conducted a roving survey of five major river systems and adjacent, creek, lake, and pond habitats located within the northeastern United States. Fish sounds were recorded in 49% of 175 locations. Air movement sounds, including fast repetitive tick (FRT), occurred at 41% of the locations. Sluggish creeks had the highest occurrence of fish sounds (71%). Although highly variable, creeks and brooks had the lowest noise levels and rivers the highest. Fish sounds were more frequent in low noise habitats than high noise habitats, but the effect of masking on detection is not clear. Within main-stem river habitats, fish sound diversity tended to increase along a gradient from high elevation to the sea. Follow-up studies validated air movement sounds produced by alewife, white sucker, and brook, brown and rainbow trout through direct observation or in observations where only single species were present. Sounds produced by all five species are of the "air movement" type which is poorly understood but occurs widely in freshwater habitats. Although air movement sounds are likely incidental to physiological processes, they appear to be uniquely identifiable to species and, hence, hold promise for passive acoustic studies of freshwater soundscapes and fish behavior.

4:20

5pAB9. Characterizing soundscapes and larval fish settlement in tropical seagrass and mangrove habitats. Ian T. Jones, Justin Suca, Joel Llopiz, and T. Aran Mooney (Biology, Woods Hole Oceanographic Inst., Woods Hole Oceanographic Inst., 266 Woods Hole Rd, Marine Res. Facility 225 (MS #50), Woods Hole, MA 02543, ijones@whoi.edu)

Recent evidence suggests soundscapes of coral reefs may provide acoustic cues that larval reef fish utilize during settlement. Seagrass and mangrove habitats are further important refuges for larvae and juveniles of many fishes; however, compared to reefs, less is known about the characteristics of tropical seagrass and mangrove soundscapes and their potential as settlement cues. We deployed light traps to assess fish larvae settlement and passive acoustic recorders to study the "ecoacoustics" at mangrove, seagrass, and coral reef sites around two bays in St. John, U.S. Virgin Islands. Light traps were deployed nightly around the third quarter and new moon, and 24 h periods of acoustic recordings were taken during the same time. Fish larvae were counted and identified to the lowest possible taxonomic level. Focusing on biotic soundscape components, diel trends in metrics such as sound pressure level, power spectral density, and snap counts of snapping shrimp were assessed. Although what role mangrove and seagrass soundscapes may play in fish settlement remains unclear, these soundscape and larval settlement data provide foundations for deeper investigations of the relationship between acoustic and larval ecology in these essential nursery habitats.

4:40

5pAB10. Toward an unsupervised and automated analysis framework for large acoustic datasets to assess animal biodiversity and ecosystem health. Yu Shiu, Ashakur Rahaman, Christopher W. Clark, and Holger Klinck (BioAcoust. Res. Program, Cornell Univ., 159 Sapsucker Woods Rd., Ithaca, NY 14850, atoultaro@gmail.com)

Passive acoustic monitoring is a promising and non-invasive method to assess the biodiversity and potentially health of terrestrial and marine ecosystems. Over the last decade, various methods have been proposed to extract information on the animal biodiversity primarily based on acoustics indices. Several recent studies have shown that ecological relevance and effectiveness of these indices remain uncertain. We propose a new, multistep method to estimate animal biodiversity from acoustic datasets by applying an unsupervised detection and classification technique. Our semi-automated framework extracts every acoustic event with a pre-defined signal-to-noise ratio. In a second step, the detected events are grouped into classes based on the similarity of acoustic features. The number of resultant classes are linked to animal biodiversity in an area by applying a transfer function, which is established using manually/expert reviewed class labels. Our framework provides diel and seasonal changes in the overall number of sound classes as well as number of acoustic events in each class. We will demonstrate its performance by application to three datasets collected in the Chukchi Sea, Alaska, Sapsucker Woods Sanctuary, Ithaca, NY, and Abel Tasman National Park, New Zealand.

5:00

5pAB11. Using automated acoustic monitoring to detect elevational migration in the avian community of the Gondwana Rainforests of Australia. Elliot Leach (Environ. Futures Res. Inst., Griffith Univ., 170 Kessels Rd., Brisbane, QLD 4111, Australia, elliot.leach@griffithuni.edu.au), Chris Burwell (Biodiversity Program, Queensland Museum, Brisbane, QLD, Australia), Darryl Jones, and Roger Kitching (Environ. Futures Res. Inst., Griffith Univ., Brisbane, QLD, Australia)

Climate change presents the most significant threat to Australia's rainforest avifauna. In order to determine the future impacts of climate change and make informed conservation decisions, baseline information on species distributions, elevational preferences, and seasonal movements is necessary. Traditionally, generating data such as these over large spatio-temporal scales has been difficult and costly. However, the recent development of cheap, reliable bioacoustic recorders has facilitated such data collection. By using automated acoustic recorders in subtropical rainforest along two elevational gradients in northern New South Wales, Australia, we were able to continuously monitor the avian community for a 14-month period. The data generated during this project allowed us to detect seasonal elevation migration amongst resident species, the arrival and departure times of migratory species and the breeding behavior of cryptic species. This research also represented the first comprehensive avian biodiversity survey conducted in the region. Here, we present our results from the automated acoustic monitoring, and discuss the implications for future research and monitoring of the avian community in the Gondwanan Rainforests of Australia.

5:20

5pAB12. Large scale passive acoustic monitoring using compact arrays of synchronized hydrophones. Ildar R. Urazghildiiev (JASCO Appl. Sci. (Alaska) Inc., 19 Muriel St., Ithaca, NY 14850, ildar.urazghildiiev@jasco.com) and David E. Hannay (JASCO Appl. Sci., Victoria, BC, Canada)

Modern and future ecosystem-level data processing techniques need to solve the problem of detecting, classifying, localizing, tracking, and estimating density (DCLTDE) concurrently from all sounds an acoustic recorder detects. To solve this problem, we propose a technique based on three major components: the compact array of synchronized hydrophones; the automatic DCLTDE technique; and the software to visualize and to rapidly analyze the results of long-term data processing. The compact array provides an additional information about the azimuth and elevation angles of the detected sounds, and the data processing technique uses this information to solve the required DCLTDE problems automatically. Processed acoustic recordings collected over one year in the Strait of Georgia, BC, Canada, are presented. These results demonstrated that the proposed technique dramatically increases the efficiency-to-cost ratio by decreasing the time needed for a person to analyze a huge amount of data and by increasing the amount and accuracy of the information extracted from acoustic recordings.

Session 5pAO

Acoustical Oceanography: Tools and Methods for Ocean Mapping II

Scott Loranger, Cochair
Earth Science, University of New Hampshire, 24 Colovos Road, Durham, NH 03824

Philippe Blondel, Cochair
Physics, University of Bath, University of Bath, Claverton Down, Bath BA2 7AY, United Kingdom

Invited Papers

1:20

5pAO1. Acoustic detection of macroalgae in a dynamic Arctic environment. Isfjorden (West Spitsbergen) case study. Aleksandra Kruss (Coastal Systems and Human Impacts, CNR ISMAR, Tesa 104, Castello 2737/F, Venice, Veneto 30122, Italy, aleksandra.kruss@ve.ismar.cnr.it), Jozef Wiktor, Agnieszka Tatarek, and Jozef Wiktor, Jr. (Marine Ecology, Inst. of Oceanology PAS, Sopot, pomorskie, Poland)

Acoustic imaging of seabed morphology and benthic habitats is a fast-developing tool for investigating large areas of underwater environment. Even though single- and multi-beam echosounders have been widely used for this purpose for many years, there is still much to discover, especially in terms of processing water column echoes to detect macroalgae and other scatterers (e.g., fishes, or suspended sediments) that can provide us with important information about the underwater environment and its evolution. In difficult Arctic conditions, acoustic monitoring plays an important role in the investigation of bottom morphology and in imaging habitats. In July 2016, we carried out a multidisciplinary expedition to investigate macroalgae spatial distribution in Isfjorden and to measure significant environmental features (currents, salinity, turbidity) influencing their occurrence. An area of 4.3 km^2 was mapped using single- and multibeam sonars along with underwater video recordings, CTD and ADCP measurements. We obtained a unique data set showing variability of acoustic properties among different macroalgae species, supported by very well correlated ground-truth data and environmental measurements. Modern processing techniques were used to analyze water column data signals for kelp detection. This study presents efficient tools for monitoring benthic communities and their environmental context, focusing on macroalgae acoustic characteristics.

1:40

5pAO2. Synthetic aperture sonar interferometry for detailed seabed mapping: Performance considerations. Roy E. Hansen, Torstein O. Sæbø, Stig A. Synnes, and Ole E. Lorentzen (Norwegian Defence Res. Establishment (FFI), P O Box 25, Kjeller NO-2027, Norway, Roy-Edgar.Hansen@ffi.no)

Synthetic Aperture Sonar (SAS) interferometry is a technique for detailed mapping of the seabed, with the potential of very high resolution and wide swaths simultaneously. There are several specific challenges to overcome for the technique to reach its full potential. These differ from other mapping sensor technologies, e.g., multibeam echosounders (MBES), and interferometric sidescan sonars (ISSS). In this talk, we describe the principle of SAS interferometry with emphasis on the estimation part, strongly inspired by the similar principle in synthetic aperture radar (SAR). We describe the limiting factors in using SAS interferometry for seabed depth estimation. These are related to the host platform, the measurement geometry, the sonar array design, and the signal processing. We construct an error budget where we categorize the different components that affect the overall performance. We also describe the choices and tradeoffs available in the signal processing for a given set of measurements. We show example images and depth estimates from the Kongsberg HISAS interferometric SAS collected by a HUGIN autonomous underwater vehicle.

2:00

5pAO3. Internal wave effects on seafloor imagery and bathymetry estimates. Anthony P. Lyons (Ctr. for Coastal and Ocean Mapping, Univ. of New Hampshire , Durham, NH 03824, anthony.lyons@ccom.unh.edu), Roy E. Hansen (Norwegian Defence Res. Establishment (FFI), Kjeller, Norway), James Prater (Naval Surface Warfare Ctr. Panama City Div., Panama City, FL), Warren A. Connors (NATO STO Ctr. for Maritime Res. and Experimentation, La Spezia, Italy), Glen Rice (Hydrographic Systems and Technol. Programs, NOAA, Durham, NH), and Yan Pailhas (Ocean Systems Lab., Herriot Watt Univ., Edinburgh, United Kingdom)

Large linear structures (tens of meters by several meters) have been observed recently in seafloor imagery and bathymetry collected with both synthetic aperture sonar (SAS) and multibeam echosounder (MBES) systems. It has been suggested [Hansen, *et al*., IEEE J. Oceanic Eng., 40, 621-631 (2015)] that this phenomenon is not due to the true morphology of the seafloor, but is caused by water column features related to breaking internal waves. Changes observed in acoustic intensity and bathymetry estimates are caused by a focusing of the acoustic field which results in structures that appear to be true seabed topography. In terms of seafloor mapping, these topography-mimicking features will impact the interpretation of imagery, may complicate the production of mosaics, and have the potential to cause bathymetric uncertainties exceeding International Hydrographic Organization standards. In this talk we will show that these water-column caused features may not be uncommon using examples of data collected with several different SAS and MBES systems in a variety of experimental locations.

2:20

5pAO4. Circular synthetic aperture sonar image resolution theory. Yan Pailhas and Yvan Petillot (Heriot-Watt Univ., Riccardo Campus, School of EPS, Edinburgh EH14 4AS, United Kingdom, Y.Pailhas@hw.ac.uk)

The introduction of SAS (Synthetic Aperture Sonar) systems has been a game changer for underwater surveys. The gain in resolution, compared to traditional sidescan systems, created a paradigm shift as the information contained in a SAS image switches from shadows to highlights. SAS systems traditionally perform lawnmower type surveys, but the need for multiple views in MCM (Mine-Counter Measure) tasks, for example, opened the interesting problem of target re-acquisition patterns. In particular, circular patterns maximize the aperture, thus the overall image resolution of such system. The capability of CSAS (Circular SAS) has been demonstrated on the field, but the derivation of CSAS processing has not been fully developed. The non-uniform sampling of the circular pattern in particular introduces aberrations within the field of view and a non uniform PSF (Point Spread Function). In this talk, we propose a new spatial sampling scheme which makes the CSAS PSF perfectly uniform. The theoretical closed form solution of the PSF is then derived both in time and Fourier domain. The PSF derivation naturally leads to redefine the image resolution as an energy leakage problem. Thanks to the new sampling scheme and the uniform PSF, we also propose a deconvolution method based on atom waves which increases the CSAS resolution.

2:40–3:00 Break

Contributed Papers

3:00

5pAO5. Seafloor mapping with a cylindrical array. Glen Rice (Ctr. for Coastal and Ocean Mapping, Univ. of New Hampshire, 24 Colovos Rd., Durham, NH 03824, grice@ccom.unh.edu), Ole Bernt Gammelseter (Simrad, Kongsberg, Horten, Norway), and Thomas C. Weber (Ctr. for Coastal and Ocean Mapping, Univ. of New Hampshire, Durham, NH)

Seafloor mapping is conducted with different types of arrays. Single beam mapping systems are often constructed with piston arrays. Linear arrays are used for both side scan sonar and for multibeam echo sounders. These conventional approaches to seafloor mapping are typically constrained to a single observation of any one point on the seafloor for any one pass of a moving vessel. A less conventional cylindrical array offers the opportunity to observe the majority of the seafloor from different perspectives but with the same angle with a single pass. This has the potential to improve the resulting depth estimates and seafloor backscatter products. In 2016, a Simrad Omnisonar was used to demonstrate seafloor mapping with a cylindrical array. While this array is designed for observing fish schools, a small area was successfully mapped and the results compared with a conventional bathymetric mapping system. Observations on the benefits and challenges of this approach to seafloor mapping will be discussed.

3:20

5pAO6. Improving seep detection by swath sonars with adaptive beamforming. Tor Inge B. Lønmo (Informatics, Univ. of Oslo, P.O.Box 111, Horten 3191, Norway, toribi@ifi.uio.no) and Thomas C. Weber (Ctr. for Coastal and Ocean Mapping, Univ. of New Hampshire, Durham, NH)

Detection of gas seeps is currently of interest for a wide variety of applications. The oil and gas industry can use it to monitor their installations, e.g. oil wells and pipelines. It may also contribute to the understanding of geological and biological activity in the seabed. In a climate perspective, it is also important for estimating the amount of methane that seeps into the atmosphere and monitoring CO_2 stored in geological structures. Seeps are commonly detected by bathymetric swath sonars. Bubbles are strong acoustical targets and may form clear flares in the water column display, depending on the size and density of the bubbles. Detection is often easy before the first bottom return arrive and is gradually masked by the seafloor reverberation at longer ranges. We propose to use the sidelobe suppressing properties of adaptive beamforming to suppress seafloor reverberation and extend seep detection range. To investigate this in practice we placed an artificial seep at approximately 45 m depth and mapped it with a swath sonar. We ran lines over the seep at between 0 and 80 m horizontal range. Our processing chain allows us to process each ping with both standard and adaptive beamforming, providing easily comparable results.

3:40

5pAO7. Spatio-temporal variability of bottom reverberation striation pattern in shallow water. Andrey Lunkov (A.M. Prokhorov General Phys. Inst. RAS, Vavilov St. 38, Moscow 119991, Russian Federation, landr2004@mail.ru)

Interference pattern of bottom reverberation can be a source of additional information about underwater environment. Over the last decade, the striations in bottom backscattered broadband signals in the time-frequency domain were observed in several short-term experiments (Goldhahn *et al*., 2008, Li *et al*., 2010). In our research, numerical simulations are carried out to analyze the long-term variability of bottom reverberation striation pattern in the presence of internal waves (IW), both background and solitary-like. Coherent normal mode reverberation model is applied. Backscattered chirp signals are selected from different directions by a bottom mounted horizontal array. Simulated striation patterns reveal frequency shifts that are dependent on bearing and IW type. This phenomenon can be used to monitor the spatio-temporal variability of IW. To increase the sensitivity of this approach, the time-reversal sound focusing on the seafloor is proposed. [Work supported by RFBR, 16-32-60194.]

4:00

5pAO8. Paradigm shift of underwater echo sounding technology III—Evaluation of paradigm shift echo-sounder. Ikuo Matsuo (Dept. of Information Sci., Tohoku Gakuin Univ., Tenjinzawa 2-1-1, Izumi-ku, Sendai 9813193, Japan, matsuo@mail.tohoku-gakuin.ac.jp) and Toyoki Sasakura (FUSION Inc., Tokyo, Japan)

Underwater echo sounding technology was developed 65 years ago and has been applied to various systems such as fishfinder, bathymetry, sonar, side scanning sonar, multi beam echo sounding system and synthetic aperture sonar. We have suggested a new concept that may change the history of underwater echo sounding technology, should be called as "Paradigm Shift." In the conventional system, the transmission intervals are always longer than the time of the round trip distance to the target divided by the underwater sound velocity. By adapting our new "Paradigm Shift Echo-sounder" into the system, the transmission intervals can be decided except for depending on a target and it will be possible to conduct a bathymetry survey that transmits 100 times per second. The system utilized the 7th order Gold code sequence signals. Transmitting signal is the phase modulated and four cycles of carrier signals as one bit of the Gold code sequence. The actual sea trial experiment was done by the proto type of "Paradigm Shift Echo-sounder." The experimental result was obtained by the transmission of 100 times per second at the depth of 10 to 80 m sea bottom. We also confirmed that this system could reconstruct a high-resolution echogram of the artificial reef.

4:20

5pAO9. Coastal ocean current tomography using a spatio-temporal Kalman filter. Tongchen Wang, Ying Zhang, Tsih C. Yang, Wen Xu, and Huifang Chen (College of Information Sci. and Electronic Eng., Zhejiang Univ., Hangzhou, Zhejiang 310027, China, talentwtc@163.com)

The method of ocean acoustic tomography (OAT) can be used to invert/map the ocean current in a coastal area based on measurements of two-way travel time differences between the nodes deployed on the perimeter of the surveying area. Previous work has attempted to relate the different measurements in time using the Kalman filter. Now, if the ocean dynamics or model is known, one can also determine the current field given an initial distribution or the travel-time difference data on the boundary, and can even forecast the current changes. Based on the ocean dynamics, the current field is shown to be spatially and temporally correlated. We derive their relation and use that as the state model for the Kalman filter; the coefficients are estimated from data using an auto-regressive analysis. Armed with this model, it is shown based on simulated data that the current field can be tracked as a function of time using the Kalman filter (with an arbitrary initial condition) with a higher accuracy than that estimated by OAT. The reason of the improvement, the use of spatial-temporal state model (versus using only the temporal evolution), is studied. The method has also been applied to real data.

THURSDAY AFTERNOON, 29 JUNE 2017 ROOM 312, 1:15 P.M. TO 5:40 P.M.

Session 5pBAa

Biomedical Acoustics and ASA Committee on Standards: Standardization of Ultrasound Medical Devices

Volker Wilkens, Cochair
Ultrasonics Working Group, Physikalisch-Technische Bundesanstalt, Bundesallee 100, Braunschweig 38116, Germany

Subha Maruvada, Cochair
U.S. Food and Drug Administration, 10903 New Hampshire Ave., Bldg. WO 62-2222, Silver Spring, MD 20993

Chair's Introduction—1:15

Invited Papers

1:20

5pBAa1. The International Electrotechnical Commission (IEC) and ultrasonics. Peter J. Lanctot (Int. ElectroTech. Commission, 446 Main St., Ste. 16, Worcester, MA 01608, pjl@iec.ch)

Peter Lanctot from the International Electrotechnical Commission (IEC) will provide an overview of the international standardization activities related to IEC Technical Committee 87, Ultrasonics. Since 1955, the IEC has published international standards for a wide range of applications across virtually all business sectors including medicine, electronics, consumer products, food, manufacturing industries, and defence for ultrasonic technologies. As TC 87 is in charge of Ultrasonic Standardization activities within the IEC, Peter will provide a high level overview on their standards work oriented towards ultrasonic aspects of medical equipment and the safety of non-medical applications of ultrasonic fields. He will also touch upon the important platform that IEC TC 87 and other IEC technical committees offer to companies, industries, and governments for meeting, discussing, and developing the International Standards they require. Based in Geneva, Switzerland, the International Electrotechnical Commission is the world's leading organization that prepares and publishes International Standards for all electrical, electronic and related technologies.

1:40

5pBAa2. Overview of International Electrotechnical Commission Technical Committee 87 ultrasound organization and standards. Subha Maruvada (U.S. Food and Drug Administration, 10903 New Hampshire Ave., Bldg. WO 62-2222, Silver Spring, MD 20993, subha.maruvada@fda.hhs.gov)

The International Electrotechnical Commission (IEC) is the world's leading organization that prepares and publishes International Standards for all electrical, electronic and related technologies. It comprises Technical Committees (TCs) and Subcommittees (SCs) that oversee standards development in all areas of technology. Particular to ultrasound medical devices, TC 87, Ultrasonics, is responsible for preparing standards related to measurement methods, field characterization, and safety concerns, but excluding safety and essential performance standards for equipment and systems. Responsibility for the latter falls under TC 62, Electrical Equipment in Medical Practice. SC 62B in TC 62 is in charge of all diagnostic imaging equipment and related devices, including ultrasound, and SC 62D deals with all electromedical equipment used in medical practice other than diagnostic imaging, such as ultrasound therapeutic and surgical devices. This presentation will discuss the safety standards, measurement standards, maintenance/performance documents and structure and distribution of work in TC 87 and SC62B and SC62D as an introduction to the more specific talks following.

2:00

5pBAa3. Safety and performance testing according to international standards and regulatory environment for medical ultrasound devices. Royth P. von Hahn (Medical and Health Services, TUV SUD America, 10040 Mesa Rim Rd., San Diego, CA 92126, rvonhahn@tuvam.com) and Mathias Kuhn (Medical and Health Services, TUV SUD ProDC Service, Munich, Germany)

Besides being scientifically interesting, acoustics resp. ultrasonics has a broad and diverse range of applications in industry and the medical field. Acceptance of technology and its application depend on safety and performance of technology resp. devices utilizing it. Standardization is essential to ensure safety and performance and it also helps to harmonize market access of devices all around the world. To keep up with the "state of the art" it is necessary to continuously update existing standards and develop new ones. Especially in medical ultrasound a unique combination of expertise from different scientific areas is needed to develop useful standards: In addition to acoustics, physics and HF engineering, aspects of bioeffects and measurement technology need to be represented in standard development. The presentation will give insights how such combination of expertise contributed to the current set of medical ultrasound standards and what technical and scientific challenges need to be solved for developing safety standards for the latest innovations. On actual examples, it will show the relevance of scientific research for standardization and device testing.

2:20

5pBAa4. Basic ultrasonic field measurement: Overview of standardized methods and expected developments. Volker Wilkens (Ultrason. Working Group, Physikalisch-Technische Bundesanstalt, Bundesallee 100, Braunschweig 38116, Germany, volker.wilkens@ptb.de)

Basic ultrasonic field measurement methods to support the acoustic output characterization of medical ultrasound equipment are standardized in IEC 61161 for ultrasonic power by means of radiation force balance measurements and in IEC 62127-1,2,3 for ultrasonic pressure using hydrophones. Descriptions and requirements for ultrasonic power determination seem to be broadly elaborated even, for instance, for extended frequency ranges of high frequency diagnostic systems or for very high output powers of high intensity therapeutic ultrasound devices. In contrast, hydrophone measurements often are accompanied by several challenges still not fully addressed by current standards. Important deficiencies to overcome in future editions are the frequency range of calibrations currently limited to 40 MHz and being insufficient for broadband waveform deconvolution applications, and the descriptions of deconvolution and of the corresponding measurement uncertainty determination themselves. Improvements of these technical items according to recently developed calibration and data evaluation procedures are expected to result in better quality and reliability of acoustic output parameter determination as required, for instance, for diagnostic ultrasound machines within the output declaration and output display according to IEC 60601-2-37 and IEC 62359. In addition, new projects on standards in the high intensity therapeutic area may also take advantage of such improvements.

2:40

5pBAa5. International Electrotechnical Commission (IEC) and Food and Drug Administration (FDA) efforts to develop standards for ultrasound physiotherapy. Bruce Herman (U.S. FDA,.10903 New Hampshire Ave. Bldg. 66, Silver Spring, MD 20993, bruce.herman@comcast.net)

Ultrasound physiotherapy devices produce acoustic energy, typically in the low MHz range, to generate deep heat for relief of pain and spasms, and have been used since the 1940's. The IEC and the FDA have an interrelated history of developing standards for these devices. The IEC produced the first such standard "Testing and calibration of ultrasound therapeutic equipment"(1963). The FDA, with the IEC document as guide, developed its own "Ultrasonic Therapy and Surgery Products Performance Standard"(1978, 2012). Then the IEC, with FDA involvement and using many of the FDA's concepts, produced "Medical electrical equipment—Part 2-5: Particular requirements for the basic safety and essential performance of ultrasonic physiotherapy equipment (1984, 2000, 2009) and "Ultrasonics—Physiotherapy systems—Field Specifications and methods of measurement in the frequency range 0.5 MHz to 5 MHz" (1996, 2007, 2013). FDA personnel are also currently leading an IEC effort to write a standard dealing with new, lower frequency physiotherapy. The FDA is also committed to harmonizing FDA regulations with IEC standards whenever possible. This presentation recounts the history of these efforts and examines some similarities and differences among the documents.

3:00

5pBAa6. Standards for the characterization of high intensity nonlinear ultrasound fields and power generated by therapeutic systems. Thomas L. Szabo (Biomedical Dept., Boston Univ., 44 Cummington Mall, Boston, MA 02215, tlszabo@bu.edu)

The emerging field of high intensity therapeutic ultrasound (HITU) presents multiple challenges. Early attempts to characterize high pressure fields fried both hydrophones and force balances originally designed for diagnostic ultrasound. Working Group 6 of International Electrotechnical Commission (IEC) Technical Committee 87 initially sought out methods employing current validated measurement technology. Technical Specification 62256 focused on two field characterization approaches for water. The first was using conventional calibrated hydrophones in the linear range and a scaling up of field pressure values by the applied voltage ratio. The other method involved a finely sampled 2D scan near the transducer face as input data to linear projection algorithms to recreate the entire pressure field. This computed field could then be scaled up the previously described voltage ratio. An additional benefit of the second approach was that the same scan data could be used to predict pressure and thermal levels in simulated tissue media representing clinical scenarios. Technical standard 62255 described HITU power measurements with scaling and a novel buoyancy method. Current efforts are directed at the direct measurement of high pressure fields with new technologies and the data-based nonlinear simulation of fields and bioeffect end points in water and tissues.

3:20

5pBAa7. Standards for pressure pulses used in lithotripsy, pain therapy, and other medical applications. Friedrich Ueberle (Life Sci. / Biomedical Eng., Hamburg Univ. of Appl. Sci. (HAW), Ulmenliet 20, Hamburg 21033, Germany, friedrich.ueberle@haw-hamburg.de)

Shockwave lithotripsy was first used in 1980 for the non-invasive, safe treatment of stones in the urinary tract. The first commercial lithotripters were introduced in 1983, using an underwater spark discharge as sound source, which was focused by an ellipsoidal mirror. Single steep (Shockwave) pressure pulses of few microseconds duration with 20…>100MPa amplitude are released at a rate up to 2 per second. Each treatment requires ca. 1000 to 3000 pulses. Competitors developed lithotripters with spherical piezoelectric and focused electromagnetic sources. Up today, these three source types are applied in commercial lithotripters. With the occurrence of different types of sources from different manufacturers, it became important to standardize the description of pulses and acoustic wave fields, for clinical approval and safety, the understanding of pressure pulse interaction with biological tissue and stones, and for quality control. A lithotripter safety standard was created 1997 (IEC601-2-36). In 1998, the international standard IEC61846 was released, which describes measurement methods and parameters for focused pressure pulse sources. Both standards are regularly reviewed and enhanced in maintenance projects. A new project (IEC63045) defines parameters for non-focusing and weakly focusing pressure pulse sources, which are widely used in pain therapy and other tissue applications since 1998.

3:40

5pBAa8. Fast accurate optical measurement of medical ultrasonic field in combination with numerical simulation of nonlinear propagation. Shin-ichiro Umemura and Shin Yoshizawa (Graduate School of Biomedical Eng., Tohoku Univ., Aoba 6-6-05, Aramaki, Aoba-ku, Sendai 980-8579, Japan, sumemura@ecei.tohoku.ac.jp)

Fast and accurate measurement of ultrasonic field is necessary to ensure the safety and efficacy of therapeutic as well as diagnostic applications of medical ultrasound. The most common method for the purpose is hydrophone scanning. However, it requires a long scanning time and potentially disturbs the field, which is limiting the efficiency of developing such applications. This study proposes an optical phase contrast method in combination of a CT algorithm. Ultrasonic pressure field modulates the phase of the light passing through the field. A phase plate was employed to shift the phase of the non-diffracted component of the light typically by 90 degrees. The phase modulation of the diffracted component was converted to amplitude modulation through interference with the phase-shifted non-diffracted component and then measured by camera. From the measured projected 2D data, the 3D pressure field was reconstructed by a CT algorithm. An upstream field, in which the optical phase does not wrap and the effect of nonlinear propagation can be ignored, was thereby quantified. Nonlinear ultrasonic propagation was simulated based on a pseudo spectral method using the upstream pressure field as the input. Both pressure waveform and absolute pressure from the proposed method agreed well with those directly from hydrophone scanning.

Contributed Papers

4:00

5pBAa9. Acoustic holography combined with radiation force measurements to characterize broadband ultrasound transducers and calibrate hydrophone sensitivity. Sergey Tsysar, Maxim A. Kryzhanovsky, Vera Khokhlova (Phys. dept, Lomonosov Moscow State Univ., GSP-1, 1-2 leninskie Gory, Moscow 119991, Russian Federation, sergey@acs366.phys.msu.ru), Wayne Kreider (CIMU, Appl. Phys. Lab., Univ. of Washington, Seattle, WA), and Oleg Sapozhnikov (Phys. dept, Lomonosov Moscow State Univ., Moscow, Russian Federation)

Accurate characterization of broadband ultrasound (US) transducers operating either in pulsed or CW modes is necessary for their use in biomedical applications. The acoustic holography method allows reconstruction of the pattern of vibrations at the transducer surface for use as a model boundary condition to simulate ultrasound fields in water or tissue. A calibrated hydrophone is often not available to perform such measurements. Here, it is proposed to combine transient acoustic holography with radiation force balance (RFB) measurements at multiple frequencies to quantitatively determine 3D distributions of the acoustic field at different source frequencies, the corresponding acoustic power of the source, and the hydrophone sensitivity within the frequency bandwidth of the transducer. First, a transient acoustic hologram is measured using a short-pulse excitation of the source by raster scanning the hydrophone along a surface in front of the source and recording the waveform at a large number of points (typically 10-40 thousand). Then, RFB measurements are conducted for different frequencies within the pulse bandwidth to determine axial component of the acoustic radiation force. These data are related to the corresponding values calculated from the holograms at each frequency component to determine the corresponding hydrophone sensitivities. [Work supported by RSF-14-12-00974.]

4:20

5pBAa10. A quick and reliable acoustic calibration method for a clinical magnetic resonance guided high-intensity focused ultrasound system. Satya V.V. N. Kothapalli (Biomedical Eng., Washington Univ. in Saint Louis, 4511 Forest Park Ave., Saint Louis, MO 63108, vkothapalli@wustl.edu), Ari Partanen (Clinical Sci. MR Therapy, Philips, Andover, MA), Michael Altman (Dept. of Radiation Oncology, Washington Univ. in St. Louis, Saint Louis, MO), Zhaorui Wang (Biomedical Eng., Washington Univ. in Saint Louis, Saint Louis, MO), H. Michael Gach, William Straube, Dennis Hallahan (Dept. of Radiation Oncology, Washington Univ. in St. Louis, Saint Louis, MO), and Hong Chen (Biomedical Eng., Washington Univ. in Saint Louis, Saint Louis, MO)

With the expanding use and applications of MR-HIFU in both thermal-based and pressure-based therapies, there is an urgent need to develop acoustic field characterization and quality assurance (QA) tools for MR-HIFU systems.We developed a method for quick and reliable acoustic field assessment inside the magnet bore of a clinical MRI system. A fiber-optic hydrophone with a 2-m long fiber was fixed inside a water tank that was placed on the HIFU table above the acoustic window. The long fiber allowed the MRI-incompatible hydrophone control unit to be located outside the MRI suite. MR images of the fiber were used to position the HIFU focus approximately at the tip of the fiber. The HIFU focus was electronically steered within a $5\times5\times5\text{mm}^3$ volume in synchronization with hydrophone measurements. The HIFU focus location was then identified based on the 3D field scans. Peak positive and negative pressures were measured at the focus at various nominal acoustic powers. Furthermore, focus dimensions and spatial peak pulse average intensities were assessed. The results were compared to and were consistent with standard hydrophone measurements outside the MRI suite. This method provides a useful tool for field characterization and QA of MR-HIFU systems.

4:40

5pBAa11. Use of acoustic holography to quantify and correct geometrical errors in ultrasound field characterization. Wayne Kreider (CIMU, Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, wkreider@uw.edu), Vera Khokhlova, Sergey Tsysar, and Oleg Sapozhnikov (Phys. Faculty, Moscow State Univ., Moscow, Russian Federation)

The development of medical ultrasound devices requires characterization of the ultrasound fields they radiate. Recently, acoustic holography methods have gained increased acceptance for field characterization. Because such an approach allows reconstruction of the full 3D field, the beam axis, its orientation relative to the positioner, and any field quantities of interest can be readily determined. Inherently, the accuracy of such projection methods can be sensitive to uncertainties in the positions at which field measurements are recorded. Although commonly used industrial positioning systems have linear axes with repeatability specifications much less than an ultrasound wavelength, the assembled orthogonality of three linear axes is not guaranteed. Here we analyze a typical raster scanning holography experiment by considering two distinct coordinate systems: ideal rectilinear coordinates aligned with the transducer (1.2 MHz, aperture 124 mm, F-number 1) and non-orthogonal coordinates aligned with the positioner (Velmex Unislides). By locating a distinct field feature in both projection calculations and independent hydrophone measurements, a misalignment of positioner axes of about 0.50 was identified. The impact of positioner non-orthogonality on field characterization metrics is discussed along with the potential use of this approach for *a priori* positioner calibration. [Work supported by NIH EB007643, NIH EB016118, and RSF-14-15-00665.]

5:00

5pBAa12. Two reduced-order approaches for characterizing the acoustic output of high-power medical ultrasound transducers. Vera Khokhlova, Petr Yuldashev (Phys. Faculty, Moscow State Univ., Leninskie Gory, Moscow 119991, Russian Federation, vera@acs366.phys.msu.ru), Adam D. Maxwell (Dept. of Urology, Univ. of Washington, Seattle, WA), Pavel Rosnitskiy, Ilya Mezdrokhin, Oleg Sapozhnikov (Phys. Faculty, Moscow State Univ., Moscow, Russian Federation), Michael Bailey, and Wayne Kreider (Ctr. for Industrial and Medical Ultrasound, Univ. of Washington, Seattle, WA)

An approach that combines numerical modeling with measurements is gaining acceptance for field characterization in medical ultrasound. The general method is suitable for accurate simulation of the fields radiated by therapeutic transducers in both water and tissue. Here, three characterization methods are compared. Simulations based on the 3D Westervelt equation with a boundary condition determined from acoustic holography measurements are used as the most accurate benchmark method. Two simplified methods are based on an axially symmetric nonlinear parabolic formulation, either the KZK equation or its wide-angle extension. Various approaches for setting a boundary condition to the parabolic models are presented and discussed. Simulation results obtained with the proposed methods are compared for a typical therapeutic array and a strongly focused single-element transducer, with validation measurements recorded by a fiber optic hydrophone at the focus at increasing acoustic outputs. It is shown that the wide-angle parabolic model is more accurate than the KZK model in governing diffraction effects in the nearfields of the focused beams. However, both methods give accurate results in the focal zone, even at very high outputs when shocks are present. [Work supported by and RSF 14-12-00974, P01 DK043881, NIH EB7643, and NSBRI through NASA 9-58.]

5:20

5pBAa13. Use of wide-element one-dimensional receiving arrays to measure two-dimensional lateral pressure distribution of ultrasound beams. Oleg Sapozhnikov (Phys. Faculty, Moscow State Univ., and CIMU, Appl. Phys. Lab., Univ. of Washington, Leninskie Gory, Moscow 119991, Russian Federation, oleg@acs366.phys.msu.ru), Wayne Kreider (CIMU, Appl. Phys. Lab., Univ. of Washington, Seattle, WA), Adam D. Maxwell (Dept. of Urology, Univ. of Washington School of Medicine, Seattle, WA), and Vera Khokhlova (Phys. Faculty, Moscow State Univ., and CIMU, Appl. Phys. Lab., Univ. of Washington, Moscow, Russian Federation)

Measurement of acoustic fields is an important aspect of ultrasound research and development. The corresponding data is usually collected by a raster scan using a single point receiver moved by a computer-controlled positioning system. A faster approach could be based on the use of a 1D linear multi-element array with several tens or hundreds of small elements, which is moved in a direction perpendicular to the beam axis. The drawback of such an approach is difficulty in making small, closely-positioned receiving elements that provide sufficient signal-to-noise ratio. The current paper presents an alternative design for a 1D linear array: instead of using small elements, here it is suggested to use narrow but long elements, with a width on the order of a half wavelength or less and a length larger than the extent of the ultrasound beam being studied. The drawback of the proposed long-element arrays is absence of resolution in the direction along the element. This problem is solved by incrementally rotating the array either around the axis parallel to the array surface and perpendicular to the elements or around the axis perpendicular to the array surface. [Work supported by RSF 14-15-00665, NIH R01 EB007643, and NIH R21 EB016118.]

Session 5pBAb

Biomedical Acoustics and Signal Processing in Acoustics: Diagnostic and Therapeutic Applications of Ultrasound Contrast Agents II

Tyrone M. Porter, Cochair
Boston University, 110 Cummington Mall, Boston, MA 02215

Klazina Kooiman, Cochair
Thoraxcenter, Dept. of Biomedical Engineering, Erasmus MC, P.O. Box 2040, Room Ee2302, Rotterdam 3000 CA, Netherlands

Invited Paper

1:20

5pBAb1. Ultrasound molecular imaging. Jonathan Lindner (Oregon Health & Sci. Univ., OHSU Cardiology UHN61, 3181 SW Sam Jackson Park Rd., Portland, OR 97239, lindnerj@ohsu.edu)

Non-invasive *in vivo* molecular imaging technologies are uniquely poised to be able to temporally evaluate vascular adaptations to disease and the impact of new therapies. These technologies have been used to study vascular changes in various forms of cardiovascular and inflammatory disease and cancer. The approaches for studying the macro- or microvasculature with non-invasive imaging go beyond anatomic or perfusion characterization and instead yield information on vascular phenotype by either analyzing novel signal features or using targeted contrast agents that reveal the molecular underpinnings for disease. This talk will review new developments in ultrasound molecular imaging techniques that are likely to have a substantial impact on the understanding of pathophysiology, the development of new therapies, or diagnosis of disease in patients. Topics of focus will include recent advances in molecular imaging for the evaluation of: (1) atherosclerotic disease, (2) tissue inflammation or ischemic injury; and (3) thrombus or prothrombotic environment, (4) angiogenesis and stem cells. The discussion will also include a discussion of how these different approaches may play a role to solve current clinical deficiencies.

Contributed Papers

1:40

5pBAb2. Toward detection of early apoptosis using labeled microbubbles. Tom Matula, Masaoki Kawasumi, Daiki Rokunohe, Brian MacConaghy, and Andrew Brayman (Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, matula@uw.edu)

Chemotherapy is useful for treating metastases, but on average only 50% of tumors respond to the initial choice of drug(s). Many patients thus suffer from unnecessary side effects while their tumors continue to grow, until the proper drugs are found. Because most drugs kill cancer cells by inducing apoptosis, detection of apoptosis can be used to assess drug efficacy prior to treatment. Current technologies require large cell numbers and several time points for quantification. We propose a simple drug screening tool to monitor early apoptosis using annexin-V labeled microbubbles. We investigated the parameter space for such a tool with a transparent flow chamber such that the acoustic radiation force would be relatively parallel with the focal plane of the microscope's objective. An uncalibrated PZT was activated with a small voltage from a function generator (1-10 V, 2 MHz), generating sufficient pressure to displace microbubble-labeled cells. Initial studies focused on detection of labeled leukemia cells (not apoptosis). In one study, labeled and unlabeled cells were mixed together. In another study, labeled cells were mixed with red blood cells. Displacements were measured with or without flow. The results suggest that a microbubble-based drug screening tool can be developed for real-time monitoring of apoptosis.

2:00

5pBAb3. Role of mannitol on the echogenicity of echogenic liposomes (ELIPs). Krishna N. Kumar (Dept. of Mech. Eng., The George Washington Univ., 800 22nd St. NW, Washington, DC 20052, Washington, DC 20006, krishnagwu@gwu.edu), Sanku Mallik (Dept. of Pharmaceutical Sci., North Dakota State Univ., Fargo, ND), and Kausik Sarkar (Dept. of Mech. Eng., The George Washington Univ., Washington, DC)

Although echogenic liposomes (ELIP), specially prepared lipid-bilayer coated vesicles, have proved quite effective as scatterers, the exact mechanism of their echogenicity is not well understood. However, freeze-drying in presence of mannitol has proved to be a critical component of the elaborate preparation protocol of these ELIPs. Here, we investigate the role played by mannitol in ensuring echogenicity. We investigated mannitol along with other similar sugars, such as sucrose, trehalose, and xylitol. Mannitol when freeze-dried assumes a crystalline state, while other sugars adopt glassy states. Aqueous solutions of each sugar were prepared and freeze-dried. The freeze-dried samples were re-dissolved in water and the scattered response from the solution was measured. While the solution of mannitol was found echogenic, indicating production of bubbles, others were not. If the sample was freeze-thawed before dissolution, it was not echogenic. The crystalline state of the excipient, mannitol, was necessary for echogenicity. Furthermore, other excipients such as glycine and meso-erythritol, which attain crystalline state upon freeze-drying, also were found to give rise to echogenicity in solution. The production of bubbles by crystalline mannitol upon dissolution indicates a possible mechanism of echogenicity of ELIPs.

Invited Paper

2:20

5pBAb4. Ultrasound triggered mRNA delivery to dendritic cells—Towards an *in vivo* cancer vaccination strategy. Heleen Dewitte (Pharmaceutics, UGent, Ottergemse Steenweg 460, Gent 9000,Belgium), Emmelie Stock, Katrien Vanderperren (Veterinary Sci., UGent, Gent, Belgium), Stefaan De Smedt (Pharmaceutics, UGent, Gent, Belgium), Karine Breckpot (Health Sci., VUB, Brussels, Belgium), and Ine Lentacker (Pharmaceutics, UGent, Gent 9000, Belgium, Ine.Lentacker@UGent.be)

Increasing knowledge on the crucial role of dendritic cells (DCs) in the initiation of immunity has launched a new field in cancer immunotherapy: DC vaccination. By loading patient's DCs with tumor antigens and injecting them as a vaccine, antitumor immune responses can be induced. This project aims to use theranostic mRNA-loaded MBs for ultrasound-guided, ultrasound-triggered antigen-loading of DCs within the lymph nodes *in vivo*. mRNA-loaded MBs were prepared by attaching mRNA-lipid complexes to lipid MBs via avidin-biotin linkages. MBs loaded with mRNA encoding a tumor antigen (ovalbumin, OVA) were used to sonoporate murine DCs *in vitro*. These mRNA-sonoporated DCs were then used as therapeutic vaccines in E.G7-OVA-bearing mice. *In vitro* mRNA-sonoporation of murine DCs revealed transfection efficiencies up to 27%. The potential of this technique was further assessed *in vivo* by vaccinating E.G7-OVA-bearing mice with sonoporated DCs. When mRNA-sonoporated DCs were used, tumor growth was significantly reduced and even regressed in 30% of the animals. Moreover, rechallenge with tumor cells did not lead to tumor growth, indicating long-lasting immunological protection. A CEUS study in dogs showed the MBs rapidly drained to the lymph nodes after sc injection. Moreover, this revealed detailed information on the lymphatic anatomy.

Contributed Papers

2:40

5pBAb5. Effect of molecular weight on sonoporation-mediated uptake in human cardiac cells. Danyal F. Bhutto, Emily M. Murphy (BioEng., Univ. of Louisville, 2301 S. Third St., Paul C.Lutz Hall, Rm. 419, Louisville, KY 40292-0001), John Zhao, Joseph B. Moore (Medicine, Univ. of Louisville, Louisville, KY), Roberto Bolli (Medicine, Univ. of Louisville, La Grange, KY), and Jonathan A. Kopechek (BioEng., Univ. of Louisville, Louisville, KY, jonathan.kopechek@louisville.edu)

Sonoporation of cells induced by ultrasound-driven microbubble cavitation has been utilized for intracellular delivery of molecular therapeutics. The molecular weight of therapeutic agents can vary significantly, with DNA plasmids often larger than 5 MDa, siRNAs and miRNAs ~10 kDa, and other drugs often less than 1 kDa. Some studies have suggested that sonoporation-mediated uptake may decrease at higher molecular weights due to slower diffusion rates, but experiments with equal molar concentrations have not been reported. Therefore, the objective of this study was to explore the effect of molecular weight on sonoporation-mediated uptake of fluorescein (0.33 kDa) or fluorescent dextrans (10 kDa, 2 MDa) using equal molar concentrations (100 nM). Sonoporation was induced in cultured human cardiac mesenchymal cells using lipid microbubbles and 2.5 MHz B-mode ultrasound (P4-1 transducer, Verasonics Vantage ultrasound system) and uptake by viable cells was measured with flow cytometry. No significant differences in sonoporation-mediated uptake were measured between molecules of different sizes at equal molar concentrations (ANOVA $p = 0.92$). However, significant differences in uptake were observed at different molar concentrations and acoustic pressures (ANOVA $p = 0.02$). In summary, these results suggest that the effect of molecular weight on sonoporation-mediated uptake is minimal compared to drug concentration or acoustic pressure.

3:00

5pBAb6. Enhanced delivery of a density-modified therapeutic using ultrasound: Comparing the influence of micro- and nano-scale cavitation nuclei. Harriet Lea-Banks, Christophoros Mannaris, Megan Grundy, Eleanor P. Stride, and Constantin Coussios (Eng. Sci., Univ. of Oxford, Inst. of Biomedical Eng., Old Rd. Campus Res. Bldg., Oxford OX3 7DQ, United Kingdom, harriet.lea-banks@eng.ox.ac.uk)

New strategies are required to enhance the penetration and tumor-wide distribution of cancer therapeutics, including viruses, antibodies, and oligonucleotides. We have shown previously that increasing the density of a nanoparticle can enhance its ultrasound-mediated transport, particularly when exposed to microstreaming. The current study investigates how the physical characteristics and dynamics of different cavitation nucleation agents affect the cavitation-mediated transport of therapeutics not directly bound to the gas nuclei. SonoVue (Bracco, Milan, Italy) microbubble contrast agent (with 3 μm mean initial bubble diameter) and polymeric gas-entrapping nanoparticles (with 260 nm mean initial bubble diameter) were first compared in terms of inertial cavitation threshold and cavitation persistence. Under the same ultrasound exposure conditions (centre frequencies of either 0.5 MHz or 1.6 MHz, and peak negative pressures from 0.2 MPa to 3.5 MPa) their respective cavitation emissions were found to differ in magnitude, frequency composition and duration. The resultant penetration depths of co-injected gold-coated nanoparticles in a tissue-mimicking phantom were also found to vary, with the polymeric gas-entrapping nanoparticles giving rise to greater penetration distances. This study concludes that both the characteristics of the therapeutic and the cavitation nuclei can significantly influence ultrasound-mediated drug delivery.

3:20–3:40 Break

3:40

5pBAb7. Optimization of liposome delivery using low intensity ultrasound and microbubbles. Jia-Ling Ruan (Oncology, Univ. of Oxford, Old Rd. Campus Res. Bldg., Roosevelt Dr., Oxford OX37DQ, United Kingdom, jia-ling.ruan@oncology.ox.ac.uk), Richard J. Browning (Inst. of Biomedical Eng., Univ. of Oxford, Oxford, United Kingdom), Yildiz Yesna, Borivoj Vojnovic (Oncology, Univ. of Oxford, Oxford, United Kingdom), Eleanor P. Stride (Inst. of Biomedical Eng., Univ. of Oxford, Oxford, United Kingdom), and Anne Kiltie (Oncology, Univ. of Oxford, Oxford, United Kingdom)

Liposomes are versatile drug delivery microvehicles to carry small molecule therapeutics and have a steadily growing global market share. Liposomal formulations can stabilize therapeutic compounds, promote cellular uptake, and improve biodistribution. The controlled delivery of liposomes can be further enhanced by ultrasound and microbubbles. Here we study the effect of ultrasound-mediated liposome delivery with microbubbles. By incorporating the fluorescent lipophilic dye, DiO, we investigated the effect of different liposome loading techniques and microbubble conjugation on liposome delivery *in vitro*. We found that active-loading liposomes demonstrated a better ultrasound-mediated delivery efficacy when conjugated with microbubbles than when used as a free drug. We also investigated the acoustic parameters: pressure (70 to 2000 kPa), duty cycle (10 to 50%), and PRF (2 to 1000 Hz); at a fixed 1 MHz center frequency for optimal liposome delivery with low cell detachment. Our results indicate that the liposome-carried microbubbles could be a promising new drug delivery vehicle to incorporate into the current chemoradiation therapy regime against cancer.

4:00

5pBAb8. Integration of focused ultrasound with nanomedicine for brain drug delivery. Dezhuang Ye (Mech. Eng. and Material Sci., Washington Univ. in St. Louis, 1155 Claytonia Terrace #2N, 2N, Saint Louissaint louis, MO 63117, dezhuang.ye@wustl.edu), Cinzia Federico, Vaishali Kapoor, Dinesh Thotala, Dennis Hallahan, Abdel Kareem Azab (Radiation Oncology - Cancer Biology, Washington Univ. in St. Louis, Saint Louis, MO), and Hong Chen (Biomedical Eng., Washington Univ. in St. Louis, Saint Louis, MO)

Nanomedicine is the next-generation technology for cancer therapy. Liposomes are one of such nanomedicine that have well-established clinical applications. However, their large sizes make it challenging to deliver them to the brain due to the existence of the blood-brain barrier (BBB). Previous studies have demonstrated the potential of integrating focused ultrasound (FUS)-induced BBB opening with liposomes for brain drug delivery. However, no study has characterized the spatial distributions of the liposomes and released drugs in the brain parenchyma at a microscopic level after the BBB was disrupted by the FUS. In this study, we manufactured liposomes with a fluorescent dye embedded in the lipid layer and chemotherapeutic drug doxorubicin encapsulated in the core. These liposomes were intravenously injected into the mouse followed by FUS sonication of a targeted brain location in the presence of microbubbles. The mouse brains were coronally sectioned and imaged using fluorescence microscopy. Preliminary results confirmed the successful delivery of doxorubicin to the brain. However, the liposome distribution as indicated by the dye embedded in the lipid shell was not spatially correlated with the distribution of the doxorubicin. Further studies are needed to better understand the mechanism of FUS-enabled liposome delivery.

4:20

5pBAb9. Comparative lytic efficacy of rt-PA and intermittent ultrasound in porcine versus human clots. Shenwen Huang, Himanshu Shekhar, and Christy Holland (Univ. of Cincinnati, 231 Albert Sabin Way, Mail Location: 0586, Cincinnati, OH 45229, huangsw@mail.uc.edu)

Porcine thrombi are employed routinely in preclinical models of ischemic stroke. In this study, we examined the differential lytic susceptibility of porcine and human whole blood clots to recombinant tissue-plasminogen activator (rt-PA) and the adjuvant use of an echo contrast agent, Definity®, and intermittent 120 kHz ultrasound to enhance thrombolysis. An *in vitro* system equipped with time-lapse microscopy was used to evaluate rt-PA lysis of porcine and human clots in same species or cross species plasma. The rt-PA lytic efficacy observed for porcine clots in porcine plasma was 22 times lower than for human clots in human plasma. Porcine clots did not exhibit increased lysis with adjuvant Definity® and ultrasound exposure. However, the rt-PA lytic susceptibility of the porcine clots in human plasma was similar to that of human clots in human plasma. Human clots perfused with porcine plasma did not respond to rt-PA, but adjuvant use of Definity® and ultrasound enhanced lysis. These results reveal considerable differences in lytic susceptibility of porcine clots and human clots to rt-PA, and highlight the limitations of current porcine models used to develop clinical thrombolytic therapies.

4:40

5pBAb10. Sonobactericide: An ultrasound-mediated adjunct treatment for bacterial infective endocarditis—*In vitro* proof-of-principle. Kirby R. Lattwein (Biomedical Eng., Erasmus Medical Ctr., P.O. Box 2040, Rm. Ee2302, Rotterdam 3000CA, Netherlands, k.lattwein@erasmusmc.nl), Himanshu Shekhar (Internal Medicine, Div. of Cardiovascular Health and Disease, Univ. of Cincinnati, Cincinnati, OH), Willem J. van Wamel (Medical Microbiology and Infectious Diseases, Erasmus Medical Ctr., Rotterdam, Netherlands), Andrew B. Herr (Immunobiology, Cincinnati Children's Hospital Medical Ctr., Cincinnati, OH), Christy K. Holland (Internal Medicine, Div. of Cardiovascular Health and Disease, Univ. of Cincinnati, Cincinnati, OH), and Klazina Kooiman (Biomedical Eng., Erasmus Medical Ctr., Rotterdam, Netherlands)

Infective endocarditis (IE) is a bacterial infection of the surfaces within the heart, associated with high morbidity and mortality rates. The predominant bacteria causing IE are Staphylococci, which can bind to existing thrombi on heart valves and generate vegetations (biofilms). In this *in vitro* study, we test a novel strategy called sonobactericide to treat IE using ultrasound and ultrasound contrast agents, in combination with an antibiotic and a thrombolytic. We simulated IE vegetations using human whole-blood clots infected with *Staphylococcus aureus*. Histology and live-cell imaging revealed a biofilm layer of fibrin-embedded living Staphylococci. Infected clots were treated under flow for 30 min and degradation was assessed by time-lapse microscopy imaging. Clots were exposed to human plasma flow either alone or with different combinations of therapeutics: oxacillin (172 μg/mL), recombinant tissue plasminogen activator (rt-PA; 3.15 μg/mL), intermittent (50 s on, 30 s off) continuous-wave ultrasound (120 kHz, 0.44 MPa peak-to-peak pressure), and Definity (2 μl/ml). Infected clots exposed to the combination of oxacillin, rt-PA, ultrasound, and Definity achieved 99.3 $\pm$ 1.7% clot width loss, which was greater than the other treatment arms. These results demonstrate that sonobactericide may have potential as an adjunct therapy for IE.

5:00

5pBAb11. The use of a novel microfluidics system for *in vitro* ultrasound-mediated drug delivery. Ines Beekers, Tom van Rooij (Biomedical Eng., Erasmus MC, Office Ee2302, P.O. Box 2040, Rotterdam, Zuid Holland 3000 CA, Netherlands, d.beekers@erasmusmc.nl), Martin D. Verweij (Acoust. Wavefield Imaging, Delft Univ. of Technol., Delft, Netherlands), Michel Versluis (Phys. of Fluids, Univ. of Twente, Enschede, Netherlands), Nico de Jong (Biomedical Eng., Erasmus MC, Rotterdam, Netherlands), Sebastiaan J. Trietsch (Mimetas B.V., Leiden, Netherlands), and Klazina Kooiman (Biomedical Eng., Erasmus MC, Rotterdam, Netherlands)

Ultrasound insonification of gas filled microbubbles can be used to locally enhance vascular drug delivery [1]. Understanding of the underlying mechanisms requires an *in vitro* endothelial cell model that includes 3D cell culture and flow. We propose to use the elaborate microchannel system in the OrganoPlate™ [2]. Since it has never been used with ultrasound before, we assessed the feasibility of acoustic transmission. The oscillation of microbubbles in the OrganoPlate™ was recorded during ultrasound insonification (50 kPa, 1-4 MHz, 8 cycles) using the Brandaris 128 ultra-high-speed camera at 17 Mfps. We compared these oscillations to measurements performed in an acoustically transparent OptiCell [3]. From 1 to 1.6 MHz, larger oscillation amplitudes were observed in the OrganoPlate™, resulting from pressure amplification in the microchannels. Finite element modeling of wave propagation using PZFlex supported our findings. Successful sonoporation of 3D cultured endothelial cells was achieved in preliminary experiments using CD31 targeted microbubbles. The obtained microbubble oscillations and successful sonoporation demonstrate the potential of the OrganoPlate™ for an ideal *in vitro* model to investigate ultrasound-mediated drug delivery. [1] Kooiman, *et al*., Adv. Drug Deliv. Rev., 2014. [2] Trietsch, *et al*., Lab Chip, 2013. [3] van Rooij, *et al*., Ultrasound Med. Biol., 2015.

5:20

5pBAb12. Acoustic microbubble trapping for enhanced targeted drug delivery. Luzhen Nie, Sevan Harput, James R. McLaughlan, David Cowell, Thomas Carpenter, and Steven Freear (School of Electron. Eng., Univ. of Leeds, Leeds LS2 9JT, United Kingdom, s.freear@leeds.ac.uk)

Systemic drug administration can result in low-efficiency therapeutic delivery (less than 1% to solid tumors) and potential side-effects by subjecting all tissues to toxic agents. Recently, targeted drug delivery and localized payload release has been a key goal of biomedical research. By combining ultrasound contrast agents (UCAs) with a therapeutic payload, promises potential for an enhanced therapeutic outcome. We have proposed a ultrasonic trap technique that is able to accumulate large populations of UCAs for enhanced localized release of a therapeutic agent. Sequences for UCA manipulation, fast imaging, and UCA fragmentation are interleaved with a same linear array medical ultrasound transducer. Once UCA clusters are halted in the trap within flow channels through custom-designed beam profiles, destruction pulses are transmitted to deliver therapeutic payload and simultaneous fast imaging depicts the flow dynamics. Verification of the simultaneous operation of trapping, destruction and imaging sequences is performed with a tissue-mimicking flow phantom and microscopy, respectively. Optical imaging clarifies UCAs can still be retained within the low pressure trap during the fast switch between different excitation schemes.

Session 5pBAc

Biomedical Acoustics: Therapeutic Ultrasound and Bioeffects

Robert J. McGough, Chair
Michigan State University, 2120 Engineering Building, East Lansing, MI 48864

Contributed Papers

1:20

5pBAc1. A universal data quality assessment method for human brain magnetic resonance elastography using an elastic spherical phantom. Spencer Brinker, Nathan McDannold, and Srinivasan Mukundan (Dept. of Radiology, Brigham and Women's Hospital, Harvard Med. School, 221 Longwood Ave., EBRC 521, Boston, MA 02115, SBRINKER@BWH.HARVARD.EDU)

Human brain Magnetic Resonance Elastography (MRE) is on the verge of being utilized as a clinical diagnostic tool. Several methods are being investigated at multiple clinical facilities; however each have different ways of introducing vibrations into the brain and processing image data. These differences result in dissimilar complex shear modulus values, which make it difficult to reproduce and report brain tissue mechanical properties across the research community and in longitudinal clinical studies. Human brain MRE Data Quality Assessment (DQA) reporting methods are needed. This study investigates a universal method to regulate the precision and accuracy of MRE independent of the vibration system and post processing technique. A thin-walled 15.2 cm diameter acrylic sphere was filled with silicone gel. The phantom is designed to mimic human head, neck, and torso mechanics during human brain MRE scanning and is compatible with most brain vibration methods. MRE-derived storage shear modulus yields approximately 1.5 kPa (60 Hz vibration) using a custom air pillow system (Resoundant, Inc.). The material will be cross validated with rheometry for further validation. This study provides insight on design characteristics for a standard DQA phantom for regulating the quality of human brain MRE for research reporting and in the clinic.

1:40

5pBAc2. Evaluation of shear wave generation methods for intraoperative magnetic resonance elastography in a clinical transcranial focused ultrasound system. Spencer Brinker and Nathan McDannold (Dept. of Radiology, Brigham and Women's Hospital, Harvard Med. School, 221 Longwood Ave., EBRC 521, Boston, MA 02115, SBRINKER@BWH.HARVARD.EDU)

The current standard for FDA approved real-time monitoring of transcranial Focused Ultrasound (FUS) ablation is MR-thermometry. Ultrasound-elastography and Magnetic Resonance Elastography (MRE) have shown success in monitoring FUS ablation in various parts of the body. Introducing shear wave vibrations into the brain during transcranial-FUS becomes technologically challenging given the stringent constraints of the operating space in water filled transcranial phased array devices and since patients are fixed in stereotactic headgear during treatment. This study investigates the feasibility of using intraoperative-MRE during transcranial-FUS ablation in the ExAblate 4000 (InSightec, Ltd.) platform. Acoustic radiation force induced vibration using the ExAblate 4000 transducer and common MRE surface actuation methods are evaluated for shear wave generation. Elastic phantom experiments were used to gauge the acoustic power levels and time incurred during vibrations from MRE scanning while in the FUS transducer. These values were compared to FUS and MR parameters used in clinical transcranial-FUS treatments. MRE derived 3D displacement fields within the phantom materials are evaluated for all vibration methods. The cooling water in the ultrasound traducer influenced the MR signal using current MRE pulse sequences. Future work will involve integrating MRE into sequences less sensitive to cooling water media and minimizing intraoperative-MRE scan time.

2:00

5pBAc3. Transoesophageal HIFU for cardiac ablation: *In-vivo* experiments in non-human primates. Paul Greillier (LabTau - U1032, INSERM, 30, rue des Tables Claudiennes, LYON, Rhône 69001, France, paul.greillier@inserm.fr), Bénédicte Ankou, Francis Bessière (Hôpital Louis-Pradel, Lyon, France), ali Zorgani (LabTau - U1032, INSERM, Bron, France), Wojciech Kwiecinski (Institut Langevin - Ondes et Images - ESPCI ParisTech, CNRS UMR 7587, Paris, France), Julie Magat (IHU-LIRYC - CHU Bordeaux, Pessac, France), Sandrine Melot-Dusseau, Romain Lacoste (Station de primatologie - CNRS- UPS846, Rousset, France), Bruno Quesson (IHU-LIRYC - CHU Bordeaux, Pessac, France), Mathieu Pernot (Institut Langevin - Ondes et Images - ESPCI ParisTech, CNRS UMR 7587, Paris, France), Stefan Catheline (LabTau - U1032, INSERM, Lyon, France), Philippe Chevalier (Hôpital Louis-Pradel, Lyon, France), and Cyril Lafon (Univ. of Virginia, Lyon, France)

Transesophageal HIFU was proposed as an alternative to the current atrial fibrillation treatments. The present work described a feasibility study of transesophageal thermal ablation in the heart of non-human primates. An endoscope integrating a 5MHz 64-element commercial transesophageal echocardiography probe and a 8-element HIFU transducer was built. The transducer was cooled at 5°C and ultrasonic beam could be steered over a 15 to 55 mm range. The probe was tested *in-vivo* on three 30kg-baboons. Left atrium and ventricles were exposed to repeated continuous sonications (4-15 times during 16s) at a focal intensity of 3000 W/cm^2. B-mode, shear-wave and passive elastographies were performed before and after treatments in an attempt to monitor thermal lesions. T1 mapping and contrast MR imaging were realized the day after treatment. Clinical states of the subjects during and after the treatment were positive. One lesion in the left ventricle could be evidenced by elastography and confirmed by MRI. Experiments demonstrated the feasibility of a transoesophageal HIFU procedure to produce thermal lesions in beating hearts. Further developments will aim at improving the robustness of the technique. Work supported by the FUS Foundation. [Probe designed in collaboration with Vermon SA (Tours, France).]

2:20

5pBAc4. Ultrasound exposure during collagen polymerization produces pro-migratory fiber structures. Emma Grygotis (Pharmacology and Physiol., Univ. of Rochester, School of Medicine and Dentistry, 601 Elmwood Ave., Box 711, Rochester, NY 14642, emma_grygotis@urmc.rochester.edu), Diane Dalecki (Biomedical Eng., Univ. of Rochester, Rochester, NY), and Denise C. Hocking (Pharmacology and Physiol., Univ. of Rochester, Rochester, NY)

Non-invasive techniques to control protein structure and function are needed for tissue engineering applications. This study tested the hypothesis that non-thermal effects of ultrasound can produce changes in collagen microstructure that support directional cell migration. Type I rat tail collagen was polymerized for 15 minutes in the presence of an 8-MHz ultrasound standing wave field over a range of 0 to 30 W/cm^2 spatial peak, temporal average intensity. Temperature-matched sham gels were manufactured in a heated water bath without exposure to ultrasound. To test for effects of acoustically-modified collagen on cellular behavior, fibronectin-null mouse embryonic fibroblasts or mouse skin explants were seeded on gel surfaces and cultured up to 7 days prior to imaging with phase or second harmonic generation (SHG) microscopy. Acellular acoustically-modified collagen gels were characterized by regions of radial fiber alignment, increased pore size, and denser fibers, with greater heterogeneity at higher intensities. Ultrasound-exposed gels supported rapid directional cell migration with accumulation of cells in the regions of highest SHG signal. Neither fiber alignment nor cellular migration was observed in temperature-matched sham gels. Results demonstrate that ultrasound exposure during collagen polymerization can result in functionally altered collagen microstructure, in part through a non-thermal mechanism.

2:40

5pBAc5. Fluorescence activated cell sorting via focused traveling surface acoustic waves (FTSAWs). Zhichao Ma, David J. Collins, and Ye Ai (Singapore Univ. of Technol. and Design, 8 Somapah Rd., Singapore, Singapore 487372, Singapore, mazhichao_jlu@hotmail.com)

Fluorescence Activated Cell Sorting (FACS) is an essential technique widely used in biomedical analyses. Microfluidics, benefiting at its low power and sample consumption, has enabled miniaturization of the existing bulk FACS equipment into cost-effective and portable devices. However, these devices still have limited efficiency and biocompatibility whose actuation is based on dielectrophoresis, optical tweezers, and gas valve. Acoustophoresis, the migration of particles in acoustic field, has recently emerged as a promising method to manipulate suspended particles in microscale, thanks to the rapid response and good biocompatibility. However, limited by the wide aperture of the acoustic field, cell encapsulation in ~100s μm of droplets is one of the necessary procedures in the demonstrated acoustophoresis based FACS system to ensure sort single cell at each actuation, increasing the complexity of the system. In present work, a FACS system actuated by focused traveling surface acoustic waves has been developed. Since the aperture of acoustic field is reduced to the same order of cells by exploiting the focused acoustic waves, the cell can be sorted one at a time without encapsulation. Separation of circulating tumor cells (CTCs) from white blood cells (WBCs) was demonstrated to show good performance of the system.

3:00

5pBAc6. Towards using acoustic waves as a therapeutic tool for osteogenic differentiation. Lucas R. Shearer, Anna Elefante (Mech. Eng., Univ. of Hartford, 200 Bloomfield Ave., West Hartford, CT 06117, lshearer@hartford.edu), Ahmad Arabiyat, Jacqueline Maynard, Andrea Kwaczala (Biomedical Eng., Univ. of Hartford, West Hartford, CT), and Eoin A. King (Mech. Eng., Univ. of Hartford, West Hartford, CT)

Mesenchymal stem cells that reside in the bone marrow are unique in that they have the potential to differentiate into several cell types such as bone, fat, or cartilage. The mesenchymal stem cells are responsible for maintaining bone by differentiating into osteoblasts. Previous research has demonstrated the ability to influence this differentiation process through mechanical or chemical stimulation. Delivered at various frequencies and amplitudes, mechanical vibration is being considered as a non-drug treatment of osteoporosis. This paper reports on a study conducted in the University of Hartford to examine the possibility of using acoustic excitation, rather than mechanical vibration, to control cell differentiation. *In vitro* cell cultures (containing MC3T3-E1 pre-osteoblasts) were excited with white noise in a reverberation chamber for 30 minutes each day, over a four-day testing period. Preliminary results indicate an increase in cell proliferation due to acoustic excitation, but perhaps more significantly, prove that cells can be successfully transferred to and from a controlled acoustic environment. A more rigorous test regimen, including the development of a desktop stimulation chamber, has been scheduled to further quantify the response of these cells to similar acoustic stimulus.

3:20–3:40 Break

3:40

5pBAc7. Dynamics of pulsed-ultrasound driven gas-liquid interfaces. Brandon Patterson and Eric Johnsen (Univ. of Michigan, 1231 Beal Ave., Rm 2016, Ann Arbor, MI 48109, awesome@umich.edu)

Pulsed ultrasound has been shown to be capable of inducing lung hemorrhage in mammals, though the physical mechanism remains unknown. We model ultrasound-alveolar interactions as a compressible fluid system and perform numerical experiments to investigate the relevant dynamics. Previously we demonstrated that the interaction between trapezoidal acoustic waves and perturbed gas-liquid interfaces, such as those in the lung, can generate sufficient baroclinic vorticity to appreciably deform the interface under certain conditions. In this work we study the dynamics of perturbed liquid-gas interfaces driven by a single ultrasound pulse in and out of the clinically relevant range with frequencies 1.5-3 MHz and amplitudes 1-15 MPa. We calculate theoretical stresses and strains and compare to established failure criteria. We observe that for ultrasound pulses that significantly deform the interface during the wave-interface interaction, there is residual vorticity left at the interface which can deform the interface long after the passage of the wave. This effect increases with wave amplitude, pulse duration, and initial interface perturbation amplitude. Previously presented scaling laws for the interface perturbation amplitude and arc length growth are tested. While we find that nominal interface strains greater than 1 occur we recognize that the applicability of this study to diagnostic lung ultrasound is limited by the accuracy of the model.

4:00

5pBAc8. Noninvasive and localized acoustic micropumping—An *in vitro* study of an ultrasound method that enhances drug distribution through a physiologically-relevant material. Ahmed Elghamrawy, Florentina de Comtes (BioEng., Imperial College London, Royal School of Mines Bldg., South Kensington Campus, London SW72BP, United Kingdom, ahmed.elghamrawy771@gmail.com), Hasan Koruk (Mech. Eng., MEF Univ., Istanbul, Turkey), Ali Mohammed (Mater., Imperial College London, London, United Kingdom), and James Choi (BIOEng., Imperial College London, London, United Kingdom)

Acoustic streaming—the displacement of fluid by sound—has been proposed as the mechanism for therapeutic effects, such as drug distribution enhancement, yet there have been no *direct* observation or characterization of this effect in soft tissue, making it difficult to optimize and control. We aimed to directly observe ultrasound-induced streaming during sonication in a tissue-mimicking material. We hypothesized that existing ultrasound phantoms (e.g., polyacrylamide (PAA) and gelatin) mimic the acoustic properties of tissue, but not the tissue microenvironment. Scanning electron microscopy revealed that gelatin and PAA had closed pores suggesting that they can't support acoustic streaming. In contrast, macroporous acrylamide (MPPA), a new phantom we created, had interconnected pores resembling the interstitial space of soft tissue. The focal point of a focused ultrasound (FUS) transducer was placed at the distal surface of the MPPA phantom. A model drug (Bromophenol blue) was injected in and around the focal region and different FUS parameters were evaluated. Dye clearance at the hydrogel surface was observed with a video camera. Increasing ultrasound pressure produced higher streaming. Pulsed ultrasound having the same total energy as continuous wave ultrasound produced higher streaming. Using MPPA, we were able to study acoustic streaming under different parameters. Our future work is to optimize ultrasound parameters to maximize acoustic

streaming in this material and in *in-vivo* tissue for enhancing drug distribution.

4:20

5pBAc9. Creating complex, controllable environments for scaffold-free tissue engineering using acoustic levitation. Brady J. Anderson (Phys., Utah Valley Univ., 852 North 50 East, Provo, UT 84604, bradyjam@hotmail.com), Natalie Sullivan (Chemistry, Utah Valley Univ., Orem, UT), Dolly A. Sanjinez, Samantha Hamner (Biology, Utah Valley Univ., Orem, UT), and Timothy E. Doyle (Phys., Utah Valley Univ., Orem, UT)

Tissue engineering has been limited by the complex structure of biological tissue. Methods have therefore been devised using scaffolds made from synthetic materials (polymers) or natural materials (extracellular matrix from cadavers or animal tissues). These scaffolds are artificially introduced materials with potentially adverse side effects. This project's objective was to determine the parameters necessary to create an environment to employ a scaffold-free method for building tissue. This method uses ultrasonic standing waves in fluid to generate complex patterns acting as a template to organize suspended cells into tissue structures. Parameters investigated included ultrasonic waveform shape, frequency, and intensity; water density and temperature; tissue cell size; transducer properties; and container configuration for standing-waves. Computer models were used to evaluate various container designs. Ultrasonic transducers from 0.20 to 1.0 MHz were coupled to containers and driven using an arbitrary waveform generator and RF amplifier. Polyethylene microspheres of 10-50 micrometer diameter were used to simulate biological cells. Experiments with cylindrical containers demonstrated that microspheres can be levitated in water to form layers, and revealed that the use of a square-wave modulated waveform generated more distinct layers. Experiments were also conducted with square containers using two transducers. These experiments produced a lattice-type structure of microspheres.

4:40

5pBAc10. Acoustical bioengineering of cortex-like constructs by levitational assembly of primate neural stem cells. Frederic Padilla, Terezija Miskic, Charlene Bouyer (LabTAU Unit 1032, INSERM, 151 Cours Albert Thomas, Lyon 69390, France, frederic.padilla@inserm.fr), Florence Wianny, and Colette Dehay (Stem Cell and Brain Res. Inst., Inserm U1208, Bron, France)

We combined bulk acoustic levitation with layer-by layer assembly to bioengineer cortex-like multilayered single constructs containing heterogeneous layers of cells, with micrometer-scale control over biological and structural features. The constructs were obtained by acoustically levitating, in a mix of fibrinogen and thrombin, primate Neural Stem Cells (NSCs) derived from monkey embryonic stem cells that stably express a tau-GFP fusion protein, enabling detailed visualization of living cell morphology, including dendrites and axons. The levitation process was repeated several times to superpose, in a single final construct with an interlayer distance of 180μm, the NSCs at three stages of differentiation, corresponding to brain progenitor cells and immature cerebral cortex neurons. The constructs were then cultured in neuronal differentiating medium, fixed and immunostained for confocal microscopy analyses. Immunostaining analyses revealed cells exhibiting neuronal and glial phenotypes. Complex intercellular connections formed by neurons and glial cells, and extending processes and inter- and intra-layers connections between layers of differentiated NSCs were monitored. The proposed acoustical method is potentially useful to create informative *in vitro* models of the developing primate brain, such as the multi-layered cerebral cortex, as it allows the assembly of heterogeneous cell populations, and supports proliferation and differentiation in 3D.

5:00

5pBAc11. Simulation of phased array ultrasound propagation for fluid flow regulation in enhancement of bone adaptation. Eashan Saikia (Dept. of Mech. Eng., Stony Brook Univ., 113 Light Eng. Bldg., STONY BROOK, NY 11794, eashan.saikia@stonybrook.edu), Chaudhry R. Hassan, and Yi-Xian Qin (Dept. of Biomedical Eng., Stony Brook Univ., Stony Brook, NY)

Osteoporosis and induced risk of fracture have impact on aging population and our society. It has been shown that low-intensity mechanical intervention can promote local tissue formation using non-invasive ultrasound. The goal of this work was to evaluate dynamic acoustic radiation force (ARF) induced bone fluid flow (BFF) at the site of interests using nonlinear finite element analysis. ARF simulation was first generated by a 1x5 linear array transducer and then extended to a 5x5 array submerged under degassed water used for regulating fluid pressure gradient and flow. Both acoustic and porous media constitutive models were used to generate dynamic ARF wave propagation in the tissue. Simulations were modeled on a cortical bone specimen with a signal frequency of 1MHz for a series of Duty Cycle(%) and Pulse Repetition Frequency(PRF). The results showed that the displacement values obtained against a varying PRF follows the experimental trends of (Dayton *et al*., 1997) and displacement increases significantly for a Duty Cycle increase from 12% to 15%. Induced displacements plotted against time agrees with the experimental trends (Fan *et al*., 2013). The dynamic ARF created a pressure gradient of 0.6MPa/cm resulting in a relative displacement of 8μm between two points separated by 1mm.

5:20

5pBAc12. Changes in the optical scattering and absorption spectra of *ex vivo* chicken breast tissue following exposure to high-intensity focused ultrasound. Jason L. Raymond, Robin Cleveland (Dept. of Eng. Sci., Univ. of Oxford, 17 Parks Rd., Oxford OX1 3PJ, United Kingdom, jason.raymond@eng.ox.ac.uk), and Ronald A. Roy (Dept. of Eng. Sci., Univ. of Oxford, Oxford, Oxford, United Kingdom)

Real-time acousto-optic (AO) sensing has been shown to non-invasively detect changes in *ex vivo* tissue optical properties during high-intensity focused ultrasound (HIFU) exposures. Baseline changes in optical properties have been previously measured as a function of thermal dose for chicken breast exposed to a temperature-controlled water bath (doi:10.1088/0031-9155/59/13/3249). In this work, the wavelength-dependent optical scattering and absorption coefficients of *ex vivo* chicken breast tissue exposed to HIFU were measured using an integrating sphere spectrophotometric technique employed previously. A focused HIFU transducer coupled to an acoustic lens was used to produce a large focal heating area in a thin (approximately 2 mm) section of tissue. Spatiotemporal surface temperature elevations were measured using an infrared camera and used to calculate the spatially dependent thermal dose delivered to a localized region of tissue as the exposure intensity and time were varied. Results will show how wavelength-dependent optical property changes can be used to improve the AO sensing of lesion formation during HIFU therapy as an alternative to thermometry and dosimetery. [Work supported by the F. V. Hunt Postdoctoral Fellowship, the University of Oxford, and EPSRC grant number EP/K02020X/1.]

Session 5pEA

Engineering Acoustics: Engineering Acoustics Topics IV

Alexander L. Butler, Cochair
Image Acoustics, 40 Willard St., Ste 201, Quincy, MA 02169

Ferina Saati Khosroshahi, Cochair
Vibroacoustics of Vehicles and Machines, Technische Universität München, Fakultät für Maschinenwesen, Boltzmannstraße 15, Garching b. München 85748, Germany

Contributed Papers

1:20

5pEA1. End driven bender transducer. Alexander L. Butler (Image Acoust., Inc., 40 Willard St., Ste 201, Quincy, MA 02169, abutler@imageacoustics.com), John Butler (Image Acoust., Inc., Cohasset, MA), and Victoria Curtis (Image Acoust., Inc., Quincy, MA)

Piezoelectric bender transducers provide a low frequency resonance response from a simple compact structure. They often take the form of bilaminar or trilaminar piezoelectric ceramic discs or beams. The bending occurs in the piezoelectric ceramic discs or beams mounted on a simple support with the piezoelectric ceramic layers operating in phase oppositions creating the bending. The low tensile strength of the ceramic under hydrostatic pressure can be an issue with this type of transducer. We present here an alternative means for bender transduction in which the bending beam is a higher strength material, such as a metal, and the piezoelectric active section is now the end support, replacing the simple support of the legacy benders. The end supporting piezoelectric material is driven with the inner section operating out of phase with the outer section causing the bender beam to rotate on this piezoelectric simple support. In this design the major bending occurs in a higher tensile strength material capable of operating under high hydrostatic pressures. We present low frequency designs optimized for operation when end-driven by various piezoelectric materials. Typical transducer parameters and metrics are presented along with the transmitting response.

1:40

5pEA2. Parameter variations for a porous layer simulation. Ferina Saati Khosroshahi (Chair of VibroAcoust. of Vehicles and Machines, Technische Universität München, Fakultät für Maschinenwesen, Boltzmannstraße 15, Garching b. München 85748, Germany, ferina.saati@tum.de), Lennart Moheit, and Steffen Marburg (Chair of VibroAcoust. of Vehicles and Machines, Garching b. München, Bavaria, Germany)

Porous media are widely known in the areas concerning noise absorption and interior and exterior acoustics. Among the numerical techniques to simulate a porous layer is finite element method using commercial software exist and actively update the range of models offered. In this work, an investigation is done using a no-flow basic problem. By variating material pore complexity and properties, different models are studied to enhance the understanding of the various treatments.

2:00

5pEA3. Modeling of 1D acoustic device composed of a planar beam and a fluid gap with discontinuity in thickness. Petr Honzík (Faculty of Transportation Sci., Czech Tech. Univ. in Prague, Konviktská 20, Praha 11000, Czech Republic, honzikp@fd.cvut.cz), Antonín Novák, Stephane Durand, Nicolas Joly, and Michel Bruneau (LAUM, UMR CNRS 6613, Université du Maine, Le Mans, France)

Precise modeling of 1D acoustic devices (passive or active, miniaturized or not) containing a planar beam loaded by a fluid gap and cavities is of interest in a variety of applications (transducers, acoustic filters, metamaterials, etc.). An analytical approach presented herein enables to describe the vibration of the planar elastically supported rigid beam of rectangular cross-section surrounded by very thin slits and loaded by the fluid gap, which is divided in three parts of different thicknesses (the central part being the thinner one), the thermoviscous losses originating in the fluid being taken into account. Comparing to the case of the fluid gap of uniform thickness, such a geometry provides more parameters which can be adjusted in order to achieve the required behavior (resonant or damped, etc.). The analytically calculated beam displacement is presented and compared to the numerical solution provided by finite element method (a reference against which the analytical results are tested).

2:20

5pEA4. Miniature accelerometers for sensing middle ear motion. Alison Hake, Chuming Zhao, and Karl Grosh (Mech. Eng., Univ. of Michigan, 2350 Hayward St., Ann Arbor, MI 48109, aehake@umich.edu)

Acceleration sensors are prevalent in handheld electronics, navigation systems, and physical activity monitors as well as for vibration measurement. There are many measurement applications, such as sensing human middle ear motion, that require miniature, low-noise sensors to detect low-amplitude vibration. Both capacitive and piezoresistive MEMS accelerometers have been used for vibration sensing of the middle ear. Recent capacitive designs are large (base area of 2.5 mm x 6.2 mm) and have a limited bandwidth of 6.44 kHz. Low-noise piezoresistive devices power requirements are high for hearing instruments, greater than 1 mW. Commercial piezoelectric accelerometers are a viable means for vibration detection; while small, they are still large and lacking proper electronic connections for middle ear sensing applications. The aforementioned problems motivate the design of a small, passive, broadband accelerometer with low input referred noise (IRN). In this work, we design a micromachined, piezoelectric bimorph cantilever accelerometer for broadband, low-amplitude vibration sensing. An analytic model is developed to minimize the IRN. The analytic expression allows for rapid design of many system parameters that are verified through finite element analysis.

2:40

5pEA5. Feasibility study on a highly-directional parametric array loudspeaker applicable to mobile devices. Kyounghun Been (Mech. Eng., Pohang Univ. of Sci. and Technology(POSTECH), PIRO 416, Pohang Univ. of Sci., Hyoja-dong, Nam-gu, Pohang-si, Gyeongbuk 790-784, South Korea, khbeen@postech.ac.kr), Hongmin Ahn (Mech. Eng., Pohang Univ. of Sci. and Technology (POSTECH), Pohang, Gyeongsangbuk-do, South Korea), Jin-Young Kim, In-Dong Kim (Elec. Eng., Pukyung National Univ., Busan, South Korea), Joung Hoa Kim, Gibae Lee, Jinho Bae, Chong Hyun Lee (Ocean System Eng., Jeju National Univ., Jeju, South Korea), and Wonkyu Moon (Mech. Eng., Pohang Univ. of Sci. and Technology (POSTECH), Pohang, Kyungbuk, South Korea)

A conventional parametric array (PA) loudspeaker has a relatively small aperture and produces a highly directional audible sound beam. Due to limitations arising from their size and power consumption, standard PA loudspeakers are not used in mobile devices. PA loudspeakers are typically larger than 25 cm and require more than 60 W. We show that piezoelectric micro-machined ultrasonic transducer (pMUT) arrays can emit highly directional ultrasonic sounds with a power efficiency of up to 80% [Je and Moon, Ultrasonics 53, 1124-1134 (2014)]. Highly directional sound beams with a frequency bandwidth of 13 kHz can be generated efficiently by pMUT arrays with two resonant frequencies [Je and Moon, JASA 173(4), 1732-1743 (2015)]. We used a signal processor, power amplifier, and pMUT array to build a loudspeaker system that is smaller than, and requires less power than, a conventional PA loudspeaker system. In our loudspeaker system, the maximum sound pressure level (SPL) is 75 dB, ± 3 dB frequency bandwidth is 12.5 kHz, and the total power consumption is 3.4 W. Hence, this loudspeaker system could potentially be used in mobile devices. [This work was supported by Samsung Research Funding Center, SRFC-IT1404-00.]

3:00

5pEA6. Contactless angular position control of objects using acoustic vortex beams in air. Ruben D. Muelas H., Jhon F. Pazos-Ospina, and Joao L. Ealo (School of Mech. Eng., Universidad del Valle, Universidad del Valle, School of Mech. Eng., Bldg. 351, Cali, Colombia, ruben.muelas@correounivalle.edu.co)

Acoustic vortices (AV) are progressive and special fields exhibiting a helical wavefront. One of their most important features is the ability to transfer angular momentum to matter. Using AV, emerging technologies are aiming to non-contact manipulation, micro assembly and induced rotation of objects, among other applications. By controlling the amount of energy emitted, the topological charge and the size of the AV is possible to control/adjust the angular position of objects without touching them, as long as they can stably rotate. In this work, we show experimental results on the control of the angular position of disk-like samples (diameters between 5 cm and 10 cm) by means of an AV generated using a phase array system along with a 30-element multitransducer operating at 40 kHz in air. The samples were located 10 cm far from the AV generator. In particular, we analyze the effects on the dynamics of the controlled samples due to changes in the topological charge, the amount of energy radiated and the delay law used to generate the AV. Special attention is paid to the relative size of the AV with respect to the sample. Position errors lower than 1 degree are possible for the regulation problem.

3:20–3:40 Break

3:40

5pEA7. A "composite" stepped plate for use in highly directional parametric array loudspeakers. Yonghwan Hwang, Chayeong Kim, Wonkyu Moon (Dept. of Mech. Eng., Pohang Univ. of Sci. and Technology (POSTECH), PIRO416, Postech, Hyo-ja dong, Nam gu, Po hang KS010, South Korea, serenius@postech.ac.kr),

It has been shown that stepped plate radiators with appropriately designed steps can be used in highly directional parametric array (PA) loudspeakers [Hwang and Moon, 170th Meeting of the Acoustical Society of America, 138(3), 1801(2015)]. To improve the performance of the PA loudspeaker, a relatively thin plate with steps needs to be adopted for its higher power efficiency. However, the natural frequencies of the plate and the corresponding mode shapes depend on the mass of the steps. Hence, it can be difficult to design a stepped plate suitable for use in PA loudspeakers. In this paper, we propose a novel plate design, with steps composed of a stiff, lightweight, polymer material. The steps were fabricated with a thickness of 2.3 mm from a polymer material with an elastic modulus of 1 GPa, a Poisson ratio of 0.23, and a mass density of 540 kg/m^3 on the plate with a thickness of 1 mm. Experiments and simulations were carried out to determine the acoustic characteristics of our fabricated "composite" stepped plate. [Work supported by NRF 2016R1E1A2A02945515).]

4:00

5pEA8. On timbre in urban soundscapes: The role of fountains. Laura Velardi, Jean-Pierre Hermand (LISA - Environ. HydroAcoust. Lab, Université libre de Bruxelles, av. F.D. Roosevelt 50, CP165/57, Brussels 1050, Belgium, lvelardi@ulb.ac.be), and Roberto D'Autilia (Architettura, Università di Roma Tre, Rome, Italy)

According to European Environmental Noise Directive (2002/49/EC, 25 June 2002), acoustical issues in urban planning are mostly treated in term of loudness. However, timbre of urban sound sources, including their spectral features but not only, can affect our daily life due to auditory preferences. Social studies identify the sound of water as one of the most enjoyable sounds by humans and laboratory experiments demonstrate that testers prefer soft types of water sounds, with low frequency contents and low flow rates, such as natural stream and fountain. Recent surveys acknowledge fountains as acoustical urban elements, focusing on sound pressure level and spectrum analysis. In this paper, we will discuss observations made in Rome (e.g., Capitoline Hill) in 2015 and recently in Brussels, that suggest that fountain sounds may change habits and paths of passers-by. We will then present first findings from statistical analysis of the audio samples collected *in situ*, in an attempt to identify attributes of timbre that are responsible for shifted habits.

4:20

5pEA9. Application of the algorithm for time of arrival estimation of the shock waves produced by projectiles of different calibers. Miodrag S. Vračar (Military Tech. Inst., Ratka Resanovića 1, Belgrade 11030, Serbia, vracarmiodrag@mts.rs)

The topic of this paper is review of application of the algorithm for time of arrival estimation of shock waves produced by projectiles of different calibers. Time of arrival estimation of the shock waves is based on algorithm, which consists of wavelet decomposition, identification of the narrow data segment, which holds shock wave by wavelet coefficients and estimation the time of arrival of the shock wave by conventional statistical methods. Sources of the shock waves were projectiles which caliber varied from characteristic rifles, 7.62 mm, through large gun's calibers up to 120 mm. Quality of the time of arrival estimate is verified on significant number of experiments. Estimated time of arrival is used in solving of the position of weapon when fire off happened. Localization of the shock wave sources was done by networked and in space distributed acoustic sensors. Discrete probability method, based on time differences of the time of arrivals obtained from multiple synchronized acoustic sensors, was used for localization of the weapon and these results are compared with their locations obtained by DGPS method.

4:40

5pEA10. Generalized finite difference time domain simulation on acoustical moving boundary problems. Song Wang (Music Technol., McGill Univ., 555 Rue Sherbrooke Ouest, Montreal, QC H3A 1E3, Canada, song.wang5@mail.mcgill.ca) and Qingzhi Hou (School of Comput. Sci. and Technol., Tianjin Univ., Tianjin, China)

Mesh-free methods have the potential to solve acoustic problems with moving boundary, complex geometries and in flowing fluids. In this paper, generalized finite difference time domain (GFDTD), a time-domain representation of the generalized finite different method (GFDM), is extended to solve acoustic wave problems with moving boundary. To explore the numerical performance of GFDTD, its stability and phase velocity for wave propagations are firstly analyzed by von Neumann method. Compared with the other two widely used mesh-free methods, smoothed particle hydrodynamics (SPH) and modified smoothed particle hydrodynamics (MSPH), theoretical analysis shows that GFDTD has less dispersion, which is also well verified by 1D numerical tests. The GFDTD method is then applied to simulate wave propagation inside a 2D cylinder and the results are compared with those obtained by traditional finite difference time domain (FDTD) method. Finally, a moving boundary condition is introduced into the problem domain and the corresponding input impedance is analyzed.

5:00

5pEA11. Design of quieter kitchen appliances: Sound pressure level modeling and validation of a household refrigerator using statistical energy analysis. Roberto Zarate (Mech., Universidad Nacional Autónoma de México , San Miguel de Allende, Guanajuato, Mexico), Edgardo Matus (Mabe, Querétaro, Querétaro, Mexico), Marcelo Lopez (Mech., Universidad Nacional Autónoma de México , Juriquilla, Querétaro, Mexico), and Luis Ballesteros (Mabe, Acceso B, 406, Parque industrial Jurica, Querétaro, Querétaro 76120, Mexico, Luis.Ballesteros@mabe.com.mx)

The design of efficient and quieter kitchen appliances represents a significant challenge for manufacturers. In domestic refrigerators, the compressor, condenser's fan, and evaporator's fan are important sources of noise. This paper presents the statistical energy analysis (SEA) carried out to estimate the sound pressure level generated in a domestic refrigerator's evaporator subsystem. The work focuses on the evaporator's fan and the subsystem's mechanical assembly; these constitute the main source of noise and path for vibration transmission. The results of the analysis for a frequency band noise between 100 Hz and 10 kHz are reported. A semi-anechoic chamber was used to validate the SEA analysis presented herein; the authors also set up an experiment using microphones and accelerometers complying with standard ISO-3745, achieving a correlation of up to 96%. The analysis and experimental data were then used to produce a set of design guidelines to help kitchen appliance designers in the configuration of innovative cabinet geometries and the implementation of noise reduction strategies in household refrigerators.

THURSDAY AFTERNOON, 29 JUNE 2017 ROOM 204, 1:15 P.M. TO 4:40 P.M.

Session 5pED

Education in Acoustics and Student Council: Teaching Tips for the New (or Not So New) Acoustics Faculty Members

Daniel A. Russell, Chair
Graduate Program in Acoustics, Pennsylvania State University, 201 Applied Science Bldg., University Park, PA 16802

Chair's Introduction—1:15

Invited Papers

1:20

5pED1. Diving into the deep end: Transition from student to professor. Whitney L. Coyle (Phys., Rollins College, 1000 Holt Ave., Winter Park, FL 32789, wcoyle@rollins.edu)

They kept saying don't worry—one cannot be taught how to teach. It takes experience, successes and failures, good and bad advice, and most of all, time. For a Ph.D. student this is terribly frustrating to hear. After 24 years of school, your job will now consist of a skill that cannot be mastered by studying. This aspect of a career in academia is one most are not prepared to face. Many are not willing to take that dive into the deep end and figure out how to make that transition successfully from student to professor due to an apparent lack of preparation—due to not being taught how to teach. It is not easy, but after two years in the game, I know it is worth it. This talk will focus on my experience throughout my recent start as a physics professor at a small liberal arts college. I will share some of the situations I felt prepared for (few) and those I was not (most). I will discuss my experience beginning research with undergraduate students, building a lab, writing lecture notes, and finding my voice as a new professor in the department and school.

1:40

5pED2. Collaboration, not oration: Tips for engaging students and promoting active learning in large classrooms. Laura Kloepper (Biology, Saint Mary's College, 262 Sci. Hall, Saint Mary's College, Notre Dame, IN 46556, lkloepper@saintmarys.edu)

Students state that traditional lectures are boring, and studies show that traditional lectures are ineffective and result in higher student failure rates than in active classroom settings. Unfortunately, large class sizes at many institutions prohibit some of the active learning strategies that are feasible in small classrooms. Moreover, increased time constraints of faculty members further limit one's ability to assess individual learning in the classroom. Here, I demonstrate multiple techniques to incorporate into traditional lecture formats that

increase student engagement, provide immediate assessment of lecture material, deepen critical thinking, and motivate student learning—all without increasing the time demands of the instructor.

2:00

5pED3. Using a flipped-classroom to teach shock and vibration in naval systems. Andrew R. Barnard (Mech. Eng. - Eng. Mech., Michigan Technolog. Univ., 815 R.L. Smith MEEM Bldg., 1400 Townsend Dr., Houghton, MI 49931, arbarnar@mtu.edu)

A new course, Shock and Vibration, was taught at Michigan Technological University in the spring of 2017. The senior elective course in mechanical engineering was designed to give students insight into shock and vibration problems, with specific applications in naval and marine systems. The course design used a flipped-classroom approach exclusively, meaning the students watched recorded lectures prior to each class session and in-class time was used for group problem solving, quizzes, demonstrations, and discussions. A laboratory session was also included in the course. This talk will give a brief overview of the course material, cover the pros and cons of the flipped-classroom, and provide some pointers from a first-timer's perspective.

2:20

5pED4. Reading, writing, and arithmetic: Encouraging students to read textbooks and take organized notes. Daniel A. Russell (Graduate Program in Acoust., Penn State Univ., 201 Appl. Sci. Bldg, University Park, PA 16802, drussell@engr.psu.edu)

Common practice for many graduate level physics and engineering courses is for professors to teach using traditional lectures to a student audience while using textbooks as primary references. Traditional lecture styles often fall at one of two extremes: a more historical approach where a professor writes notes on a board while students dutifully copy those notes down, or a conference style approach where lecture content is presented with PowerPoint slides while students watch without taking notes. This talk will describe two approaches the author uses in the teaching of graduate level acoustics courses: (i) to encourage students to read material from relevant textbooks prior to the topic being covered in greater depth during lecture; and (ii) a middle ground lecture method using partial notes where students are given a set of well-organized, but very incomplete, handouts which are annotated during a class session. Discussion will include examples, as well feedback from professor and students regarding the effectiveness of these approaches. While these methods are used by the author for graduate level acoustics courses, they may be effectively adapted for teaching undergraduate courses as well.

2:40

5pED5. Design of homework problems in physical acoustics. David T. Blackstock (Appl. Res. Labs., Univ. of Texas at Austin, PO Box 8029, Austin, TX 78713-8029, dtb@austin.utexas.edu) and Mark F. Hamilton (Dept. of Mech. Eng., Univ. of Texas at Austin, Austin, TX)

A variety of methods exist to help students master material in their acoustics courses. The method chosen depends a great deal on the course. In the graduate acoustics courses at UT Austin the methods include system identification and design projects in the course on acoustical transducers, measurements, and data analysis in the courses on architectural acoustics and ultrasonics, development of computer codes based on the sonar equation in the course on underwater acoustics, and a project usually requiring literature review in the course on nonlinear acoustics. Homework problems is the method of choice for the two-semester basic course on physical acoustics, because mastery of the fundamental principles of acoustics requires students to immerse themselves in the details specific to each particular topic. Learning basic acoustics is a lot like learning calculus: one learns by doing. This presentation discusses the philosophy behind creation of the homework problems that appear in the textbook *Fundamentals of Physical Acoustics* by Blackstock [Wiley 2000]. The problems do not require use of a computer. They do require judgment over plugging and chugging. The student must decide what section of a chapter applies to the problem assigned and also what approximations are permissible to simplify the work.

3:00

5pED6. Effective and efficient grading for assessing student learning. Daniel Ludwigsen (Kettering Univ., 1700 University Ave, Flint, MI 49504, dludwigs@kettering.edu)

For decades, quality of assessment has increasingly become a part of the discussion of good teaching in higher education. New faculty (and older ones) may be overwhelmed at the thought of comprehensive assessment of student learning at the level of individual courses and entire degree programs, on top of keeping up with grading and feedback for each student. However, if student assignments are used effectively and efficiently to assess student learning, students perceive the process as fair, grades become more meaningful, and data is generated to inform evidenced-based improvement of instruction. While some graduate programs and early-career mentoring include course design, learning objectives, and assessment of student learning, not all Ph. D. programs require training in even basic elements. New faculty may benefit from an introduction to a "backwards" approach to course design, a survey of learning assessment tools, and current best practices in grading. Examples from acoustics courses at Kettering University illustrate these ideas, with lessons learned from recent ABET accreditation in engineering and applied physics programs.

3:20

5pED7. How understanding concerns of female students helps all students. Tracianne B. Neilsen (Brigham Young Univ., N311 ESC, Provo, UT 84602, tbn@byu.edu)

Studies on the high attrition rate of female students in STEM fields have identified general concerns that can cause women to leave. These include lack of prior experiences with engineering and computing, fears of failing, not fitting in, or being negatively impacted by stereotypes, a desire to contribute to society, and worries about future work-life balance. While women may be more affected by these concerns, male students can experience them as well. Thus, attempts to address these concerns benefit all students. The most straightforward ways to tackle these concerns are the same as the primary recommendations found in the physics education research literature for increasing student self-efficacy and constitute principles of good mentoring [Gee *et al.,* Proc. Mtgs. Acoust. 23 025001 (2015)]. Foster a

cooperative, inclusive, encouraging environment. Talk openly about concerns and mistakes to help students exchange the fear of failure for a drive to succeed. Provide ways for students to gain experience that is lacking. Emphasize applications to social and environmental issues to increase motivation. Be aware of subtle biases and discredit stereotypes. Acknowledge struggles of the work-life balancing act and promote family-friendly policies. Combined these actions build a supportive learning environment for all students.

3:40

5pED8. Preparing students for undergraduate research experiences in acoustics. Andrew A. Piacsek (Phys., Central Washington Univ., 400 E. University Way, Ellensburg, WA 98926, andy.piacsek@cwu.edu) and Kent L. Gee (Phys. and Astronomy, Brigham Young Univ., Provo, UT)

At many primarily undergraduate institutions, and some PhD-granting universities, science and engineering faculty are expected to provide research experiences for undergraduates. One of the biggest challenges for faculty who supervise undergraduate research is preparing students to work productively with some degree of independence. In addition to having familiarity with relevant physical concepts and measurement methods, students must know how to use lab equipment safely, responsibly, and effectively. While required lab courses in typical physics and engineering departments expose students to general laboratory skills and measurement techniques, they do not typically include specialized equipment and methods used in acoustics research. This presentation will describe how an upper-division undergraduate course in acoustics can be designed to prepare students for a productive independent research experience in this field. Examples will be drawn from courses offered at Central Washington University and Brigham Young University.

Contributed Paper

4:00

5pED9. Incorporating local experts into a Physics of Music class. Jack Dostal (Dept. of Phys., Wake Forest Univ., P.O. Box 7507, Winston-Salem, NC 27109, dostalja@wfu.edu)

The Physics of Music is a general education science class open to all students at Wake Forest University. A broad range of topics are covered: wave physics, hearing and the ear, the voice and singing, musical instrument function and performance (winds, strings, and percussion), room acoustics, and more. An instructor for such a course is not usually a master of all these areas. In this talk, I will describe how we engage some of our local experts (singers, instrumentalists, piano tuners, doctors) to enhance the student experience of learning about physics and music. Activities to incorporate their specialties into the class will also be described.

4:20–4:40 Panel Discussion

THURSDAY AFTERNOON, 29 JUNE 2017 ROOM 203, 1:15 P.M. TO 5:20 P.M.

Session 5pNSa

Noise, Architectural Acoustics, and Engineering Acoustics: A Comparative Look at US and European Noise Policies

David S. Woolworth, Cochair
Oxford Acoustics, 356 CR 102, Oxford, MS 38655

Brigitte Schulte-Fortkamp, Cochair
Institute of Fluid Mechanics and Engineering Acoustics, TU Berlin, Einsteinufer 25, Berlin 101789, Germany

Chair's Introduction—1:15

Invited Papers

1:20

5pNSa1. Noise related calls to police departments in the United States. Leslie D. Blomberg (Noise Pollution Clearinghouse, Box 1137, Montpelier, VT 05601-1147, les@nonoise.org)

Noise is responsible for a significant number of calls to police departments. This paper seeks to quantify the scale and scope of the community noise complaints in police work. Fifty police departments were contacted to determine what percentage of "call for service" (the technical term for calls to police departments) were noise related. In addition, online data published by 150 cities were also included in the study. Noise complaints generally are responsible for one to five percent of calls to police departments.

1:40

5pNSa2. Following up the European Directive on Noise. Brigitte Schulte-Fortkamp (Inst. of Fluid Mech. and Eng. Acoust., TU Berlin, Einsteinufer 25, Berlin 101789, Germany, b.schulte-fortkamp@tu-berlin.de)

Since 2002 the EU Directive on Noise (END) is available for the European member states followed by a number of activities to guarantee the protection against noise. So-called noise maps and action plans have been developed with regard to the different noise sources but also depending from the structure of the areas involved. This paper will describe the development in different member states, but namely in Germany to inform about the frame of duties and use of the guideline. It will also discuss how far the involvement of people concerned is part of the development. Furthermore, the recent review of the END will be presented.

2:00

5pNSa3. Effective abatement of railway noise in Germany. Rene Weinandy (Noise Abatement in Transport, German Environment Agency, Woerlitzer Platz 1, Dessau-Rosslau 06844, Germany, rene.weinandy@uba.de)

Many people are exposed to high levels of noise that adversely affect their health and quality of life. Noise is now experienced virtually every-where and around the clock. Therefore, noise is an important environmental issue in Germany and within the EU. Under the Directives on rail traffic across Europe, the EU Commission has established pan-European noise thresholds for new types of rolling stock in the Technical Specifications for Interoperability (TSI). The most important consequence of the noise thresholds is that particularly noisy rolling stock fitted with cast iron block brakes is no longer permitted. The challenge is now to replace these brakes in especially noisy freight wagons in the current rolling stock with quieter braking systems. As a financial incentive for rail operators to refit such rolling stock with quieter brakes, track access charges are implemented in Germany. Additionally, there are a number of technical and legal measures available. The presentation will give an overview on these measures to efficiently abate railway noise in Germany.

2:20

5pNSa4. Enforcement of local community noise ordinances in the United States. Leslie D. Blomberg (Noise Pollution Clearinghouse, Box 1137, Montpelier, VT 05601-1147, les@nonoise.org)

This paper is a follow-up of *The Analysis of 500 Noise Ordinances*, presented at the 171 Meeting of the Acoustical Society of America. In the first paper, the noise ordinances from the 500 largest communities in the United States were analyzed with respect to the regulatory tools, metrics and criteria communities use to regulate noise. The prevalence of various techniques employed in noise regulations were presented and discussed. The regulatory tools and techniques analyzed included pressure level limits, as well as nuisance, disturbing the peace, plainly audible, minimum distance, and time of day based regulations. In this paper, a random sample of the 500 noise ordinances were chosen, and the police departments in those cities were contacted, to determine how the ordinances were enforced, what regulatory tools were relied upon in enforcement, and why those tools and not others in the noise ordinance were used. Particular attention is given the role of decibel based regulatory tools including A-weighted, C-weighted, and octave band metrics in police enforcement of noise ordinances.

Contributed Papers

2:40

5pNSa5. Online commentary on noise concerns, policy, and enforcement among readers in four countries. Jeanine Botta (SUNY Downstate Medical Ctr. School of Public Health, 720 East 31st St., Apartment 7G, Brooklyn, NY 11210, jeanine.botta@downstate.edu) and Karl H. Raab (The Right to Quiet Society for Soundscape Awareness and Protection, Vancouver, BC, Canada)

Decades' worth of research findings tell us that most who are affected by noise exposure do not complain about it, and that noise complaints are not an accurate means of measuring community response. Much attention has been given to understanding the motivations of serial complainers, with less given to non-complainers. Are non-complainers unaware of noise policy, or how to submit a noise complaint? Research indicates that this is the case, and also suggests that some people think complaining would be a waste of time, that some are inhibited by social factors, and that some give up when previous efforts to address noise do not succeed. While noise complaints do not accurately measure community response, some policy and enforcement decisions are influenced by noise complaint data. Are there other potential means of capturing a measure of community response? This paper examines an unexplored resource for noise-related data online, where posts cover a broad range of experiences and opinions. Collecting reader comments in response to online news stories and blog posts, the paper uses content analysis to identify common themes, finding similar and disparate social expectations about noise policy and enforcement among readers from Canada, the United States, the United Kingdom, and Germany.

3:00–3:20 Break

3:20

5pNSa6. Community perception of noise impacts—A comprehensive review of background sound data for setting appropriate design goals for source sound (continuous, impulsive, tonal). Aaron M. Farbo (Cavanaugh Tocci Assoc., 327F Boston Post Rd., Subury, MA 01776, afarbo@cavtocci.com)

Community perception of noise impact is most often influenced by the amount by which a new source of sound exceeds the existing ambient sound level. Various descriptors are used to define existing ambient sound levels. For example, the Massachusetts DEP Noise Policy uses the 90th percentile sound level (L_{A90}), while the New York State DEC Noise Policy uses the equivalent sound level (L_{Aeq}). When new transient sounds are introduced into an environment the L_{A90} or L_{Aeq} may not be the best, or only, metrics for defining a design goal or noise ordinance limit. In addition to the lowest existing ambient sound levels, reviewing the first percentile (L_{A01}) sound levels may be a more appropriate way to determine community perception of new transient sounds. Sound surveys from across the country were analyzed to determine maximum-recommended community sound levels (design goals). These design goals are based on marginal increases to the ambient sound level and comparison to existing maximum sound levels (L_{A01}); the specific margin depends on the characteristics of sound experienced at the receptor.

3:40

5pNSa7. The potential of comprehensive reference to acoustic overall scenarios and their subjective experience with regard to urban sound planning. Joachim Scheuren, Beate Altreuther, and Sonia Alves (Mueller-BBM GmbH, Robert-Koch-Str. 11, Planegg D 82152, Germany, Joachim.Scheuren@mbbm.com)

The refinement of descriptors for subjective sound assessments and growing recognition that comfortable sound environments definitely need more than a limitation of sound levels have changed the scope of engineering acoustics over the last decades. Instead of focusing on objective physical parameters more and more emphasis has been put on understanding and promoting the subjective experience of acoustic environments. However, to what extent such approaches have been applied to practice strongly depends on the applicative environment. While subjective criteria for product sound quality are widely accepted if they immediately serve the user, they apparently face difficulties if applied to the benefits of uninvolved but affected third parties. After a short review of underlying motivations the potential of comprehensive, holistic approaches which relate to the full spectrum of acoustic scenarios from the point of view of subjective experiences will be outlined and demonstrated with particular reference to urban sound planning, the acoustic planning for urban environments.

4:00

5pNSa8. Suggestions for a future ANSI sleep disturbance standard. George A. Luz (Luz Social and Environ. Assoc., Inc., 288 Ninth St. Alley, Ashland, OR 97520, luz_associates@msn.com)

Recently, the American National Standards Institute (ANSI) withdrew the standard for sleep disturbance from transportation noise which it had published in 2008. This standard failed because the original working group had adopted a (Skinnerian) "black box" approach to describe the behavior of two intertwined neural systems: (1) an energy-dependent defense response and (2) an information-based threat analysis network. This paper reviews what brain scientists have discovered about the information-based network in the years between the publication of the standard and its withdrawal and argues that it is essential for a future standard to address the role of slow wave sleep. This assertion is followed by a discussion of the pros and cons for adapting current models of noise-induced disturbance of slow wave sleep (Basner, McGuire, Davies and others) into a future standard. Recommendations for a future standard include: (1) break with the logic of the National Environmental Policy Act of 1969, (2) focus on protecting the most vulnerable, (3) address all types of anthropogenic sound, (4) provide for cost cutting through emerging technology, and (5) include procedures for mitigation.

4:20

5pNSa9. Implementing Zwicker sharpness using specific loudness calculated from the "Procedure for the Computation of Loudness of Steady Sounds." S. Hales Swift (Phys. and Astronomy, Brigham Young Univ., 2286 Yeager Rd., West Lafayette, IN 47906, hales.swift@gmail.com) and Kent L. Gee (Phys. and Astronomy, Brigham Young Univ., Provo, UT)

Sharpness as defined by Fastl and Zwicker [2007] is a measure of the weighted spectral balance of a sound, i.e., sounds with relatively greater high-frequency content have a greater sharpness than those with proportionally greater low-frequency content. The sharpness percept also influences the overall "pleasantness" of a sound and is thus important to sound quality. Measures of sharpness have been predominantly constructed to work with Zwicker's loudness model [DIN 45631/ISO 532B], with less attention to the standardized model given in "Procedure for the Computation of Loudness of Steady Sounds" [ANSI S3.4-2007]. In this paper, a mathematical method for adapting Zwicker's sharpness measure for use with the ANSI S3.4-2007 loudness standard (and other loudness metrics producing a specific loudness distribution) is discussed. Benchmarking results for the resultant metric and potential limitations of this approach are also addressed.

4:40–5:20 Panel Discussion

Session 5pNSb

Noise, Architectural Acoustics, Speech Communication, and Psychological and Physiological Acoustics: Effects of Noise on Human Comfort and Performance II

Z. Ellen Peng, Cochair
Waisman Center, University of Wisconsin-Madison, 1500 Highland Avenue, Madison, WI 53711

Lily M. Wang, Cochair
Durham School of Architectural Engineering and Construction, University of Nebraska - Lincoln, PKI 100C, 1110 S. 67th St., Omaha, NE 68182-0816

Anna Warzybok, Cochair
Department of Medical Physics and Acoustics, Medical Physics Group, University of Oldenburg, Universität Oldenburg, Oldenburg D-26111, Germany

Invited Papers

1:20

5pNSb1. Speech intelligibility and listening effort in every-day listening situations. Inga Holube, Petra von Gablenz, Sven Kissner, and Joerg Bitzer (Inst. of Hearing Technol. and Audiol., Jade Univ. of Appl. Sci., Ofener Str. 16/19, Oldenburg D-26121, Germany, Inga.Holube@jade-hs.de)

Noise and reverberation affect speech intelligibility and increase listening effort. The impact is more severe for hearing-impaired listeners than for normal-hearing listeners and might lead to less social interactions, and reduced quality of life. While lab experiments are well controlled to get reliable outcomes, every-day listening situations are far more complex. Obtaining objective data of acoustical characteristics outside a laboratory is difficult, given the required equipment and its proper handling as well as privacy concerns emerging from audio recordings in a non-regulated and populated environment. Therefore, we developed a privacy-aware smartphone-based system that allows for long-term ecological momentary assessment. This system combines descriptions of the environment and subjective ratings on predefined scales with objective features derived from the acoustical signal. In a field study, forty-seven elderly listeners used the system for about four consecutive days. Results show that the listeners spend most of their time in environments with quite high signal-to-noise ratios resulting in high speech intelligibility ratings and low listening effort. More demanding situations are comparatively rare and include restaurant or car environments. The presentation summarizes the study results and discusses their contribution to further developments of today's lab experiments towards more natural listening settings.

1:40

5pNSb2. Evaluating near-end listening enhancement in noise for normal-hearing and hearing-impaired listeners. Jan Rennies (Hearing, Speech and Audio Technol., Fraunhofer IDMT, Cluster of Excellence Hearing4All, Marie-Curie-Str. 2, Oldenburg 26129, Germany, jan.rennies@idmt.fraunhofer.de), Henning Schepker, David Huelsmeier, Jakob H. Drefs, and Simon Doclo (Medical Phys. and Acoust., Univ. of Oldenburg, Cluster of Excellence Hearing4All, Oldenburg, Germany)

Speech announcements in public spaces are often impaired by environmental noise, e.g., in train stations. In such conditions it is usually not possible to reduce the noise at the listener's location, but the target speech can be pre-processed to enhance speech intelligibility (near-end listening enhancement, NELE). This contribution presents a series of listening tests to evaluate a NELE algorithm with normal-hearing and hearing-impaired listeners. The algorithm consists of several processing stages including dynamic range compression and frequency shaping. The algorithm either works under an equal-rms-power constraint (same rms level of processed and unprocessed speech) or additionally includes an adaptively controlled increase in rms level at constant peak level. Speech intelligibility was measured using the Oldenburg sentence test with unprocessed and processed stimuli in different types of background noise and SNRs. The results indicate that large benefits can be obtained even for constant speech level, but this benefit strongly depends on background noise type and SNR. The benefit is generally smaller for hearing-impaired than for normal-hearing listeners, but interindividual differences are quite large. All subjects benefit from the adaptive rms level increase, indicating that NELE algorithms with equal-rms-power constraints may not be sufficient for hearing-impaired listeners.

2:00

5pNSb3. Self-fitting hearing aids reveal strong effects of sound quality and comfort. Peggy B. Nelson, Danne VanTasell, and Trevor T. Perry (Univ. of Minnesota, 164 Pillsbury Dr Se, Minneapolis, MN 55455, peggynelson@umn.edu)

Noisy rooms present difficult challenges for listeners with hearing loss, and sensory aids have limitations in overcoming those challenges. New technology from EarMachine(c) allows for individuals with hearing loss to self-adjust hearing aid amplification parameters. This technology can provide data to empirically answer questions about listener preferences in noise. Results from 30 listeners demonstrated gain profiles revealing extremely large individual preferences that are not well predicted by age, gender, or hearing loss. Preference judgments suggest that sound quality and comfort are important variables that can be achieved without significant loss of speech intelligibility.

2:20

5pNSb4. Influence of multiple moving noise distractors on spatial release from masking. Rhoddy Viveros (RWTH, Im Grüntal 35, Aachen 52066, Germany, rvm@akustik.rwth-aachen.de), Z. Ellen Peng (Univ. of Wisconsin–Madison , Madison, WI), and Janina Fels (RWTH, Aachen, Germany)

Previous studies on speech in noise are often limited to studying stationary sound sources in the virtual acoustic scene. Moving toward a more realistic acoustic environment, this study aims at expanding the knowledge of spatial release from masking (SRM) involving moving noises. Current mathematical models describe contributions of distractor asymmetry (on the same or different sides) and angular separation with the target in predicting SRM (e.g., Bronkhorst, 2000). In a previous study, we quantified SRM from single distractor traveling on a 90° trajectory. In this study, we add a second distractor to investigate how speech understanding is affected by distractors in movement in various trajectory angles, where only binaural cues are available. A speech-in-noise test with normal-hearing adults is performed using virtual binaural reproduction. Listeners are asked to identify spoken words by target in front (0° azimuth) under the presence of babble-noise distractors. Speech reception thresholds at 50% intelligibility are measured in 14 test conditions from 2 distractor spatial configurations (symmetrical vs. asymmetrical) X 7 trajectories (stationary at 0° and 90° and moving 15°, 30°, 45°, 60°, and 90°). Results will be presented to understand the role of binaural cues in speech understanding under the presence of moving noises.

Contributed Papers

2:40

5pNSb5. Speech reception in fluctuating noise: Which is the benefit? Nicola Prodi and Chiara Visentin (Dipartimento di Ingegneria, Università di Ferrara, via Saragat 1, Ferrara 44122, Italy, nicola.prodi@unife.it)

Speech intelligibility in a fluctuating noise is widely acknowledged to be higher than in a stationary noise with the same long-term level, as listeners take advantage of the information in the masker dips. Anyway, it is demonstrated in a companion paper (*On the relationship between a short-term objective metric and listening efficiency data for different noise types*, Proc. of Acoustics'17) that when the two noises are described with a short-term metric, able to properly follow their run-time course, the same intelligibility is found in both maskers. In fluctuating (ICRA) noise the accuracy result is reached involving a greater amount of cognitive resources and the background noise is perceived as more effortful. In this study the same methodology, based on the joint assessment of accuracy results and cognitive load, is applied in two real life scenarios (a canteen and a café), where speech reception is impaired by diffuse, non- informative, fluctuating maskers. Impulse responses and noise samples were collected in the environments and rendered in a silent laboratory room; listening tests were presented to a panel of normal-hearing young adults. The results in real environments are compared with simulations in the same listening conditions, with ICRA noise as a masker.

3:00–3:20 Break

3:20

5pNSb6. Face the music: Classical music students and the sound of performance. Stephen Dance (The Built Environment and Architecture, London South Bank Univ., London South Bank University, 103 Borough Rd., London SE1 0AA, United Kingdom, dances4u2003@yahoo.co.uk)

Since the implementation and enforcement of the European Union Physical Agents Directive (Noise) the Acoustics Group has collaborated with the Royal Academy of Music creating the noise team formed from administrators, scientists, and senior management. Our challenge was to allow these highly talented artists to practice, rehearse, and perform safely during their time at the Royal Academy of Music. This ten year project involved more than 3000 musicians measuring sound exposure of each instrument group and the hearing acuity of every student, as well as hearing surveillance of a sample of graduates. At each occurrence, the students were questioned as to their aural environment. The paper will focus upon the hearing acuity of undergraduates after studying music for a period of four years.

3:40

5pNSb7. MRI investigation of the ear canal deformation due to earplugs: A first step toward understanding wearing comfort. Simon Benacchio (IRSST, 505 Boulevard de Maisonneuve O, Montréal, QC H3A 3C2, Canada, Simon.Benacchio@irsst.qc.ca), Arthur Varoquaux, Arnaud Le Troter (Aix Marseille Université, CNRS, CRMBM/CEMEREM UMR 7339, Marseille, France), Eric Wagnac, Olivier Doutres (Ecole de technologie supérieure, Montréal, QC, Canada), David Bendahan, Virginie Callot (Aix Marseille Université, CNRS, CRMBM/CEMEREM UMR 7339, Marseille, France), and Franck C. Sgard (IRSST, Montreal, QC, Canada)

The effective protection of earplugs is related to the acoustical attenuation but also to the comfort they provide. Earplugs are acoustic seals used to reduce sound transmission in the ear canal. A perfect contact between ear canal and earplug should ideally avoid acoustic leaks leading to under-protection. Practically, this acoustic seal is achieved applying a greater or lesser deformation on the ear canal walls which depends on the earplug insertion and type. However, a too large deformation can modify the acoustic attenuation and cause physical pain due to mechanical pressure exerted onto ear canal walls leading the wearer to take earplugs off and reducing their protection. This study aims at investigating the ear canal deformation of one human subject due to insertion of various earplugs known for providing different levels of protection and different levels of comfort. The shape of the open and occluded right and left ear canals of the subject is measured using MRI technology. The deformation along the ear canal is estimated using an imaging registration method. The results show large displacements at the ear canal entrance for all earplugs and non negligible displacements after the first bend of the ear canal for some of them.

4:00

5pNSb8. How to reduce energy production plant noise annoyance using a sound design approach. Laurent Brocolini (ETIS - UMR 8051 Université de Cergy-Pontoise St-Martin, 11 rue Jean Mentelin, Strasbourg 67200, France, laurent.brocolini@gmail.com), Catherine Lavandier (ETIS - UMR 8051 Université de Cergy-Pontoise St-Martin, Cergy-Pontoise, France), Isabelle Schmich-Yamane, and Marion Alayrac (EDF - DTG, Grenoble, France)

Industrial plant noise is the result of a combination of many various sound sources. The spectral composition of these sound sources varies depending of the equipments: wide-band noises mainly related to cooling towers, tonal noises such as transformers and some other noises with spectral components in middle frequencies. The aim of this paper is to investigate the pleasantness/unpleasantness of industrial noise with a "sound design" approach. Sixteen different industrial stimuli were created combining cooling tower noise, transformer noise and another industrial source noise, mixed with two kind of background noise ("traffic road" and "nature"). With a computer tool, twenty-two people were able to change the composition of the industrial sound (without changing the overall SPL of 40 dB(A)) and design a better sound target by improving sound quality. The first result of this study is that the average optimized stimulus is composed with more of cooling tower noise and less transformer noise. This average stimulus is "a little more pleasant" than the original sound (on a continuous scale from "much more unpleasant" to "much more pleasant"). This increase in pleasantness (with constant overall level) has been translated into an equivalent reduction of about 2.5 dB(A) of the original sound.

4:20

5pNSb9. Fitness noise impact in mixed use buildings. Walid Tikriti (Acoustonica, LLC, 33 Pond Ave., Ste. 201, Brookline, MA 02445, wtikriti@acoustonica.com)

The paper discusses noise impact from fitness activities in high rise mixed used buildings. The presented project will be discussing noise complaints from residences living above and below the fitness center. The project is located in Addison, TX. The complaints from residents are related to impact noise from fitness room class activities. Onsite sound measurements were taken and analyzed for noise source. Noise mitigation solutions were provided and implanted by the clients.

4:40

5pNSb10. A review about hearing protection comfort and its evaluation. Olivier Doutres (Mech. Eng., École de technologie supérieure, 1100 Notre Dame Ouest, Montréal, QC H3C 1K3, Canada, olivier.doutres@etsmtl.ca), Franck C. Sgard (Institut de recherche Robert-Sauvé en santé et en sécurité du travail, Montreal, QC, Canada), and Jonathan Terroir (INRS, Vandoeuvre Les Nancy, France)

Though it should ideally be the last choice in terms of noise exposure reduction, hearing protection devices (HPDs) remain the most commonly used noise control solution. However, the lack of comfort of HPDs can make it difficult for the worker to consistently and correctly wear them during work shift. It can thereby decrease their effective protection. Numerous studies have addressed the comfort of HPDs since the late fifties. These works mainly differ on (i) their definition of "comfort", (ii) how "comfort" is measured (questionnaires), (iii) their measurement conditions (laboratory versus field, naïve versus experienced wearers, type of tested HPDs…) and finally, (iv) their conclusions. The objective of this paper is to propose a comprehensive literature review of these works and to put them into perspective regarding a definition of HPD comfort based on three main components: (1) the physical one which is related to the human perception of the acoustical, biomechanical and thermal interactions between the HPD and the ear, (2) the functional one which is associated to the ergonomic aspects of the HPD and its capacity to fulfill its objectives, and (3) the psychological one which is linked to the wearer feeling in terms of acceptability, satisfaction, or habituation.

5:00

5pNSb11. A review about hearing protection comfort and its evaluation. Olivier Doutres (Mech. Eng., École de technologie supérieure, 1100 rue Notre-Dame Ouest, Montréal, QC H3C 1K3, Canada, olivier.doutres@etsmtl.ca), Franck C. Sgard (Institut de recherche Robert-Sauvé en santé et en sécurité du travail, Montreal, QC, Canada), and Jonathan Terroir (Laboratoire "Acoustique au Travail", INRS, Vandoeuvre Les Nancy, France)

Though it should ideally be the last choice in terms of noise exposure reduction, hearing protection devices (HPDs) remain the most commonly used noise control solution. However, the lack of comfort of HPDs can make it difficult for the worker to consistently and correctly wear them during work shift. It can thereby decrease their effective protection. Numerous studies have addressed the comfort of HPDs since the late fifties. These works mainly differ on (i) their definition of "comfort", (ii) how "comfort" is measured (questionnaires), (iii) their measurement conditions (laboratory versus field, naïve versus experienced wearers, type of tested HPDs…) and finally, (iv) their conclusions. The objective of this paper is to propose a comprehensive literature review of these works and to put them into perspective regarding a definition of HPD comfort based on three main components: (1) the physical one which is related to the human perception of the acoustical, biomechanical and thermal interactions between the HPD and the ear, (2) the functional one which is associated to the ergonomic aspects of the HPD and its capacity to fulfill its objectives, and (3) the psychological one which is linked to the wearer feeling in terms of acceptability, satisfaction or habituation.

Session 5pPA

Physical Acoustics: General Topics in Physical Acoustics IV

Mark S. Wochner, Chair
AdBmTechnologies, 3925 W. Braker Ln, Austin, TX 78759

Contributed Papers

1:40

5pPA1. Time-domain characterization of the nonlinear viscoelastic memory of rocks with ultrasound waves. Thomas L. Szabo (Biomedical Dept., Boston Univ., 44 Cummington Mall, Boston, MA 02215, tlszabo@bu.edu), Xuan Feng, Cai Liua (College of Geo-Exploration Sci. and Technol., Jilin Univ., Changhun, China), Stephen Brown, Daniel Burms, and Michael Fehler (Earth Resources Lab., Massachusetts Inst. of Technol., Cambridge, MA)

The nonlinear elastic response of rocks is known to be caused by internal microstructure, particularly cracks and fluids. In order to quantify this nonlinearity, we describe a new sensitive acoustic method to measure nonlinear viscoelastic properties of rocks directly in the time domain. Our method utilizes co-propagating longitudinal acoustic waves to perturb rocks with dynamically applied strain. Data indicates that rocks have a short period nonlinear viscoelastic memory that is a function of the time history of the loaded strain. We developed a phenomenological model with second and third order nonlinear elastic constants and a memory strength parameter to describe the short period nonlinear viscoelastic memory and the nonlinear elastic behavior of rock. Results of experiments on Crab Orchard sandstone and Lucite samples are presented. We have found both Lucite and Crab Orchard sandstone to be viscoelastic but the sandstone is much more nonlinear. These new observations and methods have significance for quantifying changes in microstructure and pore fluids in rocks and other materials.

2:00

5pPA2. Nonlinear behavior of different sized granular bead media on a vibrating clamped circular elastic plate. Claire E. Haas and Murray S. Korman (Phys. Dept., U.S. Naval Acad., Phys. Dept., 572 C Holloway Rd., Annapolis, MD 21402, claire.haas@hotmail.com)

This experiment uses a soil plate oscillator (SPO), or a circular acrylic (elastic) plate clamped into place using flanges, to study the nonlinear oscillatory behavior of a column of granular media (with selected diameters) that is supported by the plate. Secured on the underside of the plate's center is a rare earth magnet that is driven by an AC coil which is controlled by an amplified swept sinusoidal chirp. An accelerometer is fastened to the magnet and a spectrum analyzer measures the plate's acceleration versus frequency. In separate measurements of nonlinear tuning curve response, the plate is loaded with 350 grams of spherical soda lime beads of diameters 2, 3, 4, 5, 6, 7, and 10 mm. The driver current is changed in 40 evenly spaced increments, holding other parameters fixed, during each sweep. A study of the backbone curve (rms acceleration a(f) peak versus frequency f) for each bead diameter exhibits a frequency "softening" vs. amplitude—similar to the nonlinear mesoscopic elasticity observed in resonance experiments in geo-materials. After some initial curvature, the plot has a negative linear slope with increasing amplitude. Smaller beads show significantly more acceleration response than larger beads while the slopes have somewhat comparable values.

2:20

5pPA3. Evolution of Rayleigh streaming flow velocity components in a resonant waveguide at high acoustic levels. Virginie Daru (DynFLuid Lab/LIMSI, ENSAM/CNRS, 151, boulevard de l'hôpital, PARIS 75013, France, Virginie.Daru@ensam.eu), Hélène Bailliet (Pprime/ENSMA/ENSIP, Poitiers, France), Catherine Weisman, Diana Baltean-Carlès (Sorbonne Universités UPMC/CNRS, Paris, France), and Ida Reyt (DynFLuid Lab/LIMSI, ENSAM/CNRS, PARIS, France)

The interaction between an acoustic wave and a solid wall generates a mean steady flow called Rayleigh streaming, generally assumed to be second order in a Mach number expansion. This flow is well known in the case of a stationary plane wave at low amplitude: it has a half-wavelength spatial periodicity and the maximum axial streaming velocity is a quadratic function of the acoustic velocity amplitude at the antinode. For higher acoustic levels, additional streaming cells have been observed. In the present study, results of LDV and PIV measurements are compared to direct numerical simulations. The evolution of axial and radial velocity components for both acoustic and streaming flows is studied from low to high acoustic amplitudes. Two streaming flow regimes are pointed out, the axial streaming dependency upon acoustics going from quadratic to linear. The hypothesis of the radial streaming velocity being of second order in a Mach number expansion is shown to be invalid at high amplitudes. The change of regime occurs when the radial streaming velocity amplitude becomes larger than the radial acoustic velocity amplitude, high levels being therefore characterized by nonlinear interaction of the different velocity components.

2:40

5pPA4. Numerical and experimental investigation of the role of inertia on acoustic Rayleigh streaming in a standing waveguide. Diana Baltean-Carlès (Sorbonne Universités, UPMC Univ. Paris 6 and LIMSI, CNRS, Sorbonne Universités, UPMC Univ Paris 06, UFR d'Ingénierie, 4 Pl. Jussieu, LIMSI, CNRS, Université Paris-Saclay, Bât. 508, Rue John Von Neumann, Campus Universitaire, F-91405 Orsay Cedex, France, Paris 75252, France, baltean@limsi.fr), Virginie Daru (ENSAM-Dynfluid and LIMSI, CNRS, PARIS, France), Catherine Weisman (Sorbonne Universités, UPMC Univ. Paris 6 and LIMSI, CNRS , Paris, France), Hélène Bailliet (Institut Pprime, CNRS - Université de Poitiers - ENSMA, ENSIP, Poitiers, 86022 , Poitiers, France), and Ida Reyt (Institut Pprime, CNRS - Université de Poitiers - ENSMA, ENSIP, Poitiers, 86022 , PARIS, France)

Rayleigh streaming is a mean flow generated by the interaction between a standing wave and a solid wall. In the case of a low amplitude wave inside a cylindrical resonator, the streaming pattern along a quarter wavelength is composed of two toroidal cells: An inner cell close to the tube wall and an outer cell in the core. In the present work the effect of inertia on Rayleigh streaming at high acoustic level is investigated numerically and experimentally. To this effect, time evolutions of streaming cells in the near wall region and in the resonator core are analyzed. For the analysis of the outer cell, an analogy with the lid-driven cavity in a cylindrical geometry is proposed. It is shown that the outer cell is distorted due to convection, but the previously observed emergence of an extra cell cannot be recovered. Inertial effects on the established streaming flow pattern are further investigated

numerically by solving time averaged Navier-Stokes equations with an imposed acoustic source. Results are similar to those obtained from the lid-driven cavity simulations. Therefore inertial effects cannot be considered as responsible for the mutation of streaming at high acoustic levels.

3:00–3:20 Break

3:20

5pPA5. Simulating the energy cascade of shock wave formation process in a resonator by gas-kinetic scheme. Xiaoqing Zhang (School of Energy and Power Eng., Huazhong Univ. of Sci. and Technol., Wuhan 430074, China, zhangxqhust@sina.com), Chengwu Qu (School of Energy and Power Eng., Huazhong Univ. of Sci. and Technol., Wuhan, Hubei, China), and Heying Feng (Hunan Provincial Key Lab. of Health Maintenance for Mech. Equipment, Hunan Univ. of Sci. and Technol., Xiangtan, Hunan, China)

The temporal-spatial evolution of the sound field in a cylindrical resonator driven by a piston was simulated by gas-kinetic scheme (GKS), and periodic shock waves propagating back and forth were observed in the resonator with finite displacement amplitude of the driving piston. It can been seen from the instantaneous spatial distribution of the oscillatory pressure and velocity in the resonator that the oscillatory pressure is a one-dimensional,but the oscillatory velocity exhibits a quasi-one-dimensional distribution due to the effect of the boundary layer.The evolution process of shock wave from the sinusoidal wave to the zigzag wave was demonstrated with the increasing of the driving displacement amplitude of the piston. Especially, simulation results revealed that the acoustic intensity was linearly proportional to the driving displacement amplitude of the piston, rather than the square type. The studied results demonstrated well the process of harmonic energy cascade by the FFT analysis, at the same time, the physical explanations has been also given. The simulation results of acoustic intensity were well consistent with the existing experimental results, which verifies the validity of GKS modeling method.

3:40

5pPA6. Comparison between the Gol'dberg number and the Morfey-Howell *Q/S* for finite-amplitude plane waves. Won-Suk Ohm (School of Mech. Eng., Yonsei Univ., 50 Yonsei-ro, Seodaemun-gu, Seoul 120-749, South Korea, ohm@yonsei.ac.kr), Kent L. Gee, and Brent O. Reichman (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT)

In this paper we compare the two most popular indicators of nonlinearity, namely, the Gol'dberg number [Sov. Phys. Acoust. 2, 346-350 (1956)] and the Morfey-Howell *Q/S* [AIAA J. 19, 986-992 (1981)] for the case of an initially-sinusoidal, finite-amplitude plane wave in a thermoviscous medium. The Gol'dberg number Γ, defined as the ratio of the absorption length at the fundamental frequency to the shock formation distance, is a parameter that single-handedly dictates the entire history of the wave [for example, weak ($\Gamma < 1$) and strong ($\Gamma >> 1$) waves]. However, it fails to quantify the strength of nonlinearity that each individual harmonic "sees" at any given point. In other words, not every harmonic sees the apparent strength of nonlinearity according to Γ. The *Q/S*, on the other hand, is defined for each harmonic as a function of propagation distance, and hence keeps track of the ever-changing strength of nonlinearity during the course of propagation. One interesting ramification of this is that using *Q/S* it is possible to pinpoint the distance at which the value of nonlinearity relative to absorption actually becomes Γ for the fundamental component.

4:00

5pPA7. Ultraviolet laser filaments for underwater acoustic generation using shaped plasmas. Theodore G. Jones, Richard Fischer, Dmitri Kaganovich, Michael Helle, Bahman Hafizi, John Palastro (Plasma Phys., U.S. Naval Res. Lab., NRL Code 6795, 4555 Overlook Ave SW, Washington, DC 20375, ted.jones@nrl.navy.mil), Yu-Hsin Chen (Res. Support Instruments, Inc., Washington, District of Columbia), and Alexander Stamm (NRC-NRL Res. Associate, Washington, DC)

NRL is developing underwater laser acoustic generation techniques using nonlinear optics to form shaped plasmas. One or more laser pulses are focused to ionization intensity in water, where enhanced absorption causes plasma generation and intense localized heating. The size and shape of the resulting super-heated steam "shaped charge" determines the acoustic waveform, energy spectral density (ESD), and directivity. Using a single laser pulse we have demonstrated the ability to vary the ESD centroid frequency from 15 kHz to a few MHz, with lower frequencies generated by more elongated plasmas. Recent research has focused on a proposed "igniter-heater" technique utilizing two laser pulses to generate meter-scale elongated plasma (T.G. Jones, *et al.*, "Two Laser Generation of Extended Underwater Plasma," U.S. patent 9,088,123), with ESD centered near 1 kHz, which is useful for long-range sonar and communications. Key to this technique is generation of an underwater plasma filament serving as an ionized "igniter" target for a subsequent "heater" laser beam. Experiments use ultraviolet picosecond pulses, which can exploit weakly-bound "hydrated" free-electron states with high stored energy. Computational modeling includes nonlinear optical propagation and ionization. We will discuss recent experimental and modeling results. [This work was supported by NRL Base Funds.]

Session 5pPPa

Psychological and Physiological Acoustics: Psychoacoustics: Models and Perception

Christopher J. Plack, Chair
School of Health Sciences, University of Manchester, Ellen Wilkinson Building, Oxford Road, Manchester M13 9PL, United Kingdom

Contributed Papers

1:20

5pPPa1. Analysis of different modeling approaches of the cochlea impedance in FE models of the human middle ear. Lucas C. Lobato (Mech. Eng., Federal Univ. of Santa Catarina, Av. Roraima n° 1000, Santa Maria, Rio Grande do Sul 97105-900, Brazil, lucascostalobato@gmail.com), Stephan Paul (Mech. Eng., Federal Univ. of Santa Catarina, Joinville, Brazil), and Julio A. Cordioli (Mech. Eng., Federal Univ. of Santa Catarina, Florianópolis, SC, Brazil)

Finite element models of the human middle ear are important tools in the development of implantable hearing aids or to study the impact of surgeries on middle ear dynamics. Thus, the cochlea impedance seen by the middle ear is a very important parameter when modeling the middle ear dynamic behavior. The cochlea impedance can be seen as the load effect provide by cochlea fluid over the stapes footplate. Although several papers describe human middle ear FE models, there is no consensus about how to better include the cochlea impedance at FE model. In this work, a review of the different modeling approaches found in the literature is presented. Each approach was implemented, and the results for the middle ear frequency response (stapes velocity versus tympanic membrane pressure) were compared. The results are also compared with experimental data for the middle ear frequency response. An alternative model is also proposed that includes measured data for the cochlea impedance. From the results, it was observed significantly difference between methods in frequencies above 900 [Hz]. It was concluded that using the experimental data, it is added more damping at high frequencies.

1:40

5pPPa2. Objective psychoacoustics. Monika Gatt, Marcus Guettler (Faculty of Mech. Eng., Tech. Univ. of Munich, Boltzmannstr. 15, Munich 85748, Germany, monika.gatt@tum.de), and Steffen Marburg (Faculty of Mech. Eng., Tech. Univ. of Munich, Muenchen, Germany)

An intuitive psychological understanding of sound is given by nature. Being aware of the meaning of sound often requires an identifying process of an ethical and physical explanation (propter hoc and post hoc). For an objective, scientific and ethical result, one has to investigate sound in the frames of psychoacoustic parameters. But objective psychoacoustics has to fulfill psychological criteria, discuss aesthetic evaluation as well as reflect physical sources. The tension arises between the subjective listeners, whose decisions of participation in the test are based on free will, and the researchers in acoustics, whose experiment setup is orientated to achieve independent results. Both parties are driven by empirical and metaphysical pragmatism and the awareness of the difference between subjectivity and objectivity. The discussion is held with respect to the gender, profession, age (childhood and elderly people) and nationality. With this information, the question arises how to get from an individual point of view to an overall perspective. For this purpose, the chair of Vibroacoustics of Vehicles and Machines of the Technical University of Munich is in the process of developing an open source psychoacoustic toolbox. The results will be analyzed in a systematic and objective method.

2:00

5pPPa3. Application of Wald sequential test in staircase up-down adaptive procedures. Jan Zera (Inst. of Radioelectronics and Multimedia Technol., Faculty of Electronics and Information Technol., Warsaw Inst. of Technol., Nowowiejska, 15/19, Warsaw 00-665, Poland, j.zera@ire.pw.edu.pl)

An advantage of adaptive, staircase, up-down procedures in sensory threshold measurements is that such procedures are based on simple rules for signal level setting and are relatively robust to fluctuations of the subject's attention during the measurement. A disadvantage of those procedures is that they allow to estimate the threshold level only for a few points on the psychometric function. In this study an adaptive, up-down staircase procedure is proposed which uses a decision rule based on the Wald sequential statistical test, similar to that applied in the PEST adaptive method. The Wald test allows to determine the signal threshold level for any point on the psychometric function and is simple to implement. Numerical simulations and results of an experiment performed on real subjects have shown that the staircase procedure combined with a Wald test for signal level setting reproduces the tracks of standard up-down procedures and may estimate the sensory threshold with similar accuracy. The modified procedure is thus plausible for various kind of threshold measurements in psychoacoustic studies.

2:20

5pPPa4. Modeling listening performance with decision forests. Frank S. Mobley (Human Effectiveness Directorate, U.S. Air Force, 2610 Seventh St., Bldg. 441, Wright-Patterson AFB, OH 45433, frank.mobley.1@us.af.mil)

Human audibility measurements of rotary-wing aircraft in a series of real auditory ambient environments is compared with output from a random decision forest machine learning algorithm. A model of human audition in different acoustic environments is built on an error function, but the coefficients of the error function must be determined with detailed human subject data. A random decision forest is constructed from a large set of existing human detection data to test the algorithm's ability to generate human surrogate data. The forest algorithm is shown to reproduce predictions of the human response for the test ambient data in all cases with R^2 of 0.9 or greater. It produces results at a rate of 6 to 20 minutes per prediction, depending upon the number of calculations, which is significantly faster than collecting results using human testing.

2:40–3:00 Break

3:00

5pPPa5. On the existence of sex, menstrual-cycle, and race differences in psychoacoustical performance. Dennis McFadden, Edward G. Pasanen, Mindy M. Maloney (Psych., Univ. of Texas, Austin, 108 E. Dean Keeton A8000, Austin, TX 78712-1043, mcfadden@psy.utexas.edu), Erin M. Leshikar, and Michelle H. Pho (Commun. Sci. and Disord., Univ. of Texas, Austin, Austin, TX)

Performance was measured on several common psychoacoustical tasks for about 70 subjects. Subjects were tested in same-sex crews of 4—8 members. Testing required 8—10 weeks for each crew to assure that all subjects were well practiced on all tasks. Of initial interest were any differences by sex and by menstrual cycle. Correlation and effect size were the primary measures of interest. Resampling was used to determine implied significance for the various comparisons studied. During analysis, racial background turned out to be a relevant variable; because this was unexpected, recruitment by, and documentation of, race was poor, so the best we could do was categorize subjects as Non-White or White. For essentially all of the psychoacoustical tasks measured, the sex and menstrual differences were non-significant when the results were pooled across race. However, when the subjects were partitioned by race, small-to-moderate sex and menstrual differences did emerge, primarily for the White subjects. Correlations between certain behavioral tasks were moderately high, and this also varied somewhat by race. Past studies reporting sex or menstrual differences likely consisted primarily of White subjects. Multiple lines of evidence suggest that these race differences are attributable to the intracochlear concentration of melanin.

3:20

5pPPa6. The perception of musical consonance beyond the traditional existence region of pitch. Samuele Carcagno, Saday Lakhani (Dept. of Psych., Lancaster Univ., Lancaster, United Kingdom), and Christopher J. Plack (Dept. of Psych., Lancaster Univ., Ellen Wilkinson Bldg., Oxford Rd., Manchester M13 9PL, United Kingdom, chris.plack@manchester.ac.uk)

The traditional notion that pitch perception breaks down for frequency components above 5 kHz was challenged by a study [Oxenham *et al.*, Proc. Natl. Acad. Sci. U. S. A. 108, 7629-7634 (2011)] showing that complex tones high-pass filtered at 6 kHz convey the perception of melodic intervals. We investigated whether complex tones high-pass filtered above 6 kHz can also convey the perception of harmonic intervals. Eleven listeners rated the pleasantness of consonant and dissonant dyads (two-note chords) consisting of harmonic complex tones band-pass filtered either in a low-frequency (1-6 kHz) or in a high-frequency (7-12 kHz) region. The two tones in each dyad were presented to opposite ears. In the high-frequency condition the dyads were presented with a low-pass noise to mask combination tones. Pleasantness ratings were significantly higher for consonant than for dissonant intervals in both frequency regions. This preference for consonant dyads in the high-frequency region was not observed when the tones were made inharmonic by shifting all their components by a fixed offset so as to preserve envelope periodicity cues. The results indicate that the perception of consonance is preserved for tones composed entirely of components above the traditional existence region of pitch.

3:40

5pPPa7. Amplitude envelope reconstruction based on signal periodicity. Volker Dellwo (Phonet. Lab., Univ. of Zurich, Plattenstrasse 54, Phonet. Lab., Zurich 8005, Switzerland, volker.dellwo@uzh.ch), Thayabaran Kathiresan, and Lei He (Phonet. Lab., Univ. of Zurich, Zurich, Zurich, Switzerland)

Speech can be modeled as the sum of high frequency carrier signals (temporal fine structure; TFS) from different frequency bands modulated in amplitude by a low frequency modulator (amplitude envelope; ENV). Previous research revealed that ENV cues alone can be sufficient for speech intelligibility. The present research tested the role of signal periodicity to reconstruct ENV information. Signal periodicity was measured by harmonics-to-noise ratio (HNR) with HNR = 10*log10(signal/noise) in 60 msec windows at 10 msec steps in each of 6 frequency bands (Greenwood spacing) of different speech signals (bw = 8 kHz). Resulting HNR contours were correlated with respective ENVs in each frequency band and high correlations ($r>0.8$) could be obtained. This was also true when ENV was removed in each band (multiplying the signal with the inverse Hilbert ENV) prior to processing of the HNR contour. HNR contours were used to amplitude modulate noise (noise vocoding). The HNR vocoded speech revealed to be equally intelligible as ENV vocoded speech. Results give rise to the view that speech periodicity—which is extremely robust towards linear system distortions—might lend itself for recovering ENV cues in the auditory processing of speech.

4:00

5pPPa8. Relative psychoacoustic analysis for sound design purpose. Samira Mohamady (Hainbuchenstrasse 8, Aachen, NRW 52074, Germany, samira.mohamady.82@gmail.com)

In sound design, besides psychoacoustics analysis of acoustic system, the expected deviation of the sound metrics parameters due to imperfect modeling and measurement of the system play an important role. In this study, a new sight is given to calculate the expected deviation in the perception of sound in acoustic system with the motoring the possible deviation of the system under test. The results are discussed to cover the most common problems in sound design process.

4:20

5pPPa9. On the capabilities of the psychoacoustic Matlab toolbox ASAPP in comparison with commercial algorithms. Magdalena Scholz (Chair of VibroAcoust. of Vehicles and Machines, Tech. Univ. of Munich, Boltzmannstr. 15, Garching 85748, Germany, magdalena.scholz_tum@web.de), Monika Gatt, Marcus Guettler (Chair of VibroAcoust. of Vehicles and Machines, Tech. Univ. of Munich, Munich, Germany), and Steffen Marburg (Chair of VibroAcoust. of Vehicles and Machines, Tech. Univ. of Munich, Muenchen, Germany)

Vibrating surfaces generate sound. Every moving part of a vehicle or a device can create sound. This sound influences the user's perception of this device. Hence, sound design is increasingly prevailing. An analysis of sound with regard to its perception, i.e., to psychoacoustic parameters, is not only needed for the optimization of the user's experience, but also to analyze systems. An experienced mechanic can detect a defect in a vehicle by listening to its sound, which is hardly executable by merely analyzing the frequency spectrum and the sound pressure level. Therefore, the authors use psychometric parameters to emulate the analysis done by the mechanic in the previous example. For such kind of analysis, the open source Matlab toolbox ASAPP—Audio Signal Analysis and Psychoacoustic Parameter was developed. To evaluate the capabilities of ASAPP, the authors compare results from the toolbox with results obtained from commercial software by means of analyzing different sound samples. In addition, different models for the psychoacoustic parameters are compared with respect to the type of signals investigated.

Session 5pPPb

Psychological and Physiological Acoustics, Speech Communication, ASA Committee on Standards, Architectural Acoustics, and Signal Processing in Acoustics: Speech Intelligibility in Adverse Environments: Behavior and Modeling II

Mathieu Lavandier, Cochair
ENTPE/LGCB, Univ. Lyon, Rue M. Audin, Vaulx-en-Velin 69518, France

Virginia Best, Cochair
Dept. Speech, Language and Hearing Sciences, Boston University, 635 Commonwealth Ave., Boston, MA 02215

Invited Papers

1:20

5pPPb1. Normative data for assessing performance on a rapid, automated test of speech-on-speech masking and spatial release from masking. Frederick J. Gallun (National Ctr. for Rehabilitative Auditory Res., VA Portland Health Care System, 3710 SW US Veterans Hospital Rd., Portland, OR 97239, Frederick.Gallun@va.gov), Kasey Jakien (Dept. of Otolaryngology/Head and Neck Surgery, Oregon Health and Sci. Univ., Portland, OR), Nirmal Srinivasan (Towson Univ., Baltimore, MD), Aaron Seitz (Dept. of Psych., Univ. of California, Riverside, Riverside, CA), Sean Kampel, and Meghan Stansell (National Ctr. for Rehabilitative Auditory Res., VA Portland Health Care System, Portland, OR)

The ability to understand speech in the presence of competing sound sources is essential to communication success. This task is difficult for listeners with healthy auditory systems, but becomes substantially more difficult as listeners age and experience various insults to the system. Our laboratory has developed a rapid, automated method of measuring the relative ability to perform a standard laboratory version of this task, which has been applied to a large set of spatial separations between 0 and 135 degrees, and a range of levels and bandwidths. Performance has been obtained in over 100 listeners varying in age and hearing ability between one and fifteen times over the course of four years. We have examined both the reliability of this test, by comparing how performance changed across test sessions with the same acoustical conditions (less than 1 dB on average), and the relative influences of age and hearing loss at various separations. Normative functions based on over 100 listeners are available for a colocated control condition, a spatial separation of 45 degrees, and the difference between the two (spatial release from masking). Normative functions take into account both age, between 18 and 80 years, and moderate hearing loss in the standard audiometric frequencies. A calibrated version of the test is freely available for use on an iPad.

1:40

5pPPb2. Sequential streaming of speech sounds under normal and impaired hearing. Marion David (Psych., Univ. of Minnesota, 75 East River Parkway, Minneapolis, MN 55455, david602@umn.edu), Olaf Strelcyk (Sonova U.S. Corporate Services, Warrenville, IL), and Andrew J. Oxenham (Psych., Univ. of Minnesota, Minneapolis, MN)

Segregating and understanding speech within complex auditory scenes remains a major challenge for hearing-impaired (HI) listeners. This study compared the ability of normal-hearing (NH) listeners and HI listeners with mild-to-moderate loss to segregate sequences of speech tokens, consisting of an unvoiced fricative consonant and a vowel (CV), based on a difference in fundamental frequency (F0) and/or vocal tract length (VTL). The CVs were amplified and spectrally shaped to ensure audibility for the HI listeners. In the streaming task, the CV tokens were concatenated into sequences that alternated in F0, VTL or both. The resulting interleaved sequences were preceded by a "word" consisting of two random syllables. The listeners were asked to indicate whether the word (which varied from trial to trial) was present in the interleaved sequence. The word, if present, occurred either within one sequence or across the alternating sequences. Preliminary results showed no difference in performance between the two groups, suggesting that the listeners with mild-to-moderate sensorineural hearing loss are able to use differences in F0 and VTL to segregate speech sounds in situations where there is no temporal overlap between the competing sounds.

2:00

5pPPb3. Assessing the effects of hearing loss on speech intelligibility in reverberant environments. Jayaganesh Swaminathan, Jing Xia (Starkey Hearing Technologies, 2250 Shattuck Ave., #405, Berkeley, CA 94704, jayaganesh_swaminathan@starkey.com), Buye Xu (Starkey Hearing Technologies, Eden Prairie, Minnesota), and Shareka Pentony (Starkey Hearing Technologies, Berkeley, CA)

It has been shown that listeners with sensorineural hearing loss (SNHL) have difficulties in understanding speech in simulated reverberant environments even when wearing assistive devices such as hearing aids. However, a number of limiting factors such as poor hearing loss compensation, over simplified acoustic simulations of reverberant environments and dated statistical analysis have led to a lack

of a clear understanding of the specific factors that cause hearing-impaired listeners to have difficulties in reverberant environments. In this talk, we will present results from studies comparing normal-hearing and hearing-impaired listeners: 1) Monaural and binaural modulation detection thresholds measured with anechoic and reverberant stimuli with different carriers and 2) Speech intelligibility in typical reverberant and noisy listening environments exploring a range of acoustic parameters, such as the level of reverberation, the type of noise and various signal-to-noise ratios. Furthermore, the effect of SNHL and reverberation on the neural coding of envelope and temporal fine structure will be assessed using a physiology based model of the auditory-nerve.

2:20

5pPPb4. Reverberation limits the release from informational masking obtained by differences in fundamental frequency and in spatial location. Mickael L. Deroche (Ctr. for Res. on Brain, Lang., and Music, McGill Univ., Rabinovitch House, 3640 rue de la Montagne, Montreal, QC H3G 2A8, Canada, mickael.deroche@mcgill.ca), John F. Culling (School of Psych., Cardiff Univ., Cardiff, Wales, United Kingdom), Mathieu Lavandier (ENTPE Laboratoire Genie Civil and Batiment, Universite de Lyon, Vaulx-en-Velin, France), and Vincent Gracco (Ctr. for Res. on Brain, Lang., and Music, McGill Univ., Montreal, QC, Canada)

Differences in fundamental frequency (ΔF0s) and differences in spatial location (ΔSLs) between competing talkers can substantially enhance intelligibility of a target voice in a typical cocktail-party situation. Reverberation is generally detrimental to the use of these two cues, but it is possible to create laboratory conditions where reverberation should not disrupt the release from energetic masking produced by ΔF0s and ΔSLs. Two masker types were used: a 2-voice speech masker and a non-linguistic masker (primarily energetic) matched in long-term excitation pattern and broadband temporal envelope to speech maskers. Speech reception thresholds were measured either in an adaptive procedure with unpredictable sentences or with the coordinate response measure at fixed target-to-masker ratios, in conditions with or without ΔF0s, and with or without ΔSLs, against the two masker types in anechoic and reverberant conditions. Both methods provided a similar pattern of results. In the presence of non-linguistic maskers, ΔF0s and ΔSLs provided masking releases which, as intended, were robust to reverberation. Larger masking releases were obtained for speech maskers, presumably due to the additional informational component, but critically, they were reduced by reverberation. Several interpretations will be discussed at the meeting.

2:40

5pPPb5. Better-ear glimpsing for fully and partially degraded speech signals. Matthew Goupell (Hearing and Speech Sci., Univ. of Maryland, College Park, 0119E Lefrak Hall, University of Maryland, College Park, College Park, MD 20742, goupell@umd.edu), Olga Stakhovskaya (Hearing and Speech Sci., Univ. of Maryland, College Park, Bethesda, MD), Joshua G. Bernstein, and Douglas Brungart (Audiol. and Speech Ctr., Walter Reed National Military Medical Ctr., Bethesda, MD)

Normal-hearing listeners seem to take advantage of "better-ear glimpsing" when a target signal is masked by interfering sounds with different fluctuations in the two ears, presumably by analyzing small spectro-temporal bins and synthesizing the bins from the ear with the best signal-to-noise ratio (SNR). This study investigated the glimpsing mechanism using degraded inputs associated with bilateral cochlear implants (BICIs). BICI and NH listeners identified three-digit sequences spoken by a target talker at 0° with interfering talkers that were co-located at 0° or placed at ±60°. Glimpsing was evaluated by comparing the co-located control condition with either a standard bilateral condition (where the better-ear glimpses alternated across the ears) or a better-ear processed condition (where all the better-ear glimpses were artificially moved to one ear). Both NH and BICI listeners obtained substantial spatial benefit in the better-ear processed condition. However, the spatial benefit in the standard bilateral condition was greatly reduced for the BICI listeners and for NH listeners listening to vocoded signals. Varying the number of vocoder channels from 4 to 128 had little effect on the spatial release. The results provide insights into the cues listeners use to extract speech from fluctuating maskers. [Work supported by NIH R01-DC014948 (Goupell).]

3:00–3:20 Break

3:20

5pPPb6. Having two ears can facilitate or interfere with the perceptual separation of concurrent talkers for bilateral and single-sided deafness cochlear-implant listeners. Joshua G. Bernstein (National Military Audiol. and Speech Pathol. Ctr., Walter Reed National Military Medical Ctr., 4954 N. Palmer Rd., Bethesda, MD 20889, joshua.g.bernstein.civ@mail.mil), Matthew Goupell, Jessica Wess (Dept. of Hearing and Speech Sci., Univ. of Maryland - College Park, College Park, MD), Olga Stakhovskaya, and Douglas S. Brungart (National Military Audiol. and Speech Pathol. Ctr., Walter Reed National Military Medical Ctr., Bethesda, MD)

Binaural hearing provides normal-hearing listeners with tremendous speech-understanding benefits in complex listening conditions. While head-shadow advantages improve speech understanding in noise for bilateral (BI) and single-sided deafness (SSD) cochlear-implant (CI) listeners, it is unclear whether bilateral input can facilitate the perceptual organization of a complex auditory scene. We review a series of studies examining speech perception in the presence of interfering talkers for these listeners. For some BI-CI listeners and for SSD-CI listeners attending to their normal-hearing ear, having two ears facilitates the perceptual separation of concurrent talkers. For other BI-CI listeners and for SSD-CI listeners attending to their CI ear, bilateral input can produce speech-perception interference. Factor analyses and vocoder simulations suggest that the presence of binaural benefit or interference can depend on hearing history (duration of deafness) or physical factors (interaural spectral mismatch). To maximize binaural-hearing benefits and reduce the likelihood of interference, CI candidates should be implanted soon after deafness onset, with frequency-allocation tables optimized to minimize interaural mismatch. [Support: NIH-NIDCD/R01-DC-015798 (Bernstein/Goupell), DMRDP/DM130007 (Bernstein) and NIH-NIDCD/R01-DC-014948 (Goupell). The views expressed in this abstract are those of the authors and do not reflect the official policy of the Department of Army/Navy/Air Force, Department of Defense, or U.S. Government.]

3:40

5pPPb7. Talker variability in real-life speech recognition by cochlear implant users. Terrin N. Tamati (Dept. of Otorhinolaryngology / Head and Neck Surgery, Univ. Medical Ctr. Groningen, Hanzeplein 1, Groningen 9700RB, Netherlands, t.n.tamati@umcg.nl), Esther Janse (Ctr. for Lang. Studies, Radboud Univ. Nijmegen, Nijmegen, Netherlands), David B. Pisoni (Dept. of Psychol. and Brain Sci., Indiana Univ., Bloomington, IN), and Deniz Baskent (Dept. of Otorhinolaryngology / Head and Neck Surgery, Univ. Medical Ctr. Groningen, Groningen, Netherlands)

For adult cochlear implant (CI) users, understanding speech in the real world, outside the clinic, can be challenging. A fundamental attribute of real-life speech is the immense amount of talker variability, present from multiple talkers with diverse linguistic and developmental histories. Talker variability, beyond background noise or competing talkers, contributes greatly to real-life, adverse listening conditions. Normal-hearing listeners are able to quickly and efficiently adapt to and learn talker differences to facilitate speech recognition in these conditions. However, talker variability may be particularly detrimental to CI users, since they must rely on signals that are inherently degraded, missing spectro-temporal details. In three experiments, we examined the impact of different sources of talker variability (talkers, accents, speaking styles) on CI speech recognition. The results showed that CI users have difficulty recognizing speech with vast talker variability, and discriminating talkers' voices and accents, with a great amount of individual differences in performance. These findings suggest that the well-controlled, low-variability conditions in the lab or clinic may not reflect actual CI users' performance in daily life. [Funding: VIDI Grant (No. 016.093.397) from the Netherlands Organization for Scientific Research (NWO), the Netherlands Organization for Health Research and Development (ZonMw).]

Contributed Papers

4:00

5pPPb8. Absence of ear advantage for segmental and tonal perception in noise. Feng Gu, Themis Kwong, and Lena Wong (Div. of Speech and Hearing Sci., Univ. of Hong Kong, Rm770, Meng Wah complex, Pokfulam, Hong Kong 000000, Hong Kong, gufeng@hku.hk)

When different speech sounds are presented to the left and right ear simultaneously, right-handed listeners usually score higher for identifying stimuli presented to the right ear, reflecting the left hemisphere dominance for speech perception. Recently, the right hemisphere was proposed to contribute more than the left hemisphere for speech perception in an acoustically noisy environment. For testing this hypothesis, 40 right handed native Cantonese speakers were recruited for dichotic listening tests. Paired syllables were presented to participants in quiet and in noise (signal to noise ratio were set to 0 dB and -10 dB in two separate conditions). The two syllables in each pair differed from each other in consonant, vowels, or tones. Participants were asked to verbally repeat the syllable they thought was clearer. Accuracy rate was calculated for each ear. For the quiet condition, the mean accuracy rate was significantly higher for syllables presented to the right ear than those presented to the left ear for consonants, vowels, and tones. Whereas for both noise conditions, no significant ear advantage was found. The absence of ear advantage in noise suggested that the right hemisphere contributes more in a noisy environment than in quiet for both segmental and tonal perception.

4:20

5pPPb9. Dynamic control over the allocation of listening effort in speech perception. Matthew Winn (Speech & Hearing Sci., Univ. of Washington, 1417 NE 42nd St., Seattle, WA 98105, mwinn83@gmail.com)

Unlike typical laboratory and clinical testing situations, everyday speech communication involves hearing multiple consecutive utterances. Previous work has shown that effort allocated to perceiving degraded speech remains elevated until well after sentence offset, presumably to allow cognitive restoration of words that were missed, via contextual cues. This restorative process potentially conflicts with the perception of subsequent sentences. In this study we introduce various follow-up signals after a target sentence with the intent of interacting with the restorative processes used to recover intelligibility in difficult listening conditions. Pupillometry was used to gauge listening effort for a sentence followed by a critical 2-second window filled by silence, noise, or a random sequence of three digits; sentences were always repeated, and digits were repeated or ignored, in blocked sessions. Speech followed by noise elicited a pattern of effort that paused until the noise was over, and then resumed. Digits elicited a modest elevation in effort when they were ignored, but a considerable elevation when repeated along with the sentence. Notably, effort allocated to the sentence appears to be suppressed when the listener plans to attend to upcoming signals. Results suggest that responses to single utterances might not reflect dynamic cognitive demands in communication.

4:40

5pPPb10. Reproducibility of speech intelligibility scores in spatially reconstructed and re-reconstructed vehicle cabin noise. Ewan A. Macpherson (National Ctr. for Audiol., Western Univ., 1201 Western Rd, Elborn College 2262, London, ON N6G 1H1, Canada, ewan.macpherson@nca.uwo.ca)

We assessed the fidelity of the multichannel freefield soundfield reconstruction technique proposed by Minnaar *et al* (AES 2013) by comparing Hearing in Noise Test (HINT) speech reception thresholds (SRTs) obtained in reconstructed, and then re-reconstructed, vehicle cabin noise soundfields. Recordings were first made in a moving vehicle under a variety of driving conditions (50 km/hr, smooth/rough road, open/closed windows) using a 32-channel spherical microphone array (Eigenmike ®, m.h. acoustics) placed in the passenger's head location. Using a six-loudspeaker array in the same stationary vehicle, the soundfields were approximately reconstructed, and SRTs were obtained for 12 normally hearing listeners located in the passenger seat; a small loudspeaker reproduced target speech from the driver's position and two rear passenger positions. For comparison, the in-vehicle reconstructed soundfields (R1) and speech were re-recorded and then re-reconstructed (R2) in an anechoic chamber using a 48-loudspeaker array, where SRTs were obtained for the same listeners. Mean SRTs were uniformly lower in R2 by 0.32 to 1.87 dB depending on speaker location and driving condition. The differences were larger when the speaker was beside rather than behind the listener by a mean of 0.98 dB and larger when windows were open by a mean of 0.23 dB. The results suggest that the reconstruction technique requires minor refinement in order to yield speech intelligibility scores equivalent to those in the original environment.

5:00

5pPPb11. The effect of F0 contours on the intelligibility of speech against interfering sounds for Mandarin Chinese. Jing Chen, Hongying Yang, and Xihong Wu (Dept. of Machine Intelligence, Speech and Hearing Res. Ctr., and Key Lab. of Machine Percept. (Ministry of Education), Peking Univ., Rm. 2228, No. 2 Sci. Bldg., Ctr. for information Sci., Beijing 100871, China, chenj@cis.pku.edu.cn)

To study the role of F0 contour on speech intelligibility for Mandarin Chinese, speech reception thresholds (SRTs) of mono-tone speech and multi-tone speech were measured both in steady speech-spectrum noise (SSN) and in two-talkers' speech (TTS). The effect of informational masking (IM) in TTS masking was also examined. The mono-tone speech was produced by mono-tone sentences, which only consisted of syllables with tone 1 or tone 0. All sentences used in this study were grammatically correct but semantically anomalous to avoid the effect of sentence context. The type of TTS maskers was also manipulated as mono-tone or multi-tone. The results revealed: 1) For SSN masking, there was no significant difference between the mono-tone speech and the multi-tone speech (-7.8 dB vs. -7.3

dB on SRTs), indicating a weak effect of altering F0 contour at the sentence level. 2) However, for TTS masking, the intelligibility of multi-tone speech in multi-tone masking was significantly difficult (2 dB increase for SRT) than other conditions, indicating an important role of F0 contour in IM. These results were different from the corresponding reports for English speech, and the characteristics of F0 contour and language were analyzed and discussed. [supported by a NSFC grant 61473008.]

THURSDAY AFTERNOON, 29 JUNE 2017 ROOM 201, 1:20 P.M. TO 5:20 P.M.

Session 5pSA

Structural Acoustics and Vibration and Physical Acoustics: Numerical Methods and Benchmarking in Computational Acoustics II

Robert M. Koch, Cochair
Chief Technology Office, Naval Undersea Warfare Center, Code 1176 Howell Street, Bldg. 1346/4, Code 01CTO, Newport, RI 02841-1708

Micah R. Shepherd, Cochair
Applied Research Lab, Penn State University, PO Box 30, mailstop 3220B, State College, PA 16801

Manfred Kaltenbacher, Cochair
Mechanics and Mechatronics, TU Wien, Getreidemarkt 9, Wien 1060, Austria

Steffen Marburg, Cochair
Faculty of Mechanical Engineering, Technical University of Munich, Boltzmannstr. 15, Muenchen 85748, Germany

Invited Paper

1:20

5pSA1. Adaptive interpolation techniques for structural acoustic system response. Andrew S. Wixom (Appl. Res. Lab., Penn State Univ., P.O Box 30, M.S. 3220B, State College, PA 16801, axw274@psu.edu) and James G. McDaniel (Mech. Eng., Boston Univ., Boston, MA)

In recent years, several new techniques for interpolating the response of structural acoustic systems have been developed. Besides the various underlying approximation schemes that utilize concepts such as Padé approximants or implicitly interpolatory subspaces, the main advantage of these methods is that their structure is adaptively determined, meaning that they select interpolation points and, when appropriate, interpolation order as the algorithm proceeds. This allows users to have little to no *a priori* knowledge of the system response and yet still achieve approximations of a desired accuracy. Furthermore, the interpolation is generated at minimal cost giving rise to approximations that are efficient as well as accurate. A variety of these techniques are introduced and discussed with emphasis on the trade-offs between the methods. Then, a numerical test bed is developed to evaluate the performance of the interpolation schemes giving each access to the same computational resources. Recommendations are made based on these simulations concerning the appropriate choice of adaptive interpolation for various scenarios.

Contributed Papers

1:40

5pSA2. Comparison of the component-wise and projection-based Padé approximant methods for acoustic coupled problems. Romain Rumpler and Peter Göransson (KTH Royal Inst. of Technol., Teknikringen 8, Stockholm 10044, Sweden, rumpler@kth.se)

Several Padé-based computational methods have been recently combined with the finite element method for the efficient solution of complex time-harmonic acoustic problems. Among these, the component-wise approach, which focuses on the fast-frequency sweep of individual degrees of freedom in the problem, is an alternative to the projection-based approaches. While the former approach allows for piecewise analytical expressions of the solution for targeted degrees of freedom, the projection-based approaches may offer a wider range of convergence. In this work, the two approaches are compared for a range of problems varying in complexity, size and physics. This includes for instance the modeling of coupled problems with non-trivial frequency dependence such as for the modeling of sound absorbing porous materials. Conclusions will be drawn in terms of computational time, accuracy, memory allocation, implementation, and suitability of the methods for specific problems of interest.

2:00

5pSA3. A general substructuring synthesis method for the analysis of complex vibro-acoustic interaction systems. Zhenguo Zhang and Julien Meaud (G. W. Woodruff School of Mech. Eng., Georgia Inst. of Technol., 771 Ferst Dr., RM 135, Atlanta, GA 30332, zzgjtx@gmail.com)

A general substructuring synthesis method is proposed for analyzing structural-acoustic interaction problems in engineering applications. The approach is performed by systematically partitioning the entire system into a series of distinct components according to geometries and boundaries. The motion of substructures and fluid subdomains is then discretized and represented by the classical Rayleigh-Ritz method, while the perturbed Lagrangian method is employed to connect the disjoint substructures and subdomains by imposing approximate compatibility conditions. The established framework enables one to easily model the interior acoustics of combined fluid structural systems with geometric complexities and to effectively analyze vibro-acoustic behaviors with sufficient accuracy at relatively high frequencies. Several numerical examples are carried out to demonstrate the reliability and applicability of the present method, which illustrate a significant computational advantage as compared to classical Finite Element procedures.

2:20

5pSA4. Characteristic amplitude-frequency functions of the radiated sound power. Matthias Klaerner, Mario Wuehrl, Lothar Kroll (Chemnitz Univ. of Technol., Reichenhainer Str. 70, Chemnitz 09126, Germany, matthias.klaerner@mb.tu-chemnitz.de), and Steffen Marburg (Tech. Univ. Munich, Muenchen, Germany)

The simulation-based design of dynamically loaded and acoustically sensitive components essentially includes the determination of the radiated sound power. Regarding several simplifications different approaches based on the surface velocity can be applied, e.g. the equivalent radiated sound power (ERP). The required frequency steps of steady state dynamic simulations are determined using efficient modal super imposed models. For single modes of rectangular plates, universal amplitude-frequency functions shall be identified. Considering the damping ratio of the mode, only one resonant frequency step is further needed for the estimation of the radiated sound power in the whole given frequency range. Computationally expensive steady state simulations thus are significantly reduced.

2:40

5pSA5. A formulation for elastodynamic scattering based on boundary algebraic equations. Jordi Poblet-Puig (DC - LaCàN, Universitat Politècnica de Catalunya, C/Jordi Girona 1-3, campus Nord, B1-206, Barcelona, Catalunya E-08034, Spain, jordi.poblet@upc.edu) and Andrey V. Shanin (Dept. of Phys., Acoust. Div., Moscow State Univ., Moscow, Russian Federation)

The solution of the elastodynamic problem arises in many scientific fields such as the wave propagation in the ground, non-destructive testing, the vibration design of buildings or vibroacoustics in general. We present here an integral formulation based on the boundary algebraic equations. It leads to a boundary numerical method and has the important advantage that no contour (2D) or surface (3D) integrals need to be computed. This is very helpful in order to use combined field integral equations (numerical damping of fictitious eigenfrequencies) without the problems caused by the evaluation of hypersingular integrals. The key aspects are: (i) the approach deals with discrete equations from the very beginning; (ii) discrete (instead of continuous) tensor Green's functions are considered (the methodology to evaluate them is shown); (iii) the boundary must be described by means of a regular square grid. In order to overcome the drawback of this third condition the boundary integral is coupled, if needed, with a thin layer of finite elements. This improves the description of curved geometries and reduces numerical errors. The method is applied to the scattering of waves by objects and holes in an unbounded elastic medium and to the solution of interior elastic problems

3:00

5pSA6. Wavelet-decomposed Rayleigh-Ritz method for plate transverse vibration simulations in high frequency regime. Su Zhang and Li Cheng (Dept. of Mech. Eng., The Hong Kong Polytechnic Univ., Hung Hom, Kowloon 00000, Hong Kong, li.cheng@polyu.edu.hk)

This paper focus on the extraordinary ability of the wavelet decomposition for plate transverse vibration analyses under the framework of Rayleigh-Ritz method. Using a normalized rectangular plate as an example, 2-d Daubechies wavelet scale functions are used as admissible functions for decomposing the flexural displacement of the plate, along with the artificial springs at the boundary, to predict vibration of the plate in an extremely large frequency range with any boundary conditions. Numerical results of eigenvalues on F-F-F-F, C-C-C-C and S-S-S-S are given, in excellent agreement with predictions of classical methods. t is shown that the use of wavelet basis allows reaching very high frequencies, typically covering nearly 15000 modes within 1% error using conventional computational facility. The frequency response under S-S-S-S case is also validated. The strategy for choosing parameters of admissible functions is also investigated. Due to the vanishing moment, compactly supported and orthogonal properties, 2-D Daubechies wavelet scale functions exhibit appealing features which allow global expansion in dealing with various vibration problems.

3:20–3:40 Break

3:40

5pSA7. Diffraction formulation for the sound radiation of a loudspeaker on a rigid convex edged shape enclosure. Vincent Roggerone (LMS, Ecole polytechnique, Laboratoire de Mécanique des Solides, École polytechnique, Palaiseau 91128, France, rogger@lms.polytechnique.fr), Etienne Corteel (Sonic Emotion Labs, PARIS, France), and Xavier Boutillon (LMS, Ecole polytechnique, Palaiseau, France)

We present a method for calculating the sound radiation of a loudspeaker on a box-shape enclosure. This method is based on the iterative Asheim-Svensson formulation [Asheim *et al.* J. Acoust. Soc. Am., 2013]. In this geometrical acoustical model, the radiated sound is given by the addition of the direct sound of a baffled piston and the sound scattered by the edges of the loudspeaker enclosure. The latter are expressed in the form of multiple-order edge diffraction components. This method becomes particularly attractive at high frequencies since the maximum order required to get a correct estimation decreases with frequency. Altogether, the method becomes less costly than the Boundary Element Method (BEM) beyond a given frequency. The convergence of the method can be estimated by considering the distance traveled by the creeping wave on the enclosure instead of considering the successive orders of diffraction by each edge. An iterative formulation follows which is well-suited to slender objects such as a sound bar. Comparisons with measurements and results of a BEM calculation will be presented. [Work supported by the ANR-13-CORD-0008 EDISON 3D grant from the French National Agency of Research.]

4:00

5pSA8. Modeling and optimization of acoustic black hole vibration absorbers. Micah R. Shepherd, Cameron A. McCormick, Stephen C. Conlon, and Philip A. Feurtado (Appl. Res. Lab, Penn State Univ., PO Box 30, mailstop 3220B, State College, PA 16801, mrs30@psu.edu)

Recent studies have shown that the acoustic black hole (ABH) effect can be used to provide vibration absorption and improved structural-acoustic response. In this talk, several aspects of the design and implementation of ABH vibration absorbers will be discussed. First, to address the competing nature of the best ABH taper for vibration reduction and the underlying theoretical assumptions, a multi-objective approach is used to find the best ABH parameters where both criteria are sufficiently met. Next, the modeling challenges associated with one- and two-dimensional ABH design will be discussed and a mesh convergence study will be presented. Finally, the use of multiobjective optimization for ABH design will be discussed for aerospace and marine applications.

4:20

5pSA9. Benchmark cases for urban sound propagation. Sai Charan Trikootam, Maarten Hornikx, Gerson C. Fernandes (Bldg. Phys. and Services, Dept. of the Built Environment, Eindhoven Univ. of Technol., P.O.Box 513, Eindhoven 5600 MB, Netherlands, S.C.Trikootam@tue.nl), Twan van Hooff (Bldg. Phys. Section, Dept. of Civil Eng., KU Leuven, Leuven, Belgium), Bert Blocken (Bldg. Phys. and Services, Dept. of the Built Environment, Eindhoven Univ. of Technol., Eindhoven, Netherlands), and Timothy Van Renterghem (Dept. of Information Technol., Ghent Univ., Gent, Belgium)

Urban sound propagation is influenced by multiple reflections on horizontal and vertical surfaces, either specular or diffusive in nature, diffraction around edges and meteorology, causing refraction and scattering of sound waves. This paper initiates multiple benchmark cases for urban sound propagation, with the purpose of comparing the suitability of computational methodologies. The benchmark cases are two-dimensional cross-sections of typical urban geometries, involving all of the effects mentioned above. The sound source is either geometrically screened or is in the line of sight from the receiver's position. When meteorological conditions are included, results obtained from computational fluid dynamics simulations are used. All details of the benchmark cases are concisely described, and results from two numerical methods for outdoor sound propagation are included. Both methods solve the linearized Euler equations (LEE). The first method is the Fourier pseudospectral time-domain method (Fourier-PSTD) implemented in the open source software openPSTD v2.0. The second method is the finite difference time domain method FDTD. A detailed comparison of the results obtained from the two methods is presented.

4:40

5pSA10. Reduced order modeling in topology optimization of vibroacoustic problems. Ester Creixell Mediante (R&D, Oticon A/S, Kongebakken 9, Smørum 2765, Denmark, emed@oticon.com), Jakob Søndergaard Jensen, Jonas Brunskog (Elec. Eng., Acoust. Technol. group, Tech. Univ. of Denmark, Kgs. Lyngby, Denmark), and Martin Larsen (R&D, Oticon A/S, Smørum, Denmark)

There is an interest in introducing topology optimization techniques in the design process of structural-acoustic systems. In topology optimization, the design space must be finely meshed in order to obtain an accurate design, which results in large numbers of degrees of freedom when designing complex 3D parts. The optimization process can therefore become highly time consuming due to the need to solve a large system of equations at each iteration. Projection-based parametric Model Order Reduction (pMOR) methods have successfully been applied for reducing the computational cost of material or size optimization in large vibroacoustic models; however, new challenges are encountered when dealing with topology optimization. Since a design parameter per element is considered, the total number of design variables becomes very large; this poses a challenge to most existing pMOR techniques, which suffer from the curse of dimensionality. Moreover, the fact that the nature of the elements changes throughout the optimization (material to void or material to air) makes it more difficult to create a global basis that is accurate throughout the whole design space. In this work, these challenges are investigated and different approaches to achieving an efficient reduction technique for such problems are discussed.

5:00

5pSA11. Fast iterative solution of acoustic multiple scattering problems. Feruza Amirkulova (Mech. Eng., Western New England Univ., 124 Raymond Ave, Poughkeepsie, New York 12604, feruza.amirkulova@wne.edu)

Multiple scattering (MS) of waves by a system of scatterers is of great theoretical and practical importance. It is required in a wide variety of physical contexts such as the implementation of "invisibility" cloaks, the effective parameter characterization, etc.An iteratively computable Neumann series (NS) expansion technique is employed to expedite the MS solution. The method works if the spectral radius of the interaction matrix is less than one. The spectral properties of this matrix are investigated for different configurations of cylinders; the validity of solution is shown by modifying the number of scatterers M and their separation distance, d. The iterative solution works well for two rigid cylinders at any considered values of frequency,ω and d. The results show that the iterative algorithm is fast. The convergence rate analysis shows that the effectiveness of technique depends on the M, ω and d. Shrinking d, and rising M and ω weaken the convergence. We improve the algorithm by accelerating the overall convergence characteristics of NS. We use matrix manipulation and renormalization techniques provided for MS of fast atomic nuclei to improve NS convergence, and employ Padé approximants to expand the validity range of NS solutions. The iterative approach can be extended for a band of frequencies, and applied for multi-frequency wave propagation problems.

Session 5pSC

Speech Communication: Speech Perception (Poster Session)

Rajka Smiljanic, Chair
Linguistics, University of Texas at Austin, Calhoun Hall 407, 1 University Station B5100, Austin, TX 78712-0198

All posters will be on display from 1:20 p.m. to 5:20 p.m. To allow contributors in this session to see the other posters, authors of odd-numbered papers will be at their posters from 1:20 p.m. to 3:20 p.m. and authors of even-numbered papers will be at their posters from 3:20 p.m. to 5:20 p.m.

Contributed Papers

1:20

5pSC1. Definition of appropriate indicator and threshold values for intelligibility: Application for rolling stock's public address systems. Fabrice Aubin (Ctr. d'Ingenierie du Materiel, SNCF, 4 allée des Gemeaux, Le Mans 72100, France, fabrice.aubin@sncf.fr), Françoise DuBois (Dept. Modelisations et Experience Voyageur, SNCF, Paris, France), Cyril Mellet (Agence D'Essai Ferroviaire, SNCF, Vitry-sur Seine, France), Christophe Gerault (Ingenierie du Materiel, SNCF, LeMans, France), Christophe Lambourg, and Paul Peyret (Genesis, Aix En Provence, France)

The rolling stock's public address systems are used to broadcast safety information from the driver cab or the train supervisor. EU legislation request an audible information using the STI-PA intelligibility indicator be at least 0.45 in the whole train. The purpose of the study is to assess the current intelligibility indicator according to the operating conditions of the rolling stock in comparison with other indicators in the literature, then to study the correlation between indicators and the passenger perception and to define threshold values and objective scales of comfort. The first step was to define a methodology to characterize the rolling stock's public address systems. Secondly, two kinds of measurements were performed in a high speed train at several positions: the end-to-end public addresses system regarding its linear and non-linear characteristics in order to simulate distortion occurrence in the transmission channel at various operating condition. As a result of this study, standard intelligibility indicators showed poor correlations with intelligibility scores from the listening test, especially when the degradation is due to distortion. Fields of improvement are investigated using intelligibility indicators based on clean and degraded speech signal comparison.

5pSC2. Introduction of real-time visual feedback impairs intelligibility when other senses are degraded. Elizabeth D. Casserly (Dept. of Psych., Trinity College, 300 Summit St., Hartford, CT 06106, elizabeth.casserly@trincoll.edu) and Francesca Marino (Neurosci. Program, Trinity College, Hartford, CT)

Sensory feedback allows talkers to accurately control speech production. Under ordinary circumstances, real-time speech feedback is available in the auditory and somatosensory modalities. When sensation in these modes is degraded, the accuracy of speech motor control is impaired and intelligibility decreases. In the present study, we documented this drop in intelligibility for speech produced with degraded sensory feedback in the auditory (8-channel cochlear implant simulation) and somatosensory (1 mL benzocaine topical application) domains. Speakers ($N = 15$) were also asked, however, to produce speech in front of a mirror while their senses were degraded, giving them access to an alternate fully-specified but atypical source of sensory feedback. Given the large body of work showing the benefits of visual speech information in perception, we predicted that speakers would use the visual feedback to improve real-time control. Contrary to that prediction, naïve participants ($N = 38$) found speech produced with visual feedback (and degraded acoustic/somatosensory feedback) to be less intelligible overall than speech produced without it (and other senses still degraded). This deleterious effect of visual feedback was found for both aural intelligibility and visual (lipread) intelligibility.

5pSC3. Isolating sources of acoustic variability that diminish spectral contrast effects in vowel categorization. Ashley Assgari (Univ. of Louisville, 317 Life Sci. Bldg., Louisville, KY 40292, ashley.assgari@louisville.edu), Rachel M. Theodore (Univ. of Connecticut, Storrs, CT), and Christian Stilp (Univ. of Louisville, Louisville, KY)

Spectral contrast effects (SCEs) occur when spectral differences between earlier sounds (a precursor sentence) and a target sound are perceptually magnified. For example, listeners label a vowel as /ɪ/ (low F_1) more often when following a high-F_1 precursor sentence, and report /ɛ/ (high F_1) more often following a low-F_1 precursor sentence. Recently, these context effects were shown to be modulated by the acoustic variability across precursor sentences. When sentence mean f0 was highly variable from trial to trial, SCEs in categorization of /ɪ/ and /ɛ/ were significantly smaller than when mean f0 was more consistent across sentences (Assgari, Theodore, & Stilp, 2016 ASA). But, is acoustic variability in f0 the best predictor of context effects that influence categorization of vowels differing in F1? Here we tested whether F1 variability in precursor sentences has a comparable effect on vowel categorization to f0 variability. Listeners heard precursor sentences filtered to add a low-F_1 or high-F_1 spectral peak to produce SCEs in categorization of subsequent tokens from an /ɪ—ɛ/ continuum. Sentences were blocked based on low or high variability in f0 and/or F_1 regions. Comparisons of how f0 variability versus F_1 variability influence context effects in vowel categorization will be discussed.

5pSC4. Coda and tonal effects on perceived vowel duration. Yi-ling Chang and Yu-an Lu (National Chiao Tung Univ., F332, Humanities Bldg. 2, 1001 University Rd., Hsinchu 30010, Taiwan, yilingchang0922@gmail.com)

The shape of pitch contours has been shown to have an influence on the perceived vowel duration in that listeners tend to perceive vowels as shorter when they are produced as longer (compensatory listening strategy, Gussenhoven & Zhou, 2013). This study further investigates if the syllable structure allowed in one's native language and the frequency of tone types also affect perceived duration. An AX perceptual experiment was conducted using disyllabic structures (i.e., mepVC) in which the target vowel ([a] and [i]) in four duration steps (180, 210, 240, 270ms) and 5 pitch contours (LL, HH, HL, LH, HLH) and 7 codas ([p], [t], [k], [m], [n], [ŋ], [∅]) are embedded in the second syllable, to be compared with an anchor stimulus with a fixed vowel duration (225ms) and pitch (195Hz). Taiwanese Southern Min speakers, whose native language allows obstruent codas, and Mandarin speakers, whose native language only allows nasal codas, were recruited to participate in the experiment in which they were told to listen to pairs of meaningless words and asked to compare the durations of the second vowels

and rate them on a 7-point scale. The results showed that perceived duration cannot be solely explained by compensatory listening strategy. Instead, tone inventory, tone frequency and the syllable structure allowed in the native language all contribute to the perceived vowel duration.

5pSC5. Psychophysiological indices of listening effort due to noise masking and nonnative-accent. Alexander L. Francis, Jennifer Schumaker (Purdue Univ., SLHS, Lyles-Porter Hall, 715 Clinic Dr., West Lafayette, IN 47907, francisa@purdue.edu), Tessa Bent (Indiana Univ., Bloomington, IN), and Rongrong Zhang (Purdue Univ., West Lafayette, IN)

Physiological responses linked to listening effort may differ depending on whether the increased effort is due to masking vs. distortion of the speech signal (Francis, *et al.* 2016; Zekveld, *et al.* 2014), suggesting that listeners may engage different cognitive strategies to compensate for different sources of reduced intelligibility. Here, we assessed physiological and behavioral indices of effort in participants listening to short stories in two conditions: (1) nonnative-accented speech in quiet and (2) native-accented speech masked by speech-shaped noise. Compared to previous studies, stories were used instead of sentences to investigate effort over longer durations, while accented speech was used to reduce intelligibility more naturally than noise vocoding or text-to-speech synthesis. Masker noise levels were adjusted for each participant to ensure comparable word recognition performance across conditions. Individual characteristics including age, hearing thresholds, vocabulary, selective attention, working memory capacity, personality traits, and noise sensitivity were also measured. Smoothing spline ANOVAs and growth curve analysis will be used to determine overall effects of condition, as well as which traits best predict patterns of effort-related responses over time. Different sources of listening effort are predicted to induce different patterns of physiological response reflecting engagement of different cognitive processes, possibly in a trait-dependent manner.

5pSC6. Lexical influences on error patterns of adult listeners in competing speech perception. Alexandra Jesse (Psychol. and Brain Sci., Univ. of Massachusetts, 135 Hicks Way, Amherst, MA 01003, ajesse@psych.umass.edu) and Karen S. Helfer (Commun. Disord., Univ. of Massachusetts Amherst, Amherst, MA)

When listening to a talker while a second talker is audible, lexical characteristics of words in both speech streams influence the accuracy of word recognition. We examined how lexical properties influence the types of errors listeners make. In a competing speech task, younger, middle-aged, and older adults heard keywords placed in pairs of low-close probability sentence frames. Word frequency and neighborhood density of key words in both streams were systematically varied. As expected, susceptibility to the masker increased across the adult lifespan, indicated by an increase in errors phonologically related to the masker. The probability of masker-related errors increased in all listener groups when targets came from dense neighborhoods or when maskers came from sparse neighborhoods. Younger and middle-aged adults made more errors that were phonologically influenced by the masker or jointly by masker and target when targets were low-frequency rather than high-frequency words. Younger adults were also more likely to respond with the actual masker word. In contrast, older adults made more masker-related errors when hearing high-frequency rather than low-frequency masker words. Lexical properties of words in target and masker streams therefore influence the types of errors listeners make in competing speech situations. [Work supported by NIDCD R01 DC012057.]

5pSC7. Oral-facial kinematics and configuration drive asymmetries in visual vowel perception. Matthew Masapollo (Dept. of Cognit., Linguistic & Psychol. Sci., Brown Univ., 190 Thayer St., Providence, RI 02912, matthew_masapollo@brown.edu), Linda Polka (School of Commun. Sci. & Disord., McGill Univ., Montreal, QC, Canada), Lucie Ménard (Dept. of Linguist, Univ. of PQ at Montreal, Montreal, QC, Canada), James Morgan (Dept. of Cognit., Linguistic & Psychol. Sci., Brown Univ., Providence, RI), and Mark Tiede (Haskins Labs., New Haven, CT)

Masapollo, Polka, and Ménard (2016) have recently reported that adults show robust directional asymmetries in unimodal visual-only vowel discrimination: a change from a relatively less to a relatively more peripheral vowel (in F1-F2 articulatory vowel space) results in significantly better performance than a change in the reverse direction. In the present experiments, we examined the nature of the information that is critical to elicit these asymmetries. Toward this end, we created schematic analogues that retained the isolated kinematics (spatial direction and motion) and/or configuration (global shape) of the lips of Masapollo *et al.*'s model speaker. We found that subjects showed asymmetries while discriminating dynamic point-light displays that specified both the kinematics and configuration of the speaker's lips (Experiment 1). Moreover, this directional effect was not dependent on subjects' knowledge that the point-light stimuli were based on distal articulatory movements (Experiment 2). In contrast, no asymmetries emerged during the discrimination of abstract figure-eight shapes that depicted labial kinematic cues, but not labial configuration (Experiment 3). We interpret these findings as evidence that information for both lip shape and lip motion drive asymmetries in visual vowel perception and discuss the results in relation to general and specialized processes in speech perception.

5pSC8. The intelligibility of glimpsed speech: Effect of competing amplitude modulation. Rachel E. Miller, Bobby E. Gibbs, and Daniel Fogerty (Commun. Sci. and Disord., Univ. of South Carolina, 1229 Marion St., Columbia, SC 29201, remiller@email.sc.edu)

The present study investigated how amplitude modulation of an interrupted noise influences the intelligibility of glimpsed speech. Prior studies have demonstrated that amplitude modulation can aid speech intelligibility when speech is periodically interrupted. This study used a competing talker to define high-intensity and low-intensity speech glimpses that were used to interrupt the target speech. These interrupted intervals were filled by one of five different types of amplitude modulated noise; each noise was presented at two different presentation levels. Silent interruption was also examined. This resulted in 22 different experimental conditions. The results indicated effects of glimpsed speech level, noise modulation, and noise level. In agreement with previous studies using periodic interruption, significant benefit was obtained when the noise was amplitude modulated by the missing speech segment, but only when the modulation came from high-intensity speech segments. In contrast, amplitude modulation of the noise by previous high-intensity, preserved portions of the target sentence produced lower intelligibility scores. Additionally, speech recognition with noise amplitude modulated by a competing talker interacted with the glimpsed speech level. These results indicate that amplitude modulation of an interrupted noise can have different effects on sentence recognition, dependent on the informational content of the modulation.

5pSC9. Articulatory peripherality modulates relative attention to the mouth during visual vowel discrimination. Matthew Masapollo, Lauren Franklin, James Morgan (Cognit., Linguistic & Psychol. Sci., Brown Univ., 190 Thayer St., 190 Thayer St., Providence, RI 02912, matthew_masapollo@brown.edu), and Linda Polka (School of Commun. Sci. & Disord., McGill Univ., Montreal, QC, Canada)

Masapollo, Polka, and Ménard (2016) have recently reported that adults from different language backgrounds show robust directional asymmetries in unimodal visual-only vowel discrimination: a change in mouth-shape from one associated with a relatively less peripheral vowel to one associated with a relatively more peripheral vowel (in F1-F2 articulatory/acoustic vowel space) results in significantly better performance than a change in the reverse direction. In the present study, we used eye-tracking methodology to examine the gaze behavior of English-speaking subjects while they performed Masapollo *et al.*'s visual vowel discrimination task. We successfully replicated this directional effect using Masapollo *et al.*'s visual stimulus materials, and found that subjects deployed selective attention to the oral region compared to the ocular region of the model speaker's face. In addition, gaze fixations to the mouth were found to increase while subjects viewed the more peripheral vocalic articulations compared to the less peripheral articulations, perhaps due to their larger, more extreme oral-facial kinematic patterns. This bias in subjects' pattern of gaze behavior may contribute to asymmetries in visual vowel perception.

5pSC10. Talker identification is not improved by lexical access in the absence of familiar phonology. Deirdre E. McLaughlin, Cecilia C. Cheng, Emily J. Thurston, Sara C. Dougherty, and Tyler K. Perrachione (Speech, Lang., and Hearing Sci., Boston Univ., 635 Commonwealth Ave., Boston, MA 02215, dmclaugh@bu.edu)

Listeners identify talkers more accurately when they are familiar with both the sounds and words of the language being spoken. It is unknown whether lexical information alone can facilitate talker identification in the absence of familiar phonology. To dissociate the roles of familiar words and phonology, we developed Mandarin-English "hybrid" sentences, spoken in Mandarin, which can be convincingly coerced to sound like English when presented with corresponding subtitles (e.g., "wei4 gou3 chi1 kao3 li2 zhi1" becomes "we go to college"). Across two experiments, listeners learned to identify talkers in three conditions: listeners' native language (English), an unfamiliar, foreign language (Mandarin), and a foreign language paired with subtitles that primed native language lexical access (subtitled Mandarin). In Experiment 1 listeners underwent a single session of talker identity training; in Experiment 2 listeners completed three days of training. Talkers in a foreign language were identified no better when native language lexical representations were primed (subtitled Mandarin) than when foreign-language speech was presented, regardless of whether listeners had received one or three days of talker identity training. These results suggest that the facilitatory effect of lexical access on talker identification depends on the availability of familiar phonological forms.

5pSC11. The effects of phonotactic probability on auditory recognition of pseudo-words. Matthew C. Kelley and Benjamin V. Tucker (Linguist, Univ. of AB, University of AB, 4-23 Assiniboia Hall, Edmonton, AB T6G 2E7, Canada, mckelley@ualberta.ca)

A number of speech perception studies have been carried out to investigate how we process audio signals containing real words. However, comparatively fewer studies have been conducted looking at how we process audio signals containing phonotactically valid pseudo-words. Many traditional metrics, such as lexical frequency, used as predictors in this kind of analysis are difficult or impossible to calculate for pseudo-words, but other metrics like phonotactic probability can be calculated for such pseudo-words. Phonotactic probability is the likelihood that a particular sequence of phones occurs in a particular language, and it can be easily calculated using existing pronunciation and lexical frequency data. The present study uses the CMU Pronouncing Dictionary and the Google Ngram datasets to calculate phonotactic probability measures for each stimulus in an existing set of English auditory lexical decision responses and then statistically models the influence of phonotactic probability on the participants' reaction times to the stimuli. This modeling shows a general trend of response times decreasing as phonotactic probability increases. The results are then framed in the greater picture of speech perception and word recognition overall.

5pSC12. The effect of allophonic variation and phonotactic restriction on categorization. Yu-an Lu (National Chiao Tung Univ., Taiwan, F319, Humanities Bldg. 2, 1001 University Rd., Hsinchu 30010, Taiwan, yuanlu@nctu.edu.tw) and Jiwon Hwang (Stony Brook Univ., Stony Brook, NY)

This study presents a case from Korean which demonstrates that *allophonic* and *phonotactic* knowledge has an effect on listeners' categorization choices. In Korean, voicing is generally treated as redundant on consonants since the only voiced consonants are nasals and liquids. However, voiced oral stops can occur intervocalically as allophones of the voiceless lenis counterpart. A further restriction comes from the velar nasal [ŋ], which is restricted to coda position. Two identification experiments, in which Korean listeners were presented with a 10-step continuum from voiced oral to nasal stop in three places of articulation and were asked to decide whether they heard a nasal or not, showed an earlier boundary shift in the nasal responses in the initial position than in the medial position. This finding suggests that Korean listeners do perceive the redundant/allophonic [voice] feature and are likely to interpret a stop containing this feature as nasal in initial position but as an allophonic oral variant in medial position. An effect of place was also observed: Korean listeners were significantly more likely to perceive a nasal segment on the labial/dental continuum than on the velar continuum, suggesting a bias against hearing the phonotactically illegal *ŋ in onset.

5pSC13. Effects of the variation of phoneme duration on word processing. Catherine Ford, Filip Nenadic, Daniel Brenner, and Benjamin V. Tucker (Dept. of Linguist, Univ. of AB, Unit 204 7111 80 Ave NW, Edmonton, AB T6B0C7, Canada, cford1@ualberta.ca)

Contextually predictable, high frequency, competitor-dense words are often produced with less contrastive categories in informal conversation (Plug, 2011; Gahl *et al*., 2012; Tucker & Ernestus, 2016). One of the more frequent ways this manifests is through changes in phoneme duration, with shorter duration related to less careful speech (Gahl *et al*., 2012). However, initial observations point to large temporal variation occurring even in isolated words produced in controlled settings. The present study investigates how temporal variation affects processing speed for single words. A number of measures of temporal variation (e.g., word mean standardized phoneme duration) are compared, while controlling for a variety of psycholinguistic variables. Data from the Massive Auditory Lexical Decision project (Tucker & Brenner, 2016) was used. 232 native speakers of English responded to a subset of 26800 words from a full range of word types produced in isolation by a single speaker. Temporal variation measures are assessed based on their contribution to models predicting participant response latencies. Results offer insights into different operationalizations of temporal variation at the word level and how this durational variation influences speed of processing.

5pSC14. On the nature of vocalic representation during lexical access. Lauren Franklin and James Morgan (Brown Univ., CLPS, Brown University, Box 1821, Providence, RI 02912, lauren_franklin@brown.edu)

The current study explores vowel perception and its effects on lexical access by asking the following questions: (1) how does severity of mispronunciation in vowels affect listeners' ability to recognize familiar words? (2) What is the appropriate metric for measuring severity of vowel mispronunciation? (3) How do vowel mispronunciations and consonant mispronunciations compare in effects on spoken word recognition? Forty-eight monolingual American English speaking adults were tested in an eye tracking task requiring listeners to look to either a referent of a familiar label or a novel referent with no known label when the vowel of the familiar label was altered along one, two, or three phonological dimensions. Results showed that participants were sensitive to changes in vowels; time course analyses showed earlier divergence for vowel mispronunciations than for consonant mispronunciations in previous studies. However, vowel mispronunciations had less effect on activation of target lexical items than had consonant mispronunciations. A continuous measure of psychophysical acoustic difference better predicted participants' looking patterns than a discrete measure of phonological features (height, backness, and roundedness). Together, these results suggest that in American English, vowels condition lexical access less strongly than consonants, perhaps due in part to their more continuous perception.

5pSC15. Effects of intelligibility on within- and cross-modal sentence recognition memory. Sandie Keerstock and Rajka Smiljanic (Linguist, The Univ. of Texas at Austin, The University of Texas at Austin, 305 E. 23rd St. CLA 4.400 E9 Mail Code: B5100, Austin, TX 78712, keerstock@utexas.edu)

Listener-oriented speaking style adaptations improve sentence recognition memory in quiet and in noise (Van Engen *et al.* 2012, Gilbert *et al.* 2014). It is not clear if this advantage arises from the available cognitive resources, or enhanced memory traces. The present study investigated what may underlie the improved recognition memory for clear speech. In the within-modal condition, monolingual English listeners heard a total of 40 unique meaningful sentences in conversational and clear speaking styles, followed by a test phase in which they identified 80 audio sentences as old or new. In the cross-modal condition, the acoustic input in the test phase was replaced by written text. Results showed that while accuracy was high in both conditions, conversational-to-clear speech modifications significantly improved sentence recognition memory only in the within-modal condition. In the cross-modal condition, speaking style had no effect on sentence recognition memory. The results suggest that the benefit of clear speech over conversational speech on recognition memory lies in exaggerated acoustic-phonetic cues whose presence is necessary to activate memory

traces. Speaking style adaptations may promote the encoding and storing of lower-level acoustic information rather than higher-level semantic information.

5pSC16. The social requirements of speech perception distract drivers, not cell phones. Kevin B. McGowan (Linguist, Univ. of Kentucky, 1415 Patterson Office Tower, Lexington, KY 40506, kbmcgowan@uky.edu) and Jamison Nielsen (STEAM Acad. High School, Lexington, KY)

Talking on a cellphone impairs driving. This is true regardless of whether the cellphone is held (Strayer & Johnston 2001) or hands-free (Strayer *et al.*) (cf. Fitch *et al.* (2013)). Distraction seems to arise from split attention between the cognitive needs of driving and of listening to speech. Importantly, however, listening to the radio or podcasts is not distracting (Strayer *et al.* 2015). Perhaps the resolution to this apparent contradiction lies in the integration of social and linguistic knowledge during perception (e.g. Sumner *et al.* 2013). We hypothesize that listeners performing a task in a social listening situation may prioritize stimuli that are available to their interlocutor. We recorded listeners' responses in a between-subjects go/no-go task under a non-social control (podcast) condition and in both non-shared and shared experimental conditions. Participants in the non-shared condition listened to a story told by a live interlocutor seated in a nearby soundbooth. In the shared condition, listeners were aware that their screen was visible to this interlocutor. Data collection is still underway, but early results suggest that listeners in the shared-screen condition do indeed show less apparent impairment than listeners in the non-shared condition. This result is consistent with Sumner *et al*'s model of social speech perception and suggests that distracted driving is a question of speech perception.

5pSC17. Aspects of a biological model of categorical perception. Chris Neufeld (Univ. of Maryland, 1401 Marie Mount Hall, College Park, MD 20742, cdeneufeld@gmail.com)

Formal models of human phonetic perception are often formulated at a highly abstracted, computational level of description. One cost of such an approach is that it can be exceedingly difficult to translate a "high-level" computational theory into the "low-level" neural circuitry which implicitly underpins any theory of human perception. Here we present our initial efforts to formulate a theory of phonetic perception in terms of known neurological primitives—in this case a particular mathematical characterization of the stimulus/response characteristics of neurons in mammalian auditory cortex: the spectro-temporal receptive field. We propose that phonetic categories can be modeled as ensembles of these cells, and use computer simulations to demonstrate that such an approach exhibits the psycho-acoustical warping characteristic of categorical perception. This model has no acoustic "features" as traditionally construed: no formants, no spectral moments, no MFCCs, etc. The primitive objects of this model are simply the spectrogram-like time-frequency representations generated by the inner ear. Thus, this model is formulated entirely in terms of the peripheral and central primitives of the ascending auditory pathway. The approach also has appealing computational properties, and can be demonstrated to be a computationally tractable, efficient coding transform of speech acoustics.

5pSC18. Talker identification in three types of background noise. Alina Razak, Emily J. Thurston, Lauren E. Gustainis, Gerald Kidd, Jayaganesh Swaminathan, and Tyler K. Perrachione (Dept. of Speech, Lang. & Hearing Sci., Boston Univ., 635 Commonwealth Ave., Boston, MA 02215, arazak@bu.edu)

Perceiving speech under informational masking requires listeners to distinguish and attend to one talker from among competing talkers. However, it is unknown how well listeners are even able to identify a target talker in the presence of noise, much less how this ability is affected by factors such as noise type, signal-to-noise ratio (SNR), and speech content. In a two-day training paradigm, young adults with normal hearing learned to identify five talkers by the sound of their voice in quiet. We then tested talker identification accuracy during masking by speech-shaped noise, multi-talker babble, and a single, unfamiliar competing talker. Test stimuli contained both trained and novel sentences and were presented at three different SNRs. Talker identification was highest in quiet and fell as a function of SNR for all noise types. The change in accuracy as a function of SNR was greatest for multi-talker babble. Across all noise types and SNRs, talkers were identified more accurately from trained sentences than untrained ones. There was substantial individual variability in talker identification accuracy in noise, which appeared unrelated to listeners' pure-tone hearing levels, speech perception in noise, phonological awareness, or phonological memory, even after controlling for talker identification accuracy in quiet.

5pSC19. Does lexicality or phonemic predictability affect cross-modal identification of bisyallabic items? Kauyumari Sanchez (Psych., Humboldt State Univ., 1 Harpst St., Dept. of Psych., Arcata, CA 95521, ks2271@humboldt.edu), Lorin Lachs, Alexis Carlon (Psych., California State Univ., Fresno, Fresno, CA), and Patrick LaShell (Psych., Humboldt State Univ., Arcata, CA)

Although people can match visual-only speech to auditory-only speech (and vice-versa) when the sources match under a variety of conditions (e.g. Lachs & Pisoni, 2004; Sanchez, Dias, & Rosenblum, 2013), it is unclear to what extent this ability is mediated by abstract, cognitive processes, representations, and linguistic experience (Vitevitch & Luce, 1999; Sánchez-García, Enns, & Soto-Faraco, 2013). To address this question, the current experiment replicates and extends Vitevitch & Luce (1999) by including audio and visual stimuli presentations of bisyallabic words and non-words that vary on phonemic predictability. In the cross-modal AB matching task, participants were presented with either visual-only speech (e.g. a talker's speaking face) or auditory speech (a talker's voice) in the A position. Stimuli in the B position consisted of the opposing sensory modality, counterbalanced. A significant Phonetic Probability X Presentation Order interaction was found. People perform similarly well on high and low probability words when the presentation order is A-V, but perform worse on V-A orders differently. For V-A, people perform better than expected on low probability items compared to high probability items. These results mirror previous results obtained by Sanchez, Marshall, & Lachs (2015) cross-modal matching in different languages.

5pSC20. Reduplication and the perception of consonants under repetition. Sam Zukoff (Linguist and Philosophy, MIT, Cambridge, MA) and Benjamin Storme (Linguist and Philosophy, MIT, 16 wilson ave, Somerville, MA 02145, bstorme@mit.edu)

This paper tests a hypothesis about the perception of consonants under repetition. In the Indo-European languages, stop-sonorant-initial roots reduplicate by copying C_1: *pa-plana* (schematic); however, other cluster types, including stop-stop-initial roots, display different behaviors: e.g. *pta-ptana* (cluster-copying), *ta-ptana* (C_2-copying), *a-ptana* (non-copying), never **pa-ptana*. Zukoff (2015) proposed that this divergent behavior is induced by perceptual factors: Hypothesis1: Given the same segmental context (here, V_C), a consonant is more confusable with similar consonants (e.g. $p \sim k$) if it is immediately preceded by an identical consonant than a non-identical consonant—i.e. better identification of *p* as *p* in [...*tapt*...] than in [...*papt*...]. Hypothesis2: This effect is smaller when the repeated consonant has robust acoustic cues to contrast—i.e. better identification of repeated *p* as *p* in [...*papl*...] than in [...*papt*...]. Two native French speakers were recorded reading nonce words of the shape [{p,t,k}a{p,k}{t,l}ana]. 37 English speakers listened to the stimuli mixed with a speech-shaped noise (SNR = 3 db), and were asked to identify the second consonant ([p] or [k]). On average, the percent of correct identification was smallest when the first and second consonants were identical and the second consonant was followed by [t], in accordance with the hypotheses.

5pSC21. Effects of frequency region and number of formants in an interferer on the informational masking of speech. Brian Roberts and Robert J. Summers (Psych., School of Life and Health Sci., Aston Univ., Psych., School of Life and Health Sci., Birmingham B4 7ET, United Kingdom, r.j.summers@aston.ac.uk)

This study explored whether the extent of informational masking depends on the frequency region and number of formants in an interferer. Target formants—monotonized three-formant analogues of natural sentences—were presented monaurally, with the target ear assigned randomly on each trial. Interferers were presented contralaterally. In experiment 1, single-formant interferers were created using the time-reversed F2 frequency contour and constant amplitude, RMS-matched to F2. Interferer center frequency was matched to that of F1, F2, or F3, while maintaining the extent of formant-frequency variation (depth) on a log scale. In experiment 2, the interferer comprised either one formant (F1) or all three, created using the time-reversed frequency contours of the corresponding targets and RMS-matched constant amplitudes. Including the higher formants had little effect on interferer intensity. Interferer formant-frequency variation was scaled to 0%, 50%, or 100% of the original depth. Adding an interferer lowered target intelligibility; frequency region had relatively little effect, but increasing depth of formant-frequency variation and number of formants had independent and additive effects. This suggests that the impact on intelligibility depends primarily on the number of formants and the overall extent of frequency variation in the interferer, not on any particular frequency region. [Work supported by ESRC.]

5pSC22. Impact of noise and noise reduction on processing effort: A pupillometry study. Dorothea Wendt, Renskje K Hietkamp, and Thomas Lunner (Eriksholm Res. Ctr., Rørtangvej 20, Snekkersten, Denmark 3070, Denmark, dowe@eriksholm.com)

Speech understanding in adverse listening conditions induces cognitive demands. Processing effort typically increases for acoustically degraded speech. Moreover, processing effort depends on listener-specific factors, such as cognitive abilities and hearing loss. While noise reduction schemes are able to attenuate interfering noise, the impact on processing effort has not yet been clarified. The present study examined the benefit of noise reduction on processing effort allocated for speech recognition. The influence of noise reduction on processing effort was evaluated by measuring the Peak Pupil Dilation (PPD) of 24 hearing-impaired listeners while performing a speech recognition task. Speech recognition performances and PPDs were measured at two different signal-to-noise ratios (SNRs) corresponding to either the individual 50% correct (L50) or the 95% correct (L95) performance level with and without noise reduction. Processing effort increased at the L50 compared to the L95. Moreover, the benefit of the noise reduction was demonstrated for both the L50 and the L95, i.e., where speech recognition was at ceiling performance. The results emphasized the relevance of the PPD measure as an indicator of processing effort in situations where the traditional speech reception measures fail due to ceiling effects.

5pSC23. Processing speaker variability in Mandarin word recognition. Yu Zhang and Chao-Yang Lee (Ohio Univ., Grover W239, Ohio University, Athens, OH 45701, yuzhang2009@gmail.com)

The effect of speaker variability on Mandarin word recognition was investigated in two short-term priming experiments. Prime-target pairs that are identical (e.g., 画面image—画面image) or semantically related (e.g., 声音sound—画面image) were presented to native listeners auditorily in a lexical decision task. Since identical or related pairs usually elicit faster response relative to unrelated pairs, the question is whether prime and target pairs spoken by different speakers would affect the magnitude of priming. Forty-eight listeners made lexical decisions on the targets, and the speed and accuracy of the responses were measured. It was hypothesized that speaker variability between prime and target would result in reduced priming. However, contrary to previous studies using English materials, results from neither experiments support the hypothesis. The effect of speaker variability on priming appears to depend on listeners' language background, suggesting tone-language users may be less susceptible to speaker variability in spoken word recognition.

5pSC24. Assessing the fit between phonological feature theories and neural data. Daniel McCloy and Adrian KC Lee (Inst. for Learning and Brain Sci., Univ. of Washington, University of Washington, Box 357988, Seattle, WA 98115-7988, drmccloy@uw.edu)

Phonological features have been called the most important advance in linguistic theory of the 20th century, yet there is still no consensus regarding their epistemic basis. Many feature systems are based on articulatory properties of consonants and acoustic properties of vowels, others are solely articulatory, and perceptually-based feature systems are rare, despite the fact that phonemes (and by extension, features) allegedly reflect how speech sounds are represented in the brain. Here we examine the fit between several phonological feature systems and electroencephalographic (EEG) recordings made during speech perception. We label EEG data with phonological feature values from different systems, train classifiers to categorize the EEG signals, and assess the accuracy of classifying held-out unlabeled EEG data. By varying only the phonological feature system from which we draw the training labels, we can assess the extent to which neural data recapitulate the patterns inherent in the different feature systems.

5pSC25. Priming effects of Tone 51 Sandhi in Taiwan Southern Min. Mao-Hsu Chen (Linguist, Univ. of Pennsylvania, 4529 Spruce St., Apt 305, Philadelphia, PA 19139, chenmao@sas.upenn.edu)

Tone sandhi refers to the alternation between base and sandhi tones. In Taiwan Southern Min (TSM), a word generally changes from its base tone to the corresponding sandhi tone when it occurs in non-final positions of a syntactically determined constituent. Tone sandhi of the five unchecked tones in TSM realizes itself in a quasi-circular chain shift, the phonological opacity of which poses a theoretical challenge for both the rule-based generative framework and the output-based grammar like OT. To probe into the representation of such allotonic variation, sandhi T51 → T55 was used in an auditory-auditory priming lexical decision experiment to examine priming effects between monosyllabic real-morpheme primes and their corresponding disyllabic targets including I) T55TX, underlyingly T51TX, and II) T51e33 as adjectives, a special case where T51 remains its base tone. Each target was preceded by either a T55 prime (surface-tone match for type I target and allotone match for type II), a T51 prime (underlying-tone match for both types), or a TX unrelated control prime, where X is a tone other than 55 and 51. Results showed an overall stronger facilitation effect for T51 primes than for T55 primes, indicating the significance of underlying-tone match in auditory processing.

5pSC26. Relationship between individuals' compensation for coarticulation and production. Yanyu Long (Linguist, Cornell Univ., 203 Morrill Hall, Ithaca, NY 14853, yl2535@cornell.edu)

This study examines the relationship between compensation for coarticulation and production. We tested individuals' perceptual compensation of /s/ before /u/ and ability to imitate /s/ that were manipulated to sound equally over-coarticulated before /u/ and /a/. It is hypothesized that people who compensate more should converge less to the manipulated stimuli in the coarticulatory context. All /s/ were measured by center of gravity. Regression analyses showed that perceptual compensation is not a significant predictor of imitation of the manipulated /s/ in both contexts. We then obtained individuals' perceptual and production /s/ distributions using the tasks and used an exemplar model which assigns different weights to the existing distributions and the stimuli to predict the production in imitation. The resulting stimuli weights are indicators of the degree of imitation. Again, the results showed no effect of perceptual compensation on the stimuli weight. Instead, it depends on the perceptual distributions: within a range, the perceptual distribution mean is positively correlated with stimuli weight, suggesting more imitation when the stimuli are perceptually saliently different from the existing distribution mean. These findings show that the perceptual-production relation can be better understood using a dynamic model that takes into consideration existing and new information.

Session 5pSPa

Signal Processing in Acoustics, Engineering Acoustics, and Underwater Acoustics: Signal Processing in Side Scan Sonar Systems

Daniel D. Sternlicht, Chair
Naval Surface Warfare Center Panama City Division, 110 Vernon Avenue, Panama City, FL 32407

Chair's Introduction—1:15

Invited Papers

1:20

5pSPa1. Historical development of side scan sonar. Daniel D. Sternlicht (Naval Surface Warfare Ctr. Panama City Div., 110 Vernon Ave., Panama City, FL 32407, daniel.sternlicht@navy.mil)

The seabed mapping side scan sonar, conceived and developed in the 1950s at the U.S. Navy Mine Defense Laboratory (USNMDL) [Commander and Sternlicht, J. Acoust. Soc. Am. **137**, 2307 (2015)], was inspired by the Royal Navy ASDIC Type-162, a late-WWII hull mounted side-looking sonar capable of recording the silhouettes of bottomed U-boats. The commissioned U.S. Navy C-MK-1 SHADOWGRAPH, which operated in the Megahertz band and employed curved apertures for cross-track focusing, represents the first in a long history of military and commercial innovations. Commercial side scan sonars, typically operating in the tens or hundreds of Kilohertz, reached the marketplace in the 1960s. The SHADOWGRAPH series led to the Navy's AN/AQS-14 and French DUBM-series mine detection sonars, and the active tow vehicles of the C-MK-1 served as test platforms for the first synthetic aperture sonars (SAS) fielded at USNMDL (now the Naval Surface Warfare Center Panama City Division) during the 1970s. Through the present day these systems, military and commercial, have been augmented by revolutionary: developments in swath bathymetry side scan sonar, multibeam echo sounders, and SAS; signal processing advances in acoustic tomography and spectroscopy; and information processing advances in environmental characterization, automatic target recognition and seabed change detection.

1:40

5pSPa2. Objective acoustic shadow detection with phase-measuring sidescan sonars. Christian de Moustier (10dBx LLC, PO Box 81777, San Diego, CA 92138, cpm@ieee.org)

Interpretation of seafloor acoustic images obtained with sidescan sonars is readily improved by applying conventional radiometric corrections that compensate for known fixed gains, for spreading and absorption losses, and for the effective instantaneous area ensonified by the transmitted pulse within the sonar beam. This area is a function of the angle of incidence on the bottom, which is obtained by ray-tracing from the angle of arrival of the echo relative to vertical estimated at receiver pairs in phase-measuring sidescan sonar systems equipped with inertial motion sensors. A further improvement in the acoustic imagery is achieved by estimating the in-situ roll-plane beam pattern of the sonar using all pings in a survey, then applying the resulting normalized beam pattern function to echoes from each ping at their respective attitudes. Thus effects of the sonar system and the environment are removed while preserving the physical characteristics of the seafloor acoustic backscattering process. Then, acoustic shadows can be objectively defined below a specific threshold in the signal-to-noise ratio computed from the sample complex cross-correlation coefficient function between receivers in each pair. A single shadow threshold value applies to all the data from an entire survey, with potential benefits to automatic target recognition algorithms.

2:00

5pSPa3. Synthetic aperture sonar contrast prediction and estimation. Daniel Cook (Georgia Tech Res. Inst., 7220 Richardson Rd, Smyrna, GA 30080, dan.cook@gtri.gatech.edu) and Daniel Brown (The Penn State Univ. Appl. Res. Lab., State College, PA)

Sidescan imaging sonar performance is often described using metrics such as resolution, maximum range, and area coverage rate. These attributes alone cannot completely characterize image quality. Shadow regions often carry as much, and sometimes even more, information than the direct return from a target. The shadow contrast ratio is an important metric of image quality, and it includes contributions from the sonar hardware, the environment, and the signal processing. This presentation provides an overview of a comprehensive quantitative model for predicting the contrast ratio. This analysis of image quality has significant implications for system design, mission planning, and data processing. Key design and processing tradeoffs are highlighted using design parameters typical for high-frequency synthetic aperture sonar applications.

2:20

5pSPa4. Automated change detection for synthetic aperture sonar. Tesfaye G-Michael (Naval Surface Warfare Ctr., US Navy, 110 Vernon Ave, Rm. 2C17, Panama City, FL 32407, tesfaye.g-michael@navy.mil)

Change detection in seabed mapping is the process by which regions of interest are highlighted through the comparison of current scene data with historical data, and involves techniques that have application to a wide variety of disciplines and sensing modalities. Automated comparison of multi-temporal sidescan sonar images at the pixel (or parcel) level is challenging as the background pixels change due to a variety of factors. These include image resolution that degrades as a function of range, and temporal changes in the propagation medium and seafloor structure. With the advent of synthetic aperture sonars (SAS), which provide constant resolution over the swath and are deployed on actively navigated platforms capable of precise track repetition, the automated comparison of multi-temporal seabed images has in recent years been shown to be feasible. Change detection through the automated comparison of multi-temporal SAS imagery can be generally categorized as incoherent or coherent, where incoherent implies identifying changes in the amplitude, and coherent implies detecting changes in both amplitude and phase of the images. Utilizing the phase component may lead to detection of a physical disturbance in the scene. This presentation will provide an overview of the state-of-the-art of automated sonar seabed change detection.

2:40

5pSPa5. Computing navigation corrections for co-registration of repeat-pass synthetic aperture sonar images. Vincent Myers (Defence Res. & Development Canada, 6292 Willow St., Halifax, NS B3A 3C5, Canada, vincent.myers@drdc-rddc.gc.ca), Isabelle Quidu (ENSTA-Bretagne, Brest, France), Torstein O. Sæbø, Roy E. Hansen (FFI, Kjeller, Norway), and Benoit Zerr (ENSTA-Bretagne, Brest, France)

Interferometric processing of repeat-pass Synthetic Aperture Sonar (SAS) data requires a precise co-registration of the images in order to exploit the phase information for applications such as differential interferometry or coherent change detection. Co-registration of SAS images is made difficult by changing environmental conditions and positioning ambiguities that give rise to warping functions which are not well modeled by affine or polynomial transformations, resulting in non-negligible residual misregistration errors. The objective of this presentation will be to examine these warping functions in greater detail to gain a better physical understanding of the estimated along- and across-track displacements. A non-linear least-squares optimization is used to determine the corrections to be applied to the repeat-pass data that will yield a repeat-pass image that is co-registered with the original. These corrections include sway, heave and yaw as well as sound speed errors and element-level surge adjustments. Repeat-pass SAS data collected a Kongsberg HISAS 1020 system is used to demonstrate this optimization, showing good agreement with modeled and real data for most parameters, while determining surge estimates are shown to be particularly challenging. An examination of the resulting repeat-pass coherence will also be performed.

3:00

5pSPa6. Through-the-sensor sediment characterization from autonomous underwater vehicle sonar systems. Marcia J. Isakson (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78713, misakson@arlut.utexas.edu)

Ocean bottom sediment type is a key parameter in understanding physical processes such as sediment transport and object burial as well as acoustic processes such as propagation and backscatter. However, direct methods such as coring and grab samples are time-consuming and coverage limited. The analysis of normal-incidence echo sounding has proven to be a reliable technique for certain frequency ranges. For this method, a clear understanding of the physical processes such as scattering and reflection is critical. For higher coverage, the normal-incidence echo sounder technique can be applied to through-the-sensor measurements. This is attractive since it requires no additional equipment particularly important to autonomous underwater vehicle operations. However, through-the-sensor measurements have the additional complications of understanding hardware and signal processing algorithms that were not specifically designed for sediment characterization or normal-incidence echo sounding. In this talk, measurements from two sensors, the light-weight conformal array and the small synthetic aperture minehunter, are analyzed for sediment type. The results are compared with ground truth data. Challenges and issues involved with using these sensors for sediment characterization will be discussed. [Work supported by ONR, Ocean Acoustics.]

3:20–3:40 Break

3:40

5pSPa7. Depth estimation of water column objects using interferometric synthetic aperture sonar. Torstein O. Sæbø, Roy E. Hansen, Stig A. Synnes, and Ole E. Lorentzen (Maritime Systems, Norwegian Defence Res. Establishment, P O Box 25, Kjeller NO-2027, Norway, Torstein-Olsmo.Sabo@ffi.no)

Synthetic Aperture Sonar (SAS) is a technique which delivers sonar images of the seabed with both high resolution and large area coverage rate. SAS is therefore a well suited sensor for search of small objects on the seafloor, and is an important tool in many emerging mine countermeasures (MCM) systems. An interferometric SAS system can also resolve the angle of arrival in the vertical plane, and thus estimate the depth of an object. In this talk, we present techniques using SAS imaging for depth estimation of small objects, not on the seafloor, but located in the water column. We consider the effect of geometry, sensor settings and processing parameters. For objects of interest, we present a belief propagation inspired method for estimating the depth of the objects. This method is CPU intensive, but avoids the phase ambiguity problem encountered in standard SAS interferometry. We compare this method to a coarse depth estimate acquired from the multipath response in the SAS images as well as to ground truth. We show example images and depth estimates from the Kongsberg HISAS interferometric SAS collected by a HUGIN autonomous underwater vehicle.

4:00

5pSPa8. Synthetic aperture sonar calibration using active transponders. Brian O'Donnell and Daniel Cook (Georgia Tech Res. Inst., 7220 Richardson Rd., Smyrna, GA 30080, brian.odonnell@gtri.gatech.edu)

There is an emerging desire within the synthetic aperture sonar community to establish the theory and practical means for calibrating imagery, such that each pixel value represents an actual backscatter coefficient. Image calibration has long been standard practice for remote sensing synthetic aperture radar. However, there are a number of challenges that make calibration of sonar imagery more difficult. Among these is the fact that good inexpensive passive targets (like corner reflectors used in radar) do not have an exact counterpart in underwater acoustics. Passive calibration targets for imaging sonar, such as solid spheres or specially-designed focusing spheres, are used in practice, but active transponders offer far more capability and flexibility. The advantages and applications of active transponders are discussed, along with the current development status of a project at GTRI to design and build an affordable and recoverable calibration device.

4:20

5pSPa9. Sonar imaging in extremely shallow inshore environments from an unmanned surface vehicle. Alan J. Hunter, Zuhayr Rymansaib, Benjamin W. Metcalfe, and Peter R. Wilson (Univ. of Bath, University of Bath, Bath BA2 7AY, United Kingdom, a.j.hunter@bath.ac.uk)

Sonar imaging systems and autonomous platforms are becoming smaller and more affordable. Consequently, their applications are expanding further into the civilian domain. A good example and the focus of this work is an autonomous survey system for aiding police dive teams during underwater crime scene investigations. In this paper, we describe our ultra portable and low-cost unmanned surface vehicle (USV) currently under development for the London Metropolitan Police Service. In particular, we concentrate on the challenges associated with operating the USV's sonar payloads in the extremely shallow waters associated with relevant operational scenarios; these include, canals, lakes, and reservoirs with water depths commonly in the range of only 0.5 m to 3 m. In these cases, a key challenge is the corrupting influence of multipath reflections between the floor and water surfaces. We have taken a pragmatic approach to this multipath problem that uses in-situ assessment of the sonar performance for optimal data acquisition and interpretation. Results are presented from experimental trials conducted in operational canal environments.

Contributed Papers

4:40

5pSPa10. Signal processing trade-offs for the representation and quality assessment of acoustic color signatures. J. D. Park (Appl. Res. Lab. Penn State Univ., P.O.Box 30, State College, PA 16804, jdp971@psu.edu), Daniel Cook (Georgia Tech Res. Inst., Smyrna, GA), and Alan J. Hunter (Univ. of Bath, Bath, United Kingdom)

Acoustic color is a representation of an object, typically 2-D, which is constructed to show the evolution of the spectral response over the aspect of the object. The two natural axes for this representation are frequency and the aspect to the object. It is intuitive to assume finer resolution in both dimensions would lead to more information extractable for improved quality. However, with conventional linear track data collection methods, there is an inherent trade-off between signal processing decisions and the amount of information that can be utilized without loss of quality. In this work, this trade-off is investigated for an object with a simple geometry, and various quality metrics are discussed. For objects with spectral response that changes slowly over aspect, quality can be improved with methods such as synthetic aperture sonar processing. However, the effect of these processing decisions can depend on the type of object being represented. Other representation approaches as extensions of acoustic color will also be explored, such as time-evolving acoustic color that shows how the spectral response changes within a ping cycle.

5:00

5pSPa11. Performance of coherence based adaptive sonar beamformers. Andreas Austeng, Ole Marius H. Rindal (Dept. of Information, Univ. of Oslo, POBox 1080, Blindern, Oslo 0316, Norway, Andreas.Austeng@ifi.uio.no), Alfonso Rodriguez-Molares (Circulation and Medical Imaging, Norwegian Univ. of Sci. and Technol., Trondheim, Norway), and Roy E. Hansen (Norwegian Defence Res. Establishment (FFI), Kjeller, Norway)

Recent progress in hardware and computing power has lead to an increased interest in adaptive beamforming for sonar imaging. Several algorithms that for each pixel scale the original sonar image with a coherence estimate of the received data have been proposed. All of these algorithms claim to improve both contrast and resolution. Close examination of the claimed improvement reveals that parts of the reported effects stem from stretching of the dynamic range in the images. This effect is generally unwanted as it can be obtained without extra computation. Of importance is the actual improvement stemming from coherence scaling. In this work we examine the performance of several different coherence based adaptive beamformers (Coherence factor (CF), Generalized CF, Scaled CF, Phase CF, Spatio-Temporally Smoothed CF, and Scaled Wiener postfilter) after removing the unwanted effect from dynamic range stretching. The contrast and resolution performance are compared on simulated and experimental side scan sonar images. Even though dynamic range stretching explains most of the earlier reported improvements, there are differences among the approaches that are worth to take into consideration when choosing which algorithm to apply.

Session 5pSPb

Signal Processing in Acoustics: Audio and Array Signal Processing II

Gary W. Elko, Cochair
mh Acoustics, 25A Summit Ave, Summit, NJ 07901

Janina Fels, Cochair
Institute of Technical Acoustics, RWTH Aachen University, Neustr. 50, Aachen 52074, Germany

David C. Swanson, Cochair
Penn State ARL, 222E ARL Bldg, PO Box 30, State College, PA 16804

Contributed Papers

1:20

5pSPb1. Creation of a corpus of realistic urban sound scenes with controlled acoustic properties. Jean-Rémy Gloaguen (LAE, Ifsttar, Allée des Ponts et Chaussées Rte. de Bouaye - CS4, Bouguenais 44344, France, jean-remy.gloaguen@ifsttar.fr), Arnaud Can (LAE, Ifsttar, Bouguenais Cedex, France), Mathieu Lagrange, and Jean-François Petiot (ADTSI, IRCCyN, NANTES, France)

Sound source detection and recognition using acoustic sensors are increasingly used to monitor and analyze the urban environment as they enhance soundscape characterization and facilitate the comparison between simulated and measured noise maps using methods such as Artificial Neural Networks or Non-negative Matrix Factorization. However, the community lacks corpuses of sound scenes whose acoustic properties of each source present within the scene are precisely known. In this study, a set of 40 sound scenes typical of urban sound mixtures is created in three steps: (i) real sound scenes are listened and annotated in terms of events type, (ii) artificial sound scenes are created based on the concatenation of recorded individual sounds, whose intensity and duration are controlled to build scenes that are as close as possible to the real ones, (iii) a test is carried out to validate the level of their perceptual realism of those crafted scenes. The interest of using such corpus is then demonstrated using an important task in urban environment description: the estimation of the traffic level in urban acoustic scenes.

1:40

5pSPb2. Role of carrier signal types in perception of Thai phonemes: Implications for cochlear implant recipients. Nantaporn Saimai, Charturong Tantibundhit (Dept. of Elec. and Comput. Eng., Faculty of Eng., Thammasat Univ., Khlong Luang, Pathumthani 12120, Thailand, tchartur@engr.tu.ac.th), and Chutamanee Onsuwan (Faculty of Liberal Arts, Thammasat Univ., Bangkok, Bangkok, Thailand)

It is well known that segmental (consonants and vowels) and suprasegmental (e.g., tone and stress) speech sounds provide intrinsically different kinds of perceptual information. These differences suggest that phoneme perception might be improved in cochlear implant strategy by employing a carrier signal that is most compatible to each type of speech sounds. However, all speech vocoder strategies for cochlear implant (CI) available in most of works use one carrier signal due to their simplicity to implement. In this paper, we investigated the role of five different carrier signals for Thai speech perception on different types of phonemes, i.e., sine-carrier with/without band-pass filter (TF/TNF), noise-carrier with/without band-pass filter (NF/NNF), and temporal fine structure carrier (TFS). Each type of carrier signal was used to synthesize speech stimuli using the continuous interleaved sampling (CIS) strategy as basic and commonly used for CI. Four different psychoacoustic tests for initials, finals, vowels were performed on 20 Thai normal hearing (NH) listeners, while 13 NH listeners were used for lexical tones. Experimental results showed that sine-carriers provided the highest intelligibility scores for initials, finals, and vowels, while NNF provided the highest intelligibility scores for lexical tones. The results suggested that the way to provide an optimal speech perception for tonal language is to use two parallel carriers in speech vocoder strategy for CI.

2:00

5pSPb3. Evolving the audio equalizer. David Yonovitz (Complex Data Systems, 2560 Via Pisa, Del Mar, CA 92014, jdataAdv@san.rr.com)

Current audio equalization techniques include Shelf, Parametric, and Graphic Equalizers. They are prevalent in all professional and commercial audio implementations. These techniques modify spectral components within specified bands by applying gain or attenuation. However, current techniques each have inherent issues with their use. The issues of concern are dynamic spectrum input and the degradation of Signal-to-Noise Ratio (SNR). In the former, the input spectrum is not static, yet all the current implementations of equalization are. The equalization must be dynamic and "track" the input signal spectrum to be effective. In the case of SNR, output noise is increased when no signal is present in the spectral band when adding gain. In addition, current techniques do not posses the precision required to only encompass the signal spectral components. in an evolution of equalization with input tracking capability, signal spectral components are identified; all other spectrum may be considered as noise and can be attenuated. The evolution of audio equalizers has progressed that negates the stated issues. Its implementation is realized in the Harmonic Tracking Equalizer (HTEq). Emulations of current techniques with improvement, noise reduction, and spectral modification will be presented.

Session 5pUWa

Underwater Acoustics and Physical Acoustics: Infrasound in the Ocean and Atmosphere

Philippe Blanc-Benon, Cochair
Centre acoustique, LMFA UMR CNRS 5509, Ecole Centrale de Lyon, 36 avenue Guy de Collongue, Ecully 69134 Ecully Cedex, France

Oleg A. Godin, Cochair
NOAA-Earth System Research Lab, Naval Postgraduate School, Physics Dept., 833 Dyer Road, Bldg 232, Monterey, CA 93943-5216

Chair's Introduction—1:15

Invited Papers

1:20

5pUWa1. Numerical simulations of T-wave generation and conversion at shores: Influence of slope angles and of the SOFAR channel. Alexis Bottero, Paul Cristini, and Dimitri Komatitsch (CNRS-LMA, 4, Impasse Nikola Tesla, CS40006, Marseille 13013, France, bottero@lma.cnrs-mrs.fr)

The term T waves is generally associated to acoustic signals generated by earthquakes which end up traveling horizontally in the ocean at the speed of sound in the water. After traveling over long ranges they reach the shore, where they can be reflected or converted into (visco)elastic waves. For almost 90 years, T waves have been the subject of numerous studies in order to identify their generation mechanisms. Generally, it is supposed that seismic waves are converted to horizontally propagating T-waves by a sloping sea bottom before entering the SOFAR channel. In this study, we present a two-step parametric study of the influence of the slope on a typical T-wave generation/conversion scenario. We use a 2D spectral-element method that allows for full-waveform modeling of wave propagation. In a first step we study the amount of acoustic energy channeled from a deep earthquake below a sloping sea-floor. We then model the reflection/transmission of T waves at the shore. Our results suggest that an optimum slope angle exists for T-wave generation, and that a non-negligible amount of the channeled energy can be reflected at a sloping shore. We find no evidence of a significant influence of the SOFAR channel.

1:40

5pUWa2. Global acoustic propagation and the link between hydroacoustics and infrasound. Kevin D. Heaney (OASIS Inc., 11006 Clara Barton Dr., Fairfax Station, VA 22039, oceansound04@yahoo.com)

The ocean is nearly transparent for acoustic propagation at low frequencies, leading to the detection of signals (seismic events, volcanoes and man-made signals) at distances as large as the ocean basin. When the ocean depth approaches the acoustic wavelength (~ 1500m for 1 Hz energy) the potential for conversion of energy from ocean acoustic propagation to atmospheric infrasonic propagation exists. This was observed in an earthquake off the coast of Australia (Evers *et al.* GRL 2014) energy was observed as seismic waves, hydro-acoustic waves and then infrasound waves, the latter two on the International Monitoring System (IMS) of the Comprehensive Test Ban Treaty Organization. In this paper, global scale hydro-acoustic propagation modeling from earthquakes and volcanoes will be presented with an emphasis on the locations where conversion to atmospheric propagation will is possible. Recent observations at the Aloha Cables Observatory (Butler JASA 2016) indicate the possibility of a loss of energy from 1-5 Hz as hydro-acoustic signals propagate over the Oahu-Kauai ridge (the Kauai "Keyhole"). Hydro-acoustic and infrasound modeling of this event will be presented.

2:00

5pUWa3. Characterizing global infrasonic ocean ambient noise. Alexis Le Pichon (CEA, DAM, DIF, -, Arpajon F-91297, France, alexis.le-pichon@cea.fr), Eleonore Stutzmann (IPGP, Paris, France), Fabrice Ardhuin (IFREMER, Brest, France), and Leon Sylvain (CEA, DAM, DIF, Arpajon, France)

The ability of the International Monitoring System (IMS) global infrasound network to detect atmospheric explosions and events of interest strongly depends on station specific ambient noise which includes both incoherent wind noise and real coherent infrasonic waves. To characterize the coherent ambient noise, a broadband array processing was performed on 10 years of continuous recordings at IMS stations. Multi-year comparisons between the observed and modeled directional microbarom amplitude variations at several IMS stations using two-dimensional wave energy spectrum ocean wave products are performed to build of a reference database of infrasound oceanic sources. Microseisms are attributed the same source processes as microbaroms, involving the interaction of standing ocean waves. To further evaluate oceanic wave action models, the infrasound analysis will be supplemented with several other approaches

including microseisms collected at seismic instrumentation (single stations and arrays). The expected benefits of such studies concern the use of multi-year complementary data to finely characterize coupling mechanisms at the ocean-atmosphere interface. In return, a better knowledge of the source of the ambient ocean noise opens new perspectives by providing additional integrated constraints on the dynamics of the middle atmosphere and its disturbances where data coverage is sparse.

2:20

5pUWa4. Infrasound propagation in the atmosphere with mesoscale fluctuations induced by internal gravity waves. Igor Chunchuzov, Sergey Kulichkov, Oleg Popov, and Vitaly Perepelkin (Obukhov Inst. of Atmospheric Phys., 3 Pyzhevskii Per., Moscow 119017, Russian Federation, igor.chunchuzov@gmail.com)

The influence of the mesoscale wind velocity and temperature fluctuations induced by internal gravity waves (IGWs) on infrasound propagation in the atmosphere is studied. The statistical characteristics of the fluctuations in the parameters of infrasonic signals (such as variances and temporal spectra of the fluctuations in travel time and angle of arrival, amplitude and time duration) caused by gravity wave-associated fluctuations are studied based on the nonlinear model of shaping of the 3-D spatial spectrum of the fluctuations. The nonlinear shaping mechanism for the 3-D spectrum is associated with both the non-resonance interactions between IGWs and wave breaking processes caused by the wave-induced shear or convective instabilities. The 1-D wave number spectra (vertical and horizontal) of the mesoscale fluctuations obtained from the 3-D model are compared with the observed spectra derived from the radar, lidar and airplane temperature and wind measurements in the middle atmosphere. The results of theory and numerical modeling of infrasound scattering from gravity wave-associated fluctuations are presented. The vertical profiles of the wind velocity fluctuations in the stratosphere and lower thermosphere up to the altitudes of 130 km are retrieved from the infrasound scattered from the mesoscale wind velocity and temperature fluctuations. The results of acoustic probing of the stably stratified atmospheric boundary layer using detonation source of acoustic pulses are discussed.

2:40

5pUWa5. Infrasound in the stratosphere captured with balloon-borne sensors. Daniel C. Bowman (Ground Based Monitoring, Sandia National Labs., P. O. Box 5800, M.S. 0404, Albuquerque, NM 87185-0404, dbowma@sandia.gov), Jonathan M. Lees (Geological Sci., Univ. of North Carolina at Chapel Hill, Chapel Hill, NC), Alexis Le Pichon (CEA/DAM/DIF, Arpajon, France), and Stephen Arrowsmith (Ground Based Monitoring, Sandia National Labs., Santa Fe, NM)

Infrasound sensors launched into the stratosphere on high altitude balloons experience zero wind noise, travel across regions inaccessible to ground stations, and access a unique geoacoustic wave field that has not been examined in half a century. Recent flights in the American Southwest have detected one to several events per hour, a capability that may be due to the lack of wind noise and the balloons' presence in a stratospheric wave guide. The provenance of these events is currently uncertain, but a known explosive source was detected at greater ranges compared to nearby ground stations. A 14 day circumnavigation of Antarctica revealed that high altitude sensors have lower noise floors than International Monitoring System (IMS) infrasound stations. The ocean microbarom was continuously recorded during this flight. The sensors flew directly over spatially extended microbarom source regions circulating along the Antarctic Circumpolar Current; these signals are also detected on IMS stations. Other signals that resemble far-field explosions were detected in the stratosphere, as well as many whose sources are unknown. We look forward to the development of a new branch of geoacoustics focusing on signals recorded in the free atmosphere at extreme range, above the open ocean, and over phenomena such as severe storms that preclude ground based observation.

Contributed Papers

3:00

5pUWa6. Balloon borne infrasound platforms: Low noise, high yield. Jonathan M. Lees (Geological Sci., Univ. of North Carolina, UNC Chapel Hill, 104 South Rd., CB #3315, Chapel Hill, NC 27599-3315, jonathan.lees@unc.edu) and Daniel C. Bowman (Geological Sci., Univ. of North Carolina, Albuquerque, NM)

A balloon borne infrasound sensor circumnavigating Antarctica over the Southern Ocean shows significant broad band signals recorded on a simple microphone. A 10 hour portion of the flight traversed South American Andes, in southern Chile just north of the active Villarrica Volcano. We present signals recorded during this expedition and suggest that the high frequency signals represent transmissions from the earth, or possible bolide activity. While the source of the high frequency signals is yet unknown we provide a statistical analysis of the content during the flight. Verification of signals using chemical explosions in New Mexico demonstrate that infrasound signals are well recorded on stratospheric stations floating hundreds of km from the source. Comparison of stratospheric floating balloon platforms with ground based (IMS) infrasound stations shows considerable improvement in the noise floor, suggesting that balloon borne signals offer a high fidelity alternative to traditional acoustic installations.

3:20–3:40 Break

3:40

5pUWa7. Caustics of waves in the thermally conducting, viscous atmosphere. Oleg A. Godin (Phys. Dept., Naval Postgrad. School, Naval Postgrad. School, Phys. Dept., 833 Dyer Rd., Bldg 232, Monterey, CA 93943-5216, oagodin@nps.edu)

This paper considers waves in the atmospheric as linear acoustic-gravity waves, in which fluid's buoyancy and compressibility simultaneously serve as restoring forces. Atmosphere is modeled as an inhomogeneous, moving fluid with nearly horizontal winds and gradual variation of its composition, sound speed and wind velocity having representative spatial scales that are large compared to the wavelength. Uniform short-wave asymptotic expansions of the wave field in the presence of a simple caustic are derived and utilized to quantify and compare the effects of wave diffraction and dissipation. Unlike acoustic waves, the geometric, or Berry, phase plays an important role in the caustic asymptotics of acoustic-gravity waves. The uniform asymptotic expansion of acoustic-gravity wave field in the presence of a caustic agrees with known results in the acoustic limit. Away from the caustic on its insonified side, the uniform asymptotic solution reduces to ray-theoretical results. When the source and receiver are located sufficiently far from the caustic, ray-theoretical calculations of the wave absorption are found to be applicable for rays that approach and leave the caustic.

4:00

5pUWa8. Long-range correlations of microseism-band pressure fluctuations in the ocean. Justin S. Ball (Dept. of Geological Sci., CIRES/ Univ. of Colorado, 2200 Colorado Ave. #399, Boulder, CO 80309, justin.ball@colorado.edu), Oleg A. Godin (Earth System Res. Lab., CIRES/ National Oceanic and Atmospheric Administration, Monterey, California), Läslo Evers (Dept. of Seismology and Acoust., Royal Netherlands Meteorological Inst., De Bilt, Netherlands), and Cheng Lv (Dept. of Atmospheric and Oceanic Sci., Univ. of Colorado, Boulder, CO)

We investigate the spatial coherence of underwater infrasonic noise using a yearlong time series measured by hydrophones moored off Ascension Island. Qualitative agreement with observed cross-correlations is achieved using a simple range-dependent model, constrained by earlier, active tomographic studies in the area. In particular, the model correctly predicts the existence of two weakly dispersive normal modes in the microseism frequency range, with the group speed of one of the normal modes being smaller than the sound speed in water. The agreement justifies our interpretation of the peaks of the measured cross-correlation function of ambient noise as modal arrivals, with dispersion that is sensitive to crustal velocity structure. Our observations are consistent with Scholte to Moho head wave coupled propagation, with double mode conversion occurring due to the bathymetric variations between receivers. We thus demonstrate the feasibility of interrogating crustal properties using noise interferometry of moored hydrophone data at ranges in excess of 120 km.

4:20

5pUWa9. Rayleigh scattering of a cylindrical sound wave by an infinite cylinder. Alexander B. Baynes and Oleg A. Godin (Dept. of Phys., Naval Postgrad. School, Spanagel Hall, Rm. 215, Monterey, CA 93943-5216, abbaynes1@nps.edu)

Rayleigh scattering is often considered assuming plane incident waves. However, the full Green's function is required in a number of problems, e.g., when a scatterer is located close to the ocean surface or the seafloor. This paper considers the Green's function of the two-dimensional problem that corresponds to scattering of a cylindrical wave by an infinite cylinder embedded in a homogeneous fluid. Soft, rigid, and impedance cylinders are considered. Exact solutions of the problem involve infinite series of products of Bessel functions. Here, closed-form asymptotic solutions are derived, which are valid for arbitrary source and receiver locations outside the cylinder as long as its diameter is small relative to the wavelength. The scattered wave is given by fields of two linear image sources. The image sources are located within the cylinder at its center and at the Kelvin inversion point. The latter originated in the classical electrostatic problem involving a grounded sphere and a point charge. The new asymptotic Green's functions are employed to characterize infrasound reception by sensors mounted on cylindrical bodies. Further applications of the new asymptotic Green's functions include image-source analytical solutions to scattering problems in complex interface geometries that typically require numerical methods to be solved.

4:40

5pUWa10. Remote detection of cryoacoustic signals using noise interferometry of seafloor pressure data. Justin S. Ball, Anne F. Sheehan (Geological Sci., CIRES/ Univ. of Colorado, 2200 Colorado Ave. #399, Boulder, CO 80309, justin.ball@colorado.edu), and Ted Scambos (CIRES/ National Snow and Ice Data Ctr., Boulder, CO)

We have observed a large number of 5-15 Hz hydroacoustic arrivals in seafloor pressure data collected in 2009-2010 by the Marine Observations of Anisotropy Near Aoteroa (MOANA) experiment offshore the South Island of New Zealand. Preliminary manual picking revealed that a small number (<10%) of these signals appeared to originate from Antarctica (potentially related to ice disintegration events). Automatic detection of Antarctic signals in MOANA data using frequency-wavenumber (f-k) analysis is precluded in the 5-15 Hz band by spatial aliasing. Instead we have developed a two-stage directional detection scheme based on ambient noise Green's functions between MOANA stations, which reveal noise directivity in the amplitude asymmetry between forward and reverse lags. For each 20-minute time window of the continuous dataset, we cross-correlate all pressure data, and tabulate windows containing significant energy at the predicted acoustic arrival times between stations and originating from backazimuths consistent with Antarctica. We then beamform each candidate window to estimate signal approach angles. We find that our noise-based approach succeeds in detecting our initial manually-picked events. Back-projecting our detections indicates that many correlate with known iceberg locations from satellite remote sensing studies. We also observe that detection frequency is comparatively lowest in Austral winter, as expected.

5:00–5:40 Panel Discussion

Session 5pUWb

Underwater Acoustics: Underwater Acoustic Propagation

Cathy Ann Clark, Chair
Sensors & Sonar Systems, NUWCDIVNPT, 1176 Howell Street, B1320, R457, Newport, RI 02841

Contributed Papers

1:20

5pUWb1. Estimating rates of coherent leakage from a variety of ocean surface duct types. Adrian D. Jones (Maritime Div., Defence Sci. and Technol. Group, P.O.Box 1500, Edinburgh, SA 5111, Australia, bear-jones@adam.com.au), Alec J. Duncan (Ctr. for Marine Sci. and Technol., Curtin Univ., Perth, WA, Australia), and Zhi Yong Zhang (Maritime Div., Defence Sci. and Technol. Group, Edinburgh, SA, Australia)

In prior work, we obtained a semi-empirical algorithm for the rate of leakage from a mixed layer surface duct with linear sound speed gradients both within and below the duct. Realistic surface ducts are, however, likely to include zones of nonlinear sound speed gradient. In this paper, we made an initial test of whether the leakage rate is strongly dependent on the shape of the sound speed profile or whether it may be adequately approximated based on consideration of the trapping frequency alone. We used a normal mode model to obtain the rate of leakage for each of several nonlinear surface ducts for which the sound speed profiles can be expressed analytically. In each case, the leakage has been compared with that for a duct with a linear gradient, obtained using the algorithm above. In this comparison, the different ducts have been designed so that they all have the same trapping frequency, the same gradient at the top of the respective profiles, and the same sound speed gradient below the duct.

1:40

5pUWb2. Seismic and hydroacoustic analysis of four large underwater explosions off the east coast of Florida. Jay J. Pulli (Raytheon BBN Technologies, 1300 North 17th St., Ste. 400, Arlington, VA 22209, jpulli@bbn.com) and Kevin D. Heaney (OASIS, Inc., Fairfax Station, VA)

During the summer of 2016, the US Navy detonated four large underwater explosions off the east coast of Florida as part of shock wave testing for the Littoral Combat Ships USS Jackson and USS Milwaukee. The water depth at the explosion site is 900 m. Videos of the events provide further insight into the sources. Originally listed as magnitude 3.7 earthquakes in USGS bulletins, the source types were changed to "Experimental Explosion" as the Navy began releasing information to local news stations. These explosions were recorded on US seismic stations to distances of ~2000 km and on the hydroacoustic array at Ascension Island at a distance of ~8100 km. Seismic signals are complex but cepstral analysis reveals a clear bubble pulse period of 1.15 sec, corresponding to source depths of 50 meters. At Ascension the signals arrive at an RMS level of 136 dB at an SNR of 30 dB. We modeled the propagation loss from the source to Ascension using broadband PE and find the propagation loss to be 126 dB. Backing out this loss, we estimate source levels for the two explosions at 262 dB corresponding to explosion yields of 11,000 lbs.

2:00

5pUWb3. Effects of time dispersion on echo, reverberation, and echo to reverberation ratio in a range-dependent Pekeris waveguide. Michael A. Ainslie (Acoust. and Sonar, TNO, P.O. Box 96864, The Hague 2509JG, Netherlands, michael.ainslie@tno.nl) and Dale D. Ellis (Phys. Dept., Mount Allison Univ., Dartmouth, NS, Canada)

In shallow water, active sonar performance is typically limited by reverberation, making the prediction of target echo and reverberation, and their ratio, an important part of sonar performance prediction. In range-dependent shallow water environments, these quantities are often calculated without considering the effect of dispersion. The effect of time dispersion is considered taking examples for a range-dependent bathymetry from the 2010 Weston Memorial Workshop. Using the analytical method of [M. A. Ainslie and D. D. Ellis (in press), IEEE Journal of Oceanic Engineering], combined with normal mode predictions [D. D. Ellis (1995). The Journal of the Acoustical Society of America, 97(5), 2804-2814], neglect of time dispersion is found to result in an error of up to 16 dB in the echo level for a short CW pulse (duration 3 ms). The effect of 3D geometry is considered, and results for a cylindrically symmetric bathymetry are shown to differ by up to 14 dB from the corresponding results with a Cartesian symmetry. The difficulties associated with modeling an LFM pulse are discussed.

2:20

5pUWb4. Asymptotic stepwise coupled modes for horizontally-variable acoustic media. Charles E. White, Cathy Ann Clark (Naval Undersea Warfare Ctr., 1176 Howell St., Newport, RI 02841, charlie.e.white@navy.mil), Gopu R. Potty, and James H Miller (Ocean Eng., Univ. of Rhode Island, Narragansett, RI)

Efficient and accurate mathematical codes for the prediction of underwater sound propagation are a critical component of SONAR system development and operation. The goal of the research presented herein is to develop, implement, and verify an efficient and rigorous coupled-mode solution for acoustic wave propagation in shallow water range-dependent environments. A theoretical framework involving a range-expanded inner product for capturing the coupling between modes as they propagate through a horizontally-variable medium is presented. This framework includes a novel discretization of the range-dependent acoustic medium. A difference equations approach, which implements the inner product to account for non-adiabatic energy transfer between modes, is used to recursively compute reflection and transmission coefficients throughout the discretized environment. Increased efficiency is gained in the method due to the ability to compute coupling via closed-form algebraic expressions and in the application of asymptotic analysis to simplify the transmission and reflection coefficients. Results will be shown for a sample of waveguides with horizontally-variable bottom depth, along with comparable results from R. Evans' COUPLE model and an FEM solution produced by ARL-UT. [This work was funded by the Naval Undersea Warfare Center, Newport Rhode Island.]

2:40

5pUWb5. Underwater acoustic monitoring of regulatory fish criteria within a high flow environment. Kristjan Varnik and Erik J. Kalapinski (Tetra Tech, Inc., 160 Federal St., Tetra Tech 3rd Fl., Boston, MA 02110, kristjan.varnik@tetratech.com)

Underwater acoustic monitoring addressed at assessing compliance with regulatory fish criteria, was performed in the Kennebec river. The acoustic criteria was based on sound pressure levels, although particle velocity is also an area of increasing concern. The velocity and direction shifts of the strong currents created unique challenges in the study design, deployment and data processing of hydroacoustic monitoring equipment. Both floating and static systems were needed to capture sound pressure levels at multiple river locations.

3:00–3:20 Break

3:20

5pUWb6. Underwater measurement of source level in waveguides using equivalent source method. Tengjiao He, Shiqi Mo, and Erzheng Fang (College of Underwater Acoust. Eng., Harbin Eng. Univ., Dis.Nangang Str. Nantong No.145, Harbin 150001, China, hetengjiao@hrbeu.edu.cn)

This paper develops a method to evaluate the source level of the underwater structure in shallow water waveguides. This method requires the measurement of the pressure on a closed surface surrounding the underwater structure and the knowledge of waveguide parameters, or the measurement of the pressure and the velocity on a closed surface surrounding the underwater structure. The calculation makes use of equivalent source method(ESM), and is performed in two steps. First, the reflected and scattered fields are removed to obtain the free-field pressure that would be radiated by the measured structure in an anechoic environment. Then the recovered free-field pressure field is used to calculate the source level of the measured structure. Numerical calculation was investigated to demonstrate the feasibility of the method. The results from numerical calculations indicate that the source level of underwater structures can be obtained by using the method accurately. Furthermore, the method shows a promising way to engineering applications.

3:40

5pUWb7. Potential impact of global warming on sound propagation in monsoon induced surface mixed layers. David Browning (NUWC (retired), 139 Old North Rd., Kingston, RI 02881, decibeldb@aol.com) and Peter Herstein (NUWC(retired)., Westerly, RI)

Acoustic surface ducts in mixed layers can be an important mode of sonar propagation. The precipitation from monsoons seasonally create mixed surface layers near the equator in every major ocean. Although not as robust as mixed layers that occur at higher latitudes, they may provide the best examples of the possible global warming impact on sound propagation. This would include acoustic duct propagation properties such as attenuation, surface loss, cutoff frequency, and ambient noise. This is just a first attempt, based on preliminary estimates, to evaluate possible change. To be considered are direct effects such as ocean acidification and warming water; the seasonal impact on monsoons, like greater precipitation and duration; and finally individual transient events such as storm intensity and number (including typhoons). Given the usual monsoon variability, it may be challenging to sort out changes, but monsoon duct acoustics may be a promising new area for investigators.

4:00

5pUWb8. Effects of depth-dependent sound speed on Lloyd mirror patterns. Sheri Martinelli and Charles W. Holland (Marine & Physical Acoust., ARL Penn State, P.O. Box 30, Mailstop 3230D, State College, PA 16804-0030, slm77@psu.edu)

The classical Lloyd mirror describes the interference pattern in pressure magnitude between direct and reflected propagation paths arising from a point source at depth d in the water column at a receiver located at range R, $R >> d$. In a homogeneous medium, the magnitude of acoustic pressure has a simple expression in terms of (approximate) propagation angle: $|p|=(2/R_0)|\sin kd \cos \Theta_0$. For fast calculation of reverberation strength, we can apply this pattern to capture the effects of reflections from an idealized flat sea surface. However since we are interested in long range calculations, it is expected that the existence of sound speed gradients will affect the shape of this pattern, but how significant are these effects? To study this question, we derive a generalized Lloyd mirror for a linear-in-depth sound speed profile. We present results and discuss the implications. [Work supported by ONR through a STTR partnership with OASIS.]

4:20

5pUWb9. Acoustic wave propagation in horizontally variable shallow water. Cathy Ann Clark (Sensors & Sonar Systems, NUWCDIVNPT, 1176 Howell St., B1320, R457, Newport, RI 02841, cathy.clark@navy.mil)

A unique normal mode solution which approximates sound speed variation with depth by a sequence of isovelocity layers is verified for a sample of range independent environments. Within each layer, two closed-form fundamental solutions which satisfy the separated depth-dependent wave equation are formulated with complex analogues to be used in evanescent regions. Piecewise-continuous potential mode solutions are constructed by satisfying an upper boundary condition and extending through isovelocity layers to the bottom. The value of the Wronskian for the Green's function thus obtained is used to locate the eigenvalues of the normal modes comprising the propagating field. A closed form solution in a modified two-layer Pekeris waveguide with step-wise changes in the upper layer sound speed with range is used to study horizontal mode coupling through a series of such environments and to predict propagation loss as a function of range.

4:40

5pUWb10. Estimating the influence of ice thickness and elasticity on long-range narrow-band reverberation in a representative Arctic environment. Scott D. Frank (Mathematics, Marist College, 3399 North Ave., Poughkeepsie, NY 12601, scott.frank@marist.edu) and Anatoliy N. Ivakin (Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

A full-field perturbation approach [A. Ivakin (2016), J. Acoust. Soc. Am., **140**(1), 657-665] is modified for an ice-covered ocean and applied to estimating narrow-band long-range reverberation caused by roughness of the ice-water interface. First-order approximation of the approach is used which requires the roughness amplitudes be small compared to the acoustic wavelength. An upward refracting sound speed water column profile typical for Arctic conditions is assumed. To obtain the zeroth-order Green's function and transmission loss field used in the reverberation model, elastic parabolic equation solutions [J. Collis, S. Frank, *et al*. (2016), J. Acoust. Soc. Am., **139**(5), 2672-2682] are generated in range-independent environments. Ice is represented by two layers. The first approximates acoustic properties of a relatively thin, water-saturated transitional ice layer and is described as a stratified fluid with sound speed increasing with the distance from the ice-water interface. Second layer is introduced as a homogeneous isotropic elastic medium with fixed complex shear and longitudinal wave speeds. Effects of ice properties are discussed and demonstrated by comparing reverberation calculated for different ice layer thicknesses and shear speeds varying from zero (for a fluid model) to typical ice values. [Work supported by ONR.]

5:00

5pUWb11. Extension of the array invariant to deep-water environments. Chomgun Cho, Hee-Chun Song, and Aaron Thode (SIO, UCSD, 9500 Gilman Dr., MC 0238, La Jolla, CA 92093-0238, chomgun@ucsd.edu)

The array invariant utilizes multiple arrivals separated in beam angle and travel time for broadband signals, achieving robust source-range estimation without detailed knowledge of the environments in shallow water. This approach includes the waveguide invariant parameter β, which is close to one for surface/bottom-interacting shallow-water environments. In deep water, however, β significantly varies from large negative to positive values depending on the group of modes, while being sensitive to the sound speed variation within the water column. In this paper, two different approaches are investigated: 1) an extension of the array invariant by integrating multiple arrivals with varying β and 2) the use of dispersion relationship between several pairs of arrivals. These two approaches will be applied to a deep-water environment (4-km deep) using a 13-m aperture vertical array deployed to the sound channel axis (e.g., 330-m) from the Medium Frequency Noise experiment (MFN), where a high-frequency source (above 1 kHz) also was positioned around the sound channel axis.

Closing Ceremony

The Presidents of the Acoustical Society of America and the European Acoustics Association will make short farewell speeches and announcements of future meetings.