Session 4aAA

Architectural Acoustics and Noise: Acoustic Comfort in Building Indoor Environmental Quality (IEQ) Performance 1

Kenneth P. Roy, Cochair
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Chair's Introduction—8:55

Invited Papers

9:00

4aAA1. Contributions to acoustical comfort in the built environment, 1956–2015. Jerry Christoff, Jim Good, John LoVerde, David W. Dong, and Samantha Rawlings (Veneklasen Assoc., 1711 16th St., Santa Monica, CA 90404, jloverde@veneklasen.com)

When the senior author joined Paul S. Veneklasen & Associates in 1956, research was under way on speech intelligibility relative to background noise, not in buildings but in spacecraft! Back on earth, additional studies countered the then-popular idea that gypsum board construction was low quality with poor sound isolation, leading to the development of multi-layer gypsum board wall designs for office buildings and studios. Further investigations predicted speech intelligibility based on design features and led to development of prototype masking noise systems. A set of acoustical design elements (such as tall furniture barriers, sound-absorptive barriers and ceilings, and orienting workstations to maximize distance) emerged naturally from these studies. Within the last couple of decades, however, open-plan office design has evolved away from traditional design, incorporating fewer of these design elements. This provides opportunity to evaluate the efficacy of standard design elements in modern collaborative work areas. Occupancy surveys and acoustical measurements were performed in open office areas of six companies. The results verify the effectiveness of traditional acoustical design elements, but also indicate where typical acoustical design practices may be modified while maintaining acceptable acoustical performance in collaborative open office areas.

9:20

4aAA2. Capturing office acoustics in the design and construction process. Roman Wowk and Chris Papadimos (Papadimos Group, 222 Vallejo St., 4th Fl., San Francisco, CA 94111, roman@papadimosgroup.com)

Given the realities of the design and construction process, basic qualities for office acoustics including speech privacy, freedom from distraction and speech intelligibility are often compromised or neglected altogether. A scan of recent news articles not only yields numerous accounts of poor conditions, but several academic studies have also shown how poorly acoustics compares with other indoor environmental qualities. This cannot be overlooked as a factor affecting productivity and performance. While the marketplace has no shortage of potential solutions for office acoustics, no product or design upgrade alone is a substitute for good design and planning, regardless of its role in previous solutions. Several essential but independent elements must be considered together for successful and cost-effective outcomes. This necessitates an integrated approach with consensus between user expectations and any design limitations, early consideration of acoustics during space planning, and the right balance of upgrades to meet project criteria. However, even basic oversights during construction can unravel even a wholesome design without coordination between trades, verification of acoustic performance, and field review. This presentation will show through case studies how such a process has either been successfully executed or systematically neglected and how that has affected the outcome on various projects.

9:40

4aAA3. Architecture and the environment—How much do we care? Kenneth P. Roy (Bldg. Products Innovation Lab., Armstrong World Industries, 2500 Columbia Ave., Lancaster, PA 17603, kproy@armstrong.com)

We have been hearing about "green buildings" and the drive for energy efficiency and sustainability for quite some time now. And if we think about this, we will probably realize that this focus is really on the buildings and the cost to society that is incurred by these building in terms of environmental factors such as materials, air, and water. This of course does not address the primary reason to design, build, and operate buildings—which is really about "people." People are exposed to the indoor environmental quality, and up until now, acoustic design and performance have not been exactly exemplary, as shown by POE surveys such as by the CBE at Berkeley. So, what needs to be done to make this work??—Focus on the mission of the building, after all, the building is just a tool to help us do what we need to do. So, how do we address occupant satisfaction and performance with architecture?? And how do we integrate acoustic design into the emerging architectural trends?? That's what we need to discuss.

10:00–10:15 Break

10:15

4aAA4. Research methods to investigate the effects of acoustics on occupant comfort and productivity in the built environment. Michelle C. Vigeant and William P. Bahnfleth (Graduate Program in Acoust. and Dept. of Architectural Eng., The Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, vigeant@engr.psu.edu)

A number of factors affect building indoor environmental quality (IEQ), principally thermal environment (temperature, humidity, and air movement), indoor air quality (IAQ), lighting, and acoustics. Most studies on comfort or preference and virtually all dealing with health and productivity focus on thermal environment and IAQ. However, a recent meta-analysis showed that thermal comfort ranks only slightly higher in importance than acoustic comfort and IAQ (Frontczak & Wargocki 2011). Although not focused on acoustics, these prior studies do provide insight into appropriate methodologies. Perceived IEQ is the most widely studied since it is relatively simple to measure and can be compared to perception-based acceptability criteria. Some investigations go beyond formal surveys by including quantitative measures, such as overall productivity, student learning, and health. These data can be used for monetization of benefits and cost, where the typical finding in studies focused on the thermal environment and IAQ is as high as 10:1. Overall, the field of IEQ needs to move from perceived quality metrics to performance metrics, but more work is needed to establish sufficiently accurate quantitative measures. These advances will be especially beneficial to building occupants by providing better working environments both in terms of health and productivity.

10:35

4aAA5. The national opera theatre in Bucharest—Update of the room-acoustical properties. Wolfgang Ahnert, Tobias Behrens (ADA Acoust. & Media Consultants GmbH, Arkonastr. 45-49, Berlin D-13189, Germany, wahnert@ada-amc.eu), and Radu Pana (Univ. of Architecture and Urbanism "Ion Mincu", Bucarest, Romania)

After the destruction of the old opera house in WWII, the new edifice was designed in 1952–1953. The hall and the pit was almost not changed over the time, so based on the original design the hall acoustics was like in classic opera house not very lively, i.e., very dry. Also, the mutual hearing of the musicians in the pit and between the singers on stage and the musicians was not good. The facelift of the hall involved significant changes of the orchestra pit. A new lift has been built, and the cover of the musicians in the pit has been reduced. Also, the secondary structure of all wall and ceiling parts in the pit has been modified. The same has been done in hall, the carpet in the stalls, and the galleries has been substituted by a parquet floor, the absorbing materials in the boxes, and other wall parts was exchanged by wood panels and the completely absorbing chairs have been changed to less absorbing ones. Acoustic measurements have been done before and after the refurbishment. The paper will describe all changes and the achieved results.

10:55

4aAA6. Source qualification environments and their impact on speech privacy measurements in open offices. Sean Browne and Kenneth Good (Armstrong World Industries, 2500 Columbia Ave., Lancaster, PA 17604, sdbrowne@armstrong.com)

The ASTM standards for measuring speech privacy in an open office call for specific loudspeaker dimensions and performance attributes. The standards also details specific testing environments required to qualify those attributes and set measurement reference levels. Often times, these environments are not accessible or practical during the course of a field measurement. Several additional test environments were explored in addition to the preferred and alternate test environments defined in ASTM E1179. The reference levels from these alternate environments were used, and the resulting speech privacy measurements were compared.

Session 4aAB

Animal Bioacoustics: Avian Bioacoustics

Micheal L. Dent, Cochair
Psychology, University at Buffalo, SUNY, B76 Park Hall, Buffalo, NY 14260

Robert Dooling, Cochair
Psychology, Univ of Maryland, Baltimore Ave, College Park, MD 20742

Invited Papers

8:30

4aAB1. Auditory temporal resolving power in birds. Robert Dooling (Psych., Univ. of Maryland, Baltimore Ave., College Park, MD 20742, rdooling@umd.edu)

There has long been the notion that bird song contains acoustic features, which are inaccessible to human listeners. Field and laboratory studies suggest that birds hear both the spectral and gross temporal features of bird song much like we do. But how precisely birds perceive their auditory world still remains somewhat of a mystery. Here, I review psychophysical studies that provide evidence that birds have an exquisite sensitivity to temporal fine structure in their vocalizations putting some aspects of bird vocalizations out of reach of human hearing. Work with zebra finches, in particular, shows that these birds are capable of discriminating between positive and negative Schroeder harmonic complexes at high fundamental frequencies. These birds can also discriminate temporal fine structure changes in their vocal signals that are well below human thresholds. Whether this ability is used for communication is still not clear but it could underlie the better-than-expected ability of birds to localize sound and to assess the effects sound degradation and reverberation as a complex sound travels through the natural environment.

8:55

4aAB2. Seemingly simple songs: Black-capped chickadee song revisited. Christopher B. Sturdy (Psych., Univ. of AB, P-217 Biological Sci. Bldg., Edmonton, AB T6G2E9, Canada, csturdy@ualberta.ca)

Our lab has been studying songbird communication from an integrative perspective for over 13 years. A significant part of this program involves conducting bioacoustic analyses of vocalizations that are critical to survival. Initially, black-capped chickadees and their *chick-a-dee*call were our main research focus. In more recent years, we have turned our attention back to the *fee-bee* song, intensely studied by Weisman, Ratcliffe, and colleagues. Our re-examination of the *fee-bee* song revealed several, previously unreported features of this seemingly acoustically simple vocalization. We showed that songs contain regionalized cues for dominance status, and that females respond differentially to dominant song playback irrespective of their geographic origin. Moreover, females themselves produce a *fee-bee* song not previously reported in the literature. Female song is acoustically distinct from male song, and we have used operant conditioning experiments to identify features that can be used to identify sex of the singer. Finally, we have begun to explore neural response to different vocalizations, including song, and found that responses vary by singer, listener, and vocalization type. Our current efforts are aimed at unraveling the acoustic basis of communication in this nearly ubiquitous North American species.

9:20

4aAB3. How birds perceive the auditory scene. Micheal L. Dent (Psych., Univ. at Buffalo, SUNY, B76 Park Hall, Buffalo, NY 14260, mdent@buffalo.edu)

How are birds able to effectively communicate for survival? How do they separate auditory objects that overlap on the tympanum? How do they perceive distance, overlapping sounds, and noisy signals? In many ways, birds decipher the auditory world in the same way humans do. We know they use intensity as a primary cue for distance perception, that they spatially segregate signals and noise to better hear certain signals, that they are susceptible to auditory illusions, and that spectrotemporally complex and biologically relevant signals are often better isolated than simple tonal sounds. We know these things and more from psychophysical studies in the laboratory where birds are trained to be reliable observers, using operant conditioning techniques in a controlled environment. Subjects are trained to peck keys for food reinforcement when they detect, discriminate, identify, or localize sounds. Years of research on the foundations of hearing in these animals leads us to the present day, where birds are asked much more complicated questions in studies that more closely resemble their natural auditory world. For birds, being able to hear and isolate communication signals in the environment is crucial, and we have discovered much about how they do this in complex acoustic environments.

9:45

4aAB4. Evolution of song complexity in Bengalese finches: Sexual selection and domestication as two factors. Kazuo Okanoya (Life Sci., The Univ. of Tokyo, 3-8-1 Komaba, Meguro-ku 153-8902, Japan, cokanoya@mail.ecc.u-tokyo.ac.jp)

Bengalese finches (BFs) are domesticated strains of wild white-rumped munias (WRMs) imported from China to Japan 250 years ago. BF songs are composed of multiple chunks and each chunk is a combination of 2–4 song notes. Chunks are then arranged in a finite-state probabilistic automaton. We studied how and why BFs sing such complex songs. We found the following facts. (1) The ancestral strain sing simpler songs. (2) There is high learning specificity in WRMs but not in BFs. (3) BFs have larger song control nuclei and higher level of glutamate receptor gene expressions than WRMs. (4) Both BF and WRM females prefer complex songs as measured by the nest string assay and males with complex songs are physically fitted than the males with simpler songs. These results promoted sexual selection scenario of song complexity in BFs. We further examined factors related with domestication. We examined songs of WRMs in subpopulations of Taiwan. Where there is a sympatric species to WRMs, songs were simpler. This leads to a hypothesis that in the wild songs needed to be simple to secure species identification, but under domestication this constrains was set free. [Work supported by JSPS Kakenhi #23118003.]

10:10–10:25 Break

10:25

4aAB5. What does Da Vinci have to do with it? Laurie L. Bloomfield (Psych., Algoma Univ., 1520 Queen St. E., Sault Ste. Marie, ON P6A2G4, Canada, laurie.bloomfield@algomau.ca)

It may seem unusual to suggest a relationship between Leonardo Da Vinci (1452–1519) and the study of avian communication, as he is perhaps better known for his attempts at developing a theory of human flight based on the principles of avian flight. His approach, however, was that “from the study of structure comes the knowledge of function.” Here, I present how an understanding of the structure of the various vocalizations produced by chickadees may lend to an understanding of the function of these vocalizations. Chickadees are an excellent model system for this type of research given that they produce various calls that are comprised of individual units that may function in different ways. We have conducted several bioacoustics analyses in the search for similarities and differences among call structures, as well as attempted to delineate the bioacoustical markers that would provide meaningful information to listeners. Further, I will discuss what constitutes human “language” and how the calls of chickadees may satisfy the criteria for a non-human language. With this in mind, we use various field and laboratory techniques in an attempt to understand the structure of vocalizations which may in turn convey information regarding the function of the vocalizations.

10:50

4aAB6. Extreme vocal plasticity in adult budgerigars: Analytical challenges, social significance, and underlying neurogenetic mechanisms. Timothy F. Wright (Biology Dept., New Mexico State Univ., Foster Hall, MSC 3AF, Las Cruces, NM 88003, wright@nmsu.edu), Erina Hara (HHMI Janelia Farms, Ashburn, VA), Anna M. Young (Otterbein Univ., Colombus, OH), Marcelo Araya Salas (Biology, New Mexico State Univ., Las Cruces, NM), Christine R. Dahlin (Univ. of Pittsburgh Johnstown, Johnstown, PA), Osceola Whitney (Biology, New Mexico State Univ., Las Cruces, NM), Esteban Lucero (Anschutz Medical Ctr., Univ. of Colorado Denver, Denver, CO), and Grace Smith Vidaurre (Biology, New Mexico State Univ., Las Cruces, NM)

Vocal learning is a complex trait that is expressed differently across different taxa. While many of the best-studied species exhibit close-ended learning, in which vocal signals are learned from adult tutors during juvenile critical learning periods, other species have open-ended learning in which new signals are learned throughout life. Budgerigars (*Melopsittacus undulatus*) are an extreme example of an open-ended learner, in which both sexes have a repertoire of multiple contact calls types that continually change through adulthood to match the call types of social associates. We discuss the analytical challenges posed by the rapid plasticity of the budgerigar vocal repertoire and compare results from several different approaches to characterizing call variation. We then consider the social implications and benefits of contact call matching in fission–fusion groups. Finally, we examine the mechanisms underlying this plasticity, with a special focus on the role of the gene *FoxP2*. We find downregulation of *FoxP2* mRNA and protein in the primary parrot vocal learning center, MMSt, across a variety of social conditions in which birds also show vocal plasticity. The results support the hypothesis that *FoxP2* is a key gene regulating the neural plasticity that underlies the persistent vocal plasticity exhibited by budgerigars.

11:15

4aAB7. Context-dependent categorical perception in a songbird. Stephen Nowicki (Biology, Duke Univ., 214 Biological Sci. Bldg., Durham, NC 27708, snowicki@duke.edu)

The division of continuously variable acoustic signals into discrete perceptual categories is a fundamental feature of human speech. Other animals have been found to categorize speech sounds much the same as humans do, although little is known of the role of categorical perception by animals in their own natural communication systems. A hallmark of human categorical perception of speech is that linguistic context affects both how speech sounds are categorized into phonemes, and how different versions of phonemes are produced. I first review earlier findings showing that a species of songbird, the swamp sparrow, categorically perceives the notes that constitute its learned songs and that individual neurons in the bird’s brain show categorical responses that map onto its behavioral response. I then present more recent data, using both discrimination and labeling tests, that show how swamp sparrows perceive categorical boundaries differently depending on context. These results demonstrate that there is a more complex relationship between underlying categorical representations and surface forms in the perception of birdsong. To our knowledge, this work suggests for the first time that this higher-order characteristic of human phonology is also found in a nonhuman communication system.

Contributed Paper

11:40

4aAB8. Vocal conditioning with playback of two template sounds in budgerigars. Yoshimasa Seki (Psych., The Univ. of Maryland, 1-1 Machihata-machi, Toyohashi 4418018, Japan, yoshimasa.seki@gmail.com) and Robert J. Dooling (Psych., The Univ. of Maryland, College Park, MD)

We examined a capability of budgerigars to produce a similar vocal pattern to a sound stimulus presented immediately just before. For this purpose, we trained birds using an operant conditioning procedure. In the training, two types of the birds' own call were used as the auditory stimuli. Then, they were tested by probe stimuli (another vocal pattern of the subjects' own, other birds' vocalization). At test trials, the birds vocalized not any similar sounds in response to the probe stimuli but one of the vocal patterns which was produced at the training trials. Then, we attempted to train the birds to produce vocal patterns following playback sounds which were slightly changing as the trials went. 24-step intermediate sounds between two birds' own vocal patterns were synthesized. Those intermediate sounds were shifting step-by-step from one to the other at each trial in a single session. Eventually, a bird created some novel sounds which were similar to those intermediate stimuli and had been never produced at the training trials. Taken together, the birds did not use the playback sounds as the vocal reference under the operant procedure. However, birds might store those playback sounds as the potential vocal repertoire.

THURSDAY MORNING, 5 NOVEMBER 2015 CLEARWATER, 8:30 A.M. TO 11:55 A.M.

Session 4aBA

Biomedical Acoustics and Physical Acoustics: Numerical and Analytical Modeling of Medical Ultrasound I

Martin D. Verweij, Cochair
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Robert McGough, Cochair
Department of Electrical and Computer Engineering, Michigan State University, 2120 Engineering Building, East Lansing, MI 48824

Chair's Introduction—8:30

Invited Papers

8:35

4aBA1. Simulation of therapeutic ultrasound treatments using the hybrid angular spectrum technique. Douglas Christensen (BioEng., Univ. of Utah, 50 So Central Campus Dr., Salt Lake City, UT 84112, christen@ee.utah.edu), Scott Almquist (School of Computing, Univ. of Utah, Salt Lake City, UT), Alexis Farrer (BioEng., Univ. of Utah, Salt Lake City, UT), Dennis Parker, and Allison Payne (Radiology, Univ. of Utah, Salt Lake City, UT)

Numerical simulations play an important role in therapeutic ultrasound treatments. Simulations can help with transducer design, retroactively analyze temperature patterns for insight into treatment effectiveness, and ultimately be used for patient treatment planning. Our group has developed a rapid 3D simulation tool for ultrasound beam propagation named the hybrid angular spectrum (HAS) method. HAS is an extension of the traditional angular spectrum approach that uses fast Fourier transforms to alternate between the spatial frequency domain and the space domain as the beam propagates through inhomogeneous tissue regions. In this presentation, we briefly cover the physical and algorithmic principles underlying the HAS technique, then give examples of its use in two promising high-intensity focused ultrasound (HIFU) applications. First, we employ it to retrospectively predict the heating efficiency of transcranial treatments of 14 patients undergoing treatment for essential tremor using a large phased-array transducer. Phase aberration of the beams caused by skull irregularities is a major effect and must be modeled carefully for accurate results. Second, we model the extent of the phase aberration to be expected in our group's recently developed MRI/HIFU breast treatment system and show a correlation with the degree of breast tissue inhomogeneity in the path of the beam.

8:55

4aBA2. Methods and applications for modeling of continuous-wave ultrasound fields. T. Douglas Mast (Dept. of Biomedical, Chemical, and Environ. Eng., Univ. of Cincinnati, 3938 Cardiovascular Res. Ctr., 231 Albert Sabin Way, Cincinnati, OH 45267-0586, doug.mast@uc.edu)

Techniques are presented for efficient and accurate computation of three-dimensional, continuous-wave radiated fields from ultrasound transducers of canonical geometries. Fields of flat circular pistons are computed using exact series expressions based on orthogonal function expansions of the Rayleigh integral. Fields of concave circular pistons are computed using analogous series expansions for an integral of effective source contributions over the radiating surface; this approach accurately approximates radiated fields of many focused ultrasound transducers of interest. Fields of flat and focused rectangular transducers are computed using closed-form expressions based on a modified Fresnel approximation; accurate fields are obtained at distances large compared to the radiating aperture dimensions, so that the near field for large transducers can be accurately simulated by subdividing the radiating aperture. Efficient methods for field computation facilitate design optimization for therapy and imaging devices and methods. In simulations of therapeutic ultrasound, local heat deposition is proportional to the squared magnitude of the radiated acoustic pressure. For pulse-echo imaging, beamformed echo signals can be simulated by appropriate convolutions of transmit-receive beam products with a three-dimensional model of the scattering medium. Illustrative examples include simulation of ultrasound thermal ablation, passive cavitation imaging, and pulse-echo B-mode and echo decorrelation imaging.

9:15

4aBA3. Four interpretations of temporal memory operators in the wave equation. Sverre Holm (Dept. of Informatics, Univ. of Oslo, Gaustadalleen 23B, Oslo N 0316, Norway, sverre@ifi.uio.no)

Attenuation and dispersion in ultrasound and elastography may be modeled with convolution memory operators in the wave equation. When the attenuation follows an arbitrary frequency power law the memory is a time domain power law. Since that is also a fractional time derivative, a large body of literature on fractional derivatives then becomes available for analysis and simulation. It can also be shown that, e.g., an elementary grain shearing process in an unconsolidated saturated sediment results in fractional wave equations for both compressional and shear waves. Second, some of these wave equations can be derived from constitutive equations with memory operators, ensuring satisfaction of causality and the Kramers-Kronig relations. The fractional Kelvin-Voigt and Zener models and their resulting attenuation and dispersion will in particular be discussed. The fractional operators can also be interpreted as the result of an infinite number of basic processes. Therefore, third, a fractional wave equation can be shown to be the result of an infinite number of elementary relaxation processes. Fourth, the fractional constitutive equation can also be expressed as an infinite sum of integer order derivatives of higher order which is also equivalent to a wave equation with higher order derivative terms.

Contributed Paper

9:35

4aBA4. Modeling geometrically complex layered regions in ultrasound using a modified T Matrix approach. Gregory T. Clement (Cleveland Clinic, 9500 Euclid Ave./ND 20, Cleveland, OH 44195-0001, gclement@physics.org)

A penetrable finite region consisting of layers of arbitrary shape, density, and sound speed is considered. Solutions are formulated as a T-matrix operator that is independent of the incident field, thereby allowing scattering due to any external source field to be rapidly calculated in a single operation. In departure from standard methods, boundary terms are written in their Fourier integral forms in polar coordinates. It will be shown that the approach allows scattering terms to be reduced to a single integral equation, a set of which form a block tri-diagonal matrix; a well-studied matrix form with various methods available for efficiently solving the inverse problem. A numeric algorithm is developed around the approach and validated against a previously reported wave-vector approach. Several medically relevant examples will be shown and discussed along with the potential for such methods to aid in the development of diagnostic and therapeutic techniques. [Study funded under National Institutes of Health Grant R01EB014296.]

Invited Papers

9:50

4aBA5. Next generation full-wave ultrasound simulation: Exploiting parallelism in the move toward exascale. Bradley Treeby (Medical Phys. and Biomedical Eng., Univ. College London, Biomedical Ultrasound Group, Wolfson House, 2-10 Stephenson Way, London NW1 2HE, United Kingdom, b.treeby@ucl.ac.uk), Jiri Jaros (Faculty of Information Technol., Brno Univ. of Technol., Brno, Czech Republic), and Ben Cox (Medical Phys. and Biomedical Eng., Univ. College London, London, United Kingdom)

In recent years, the capability of performing 3D full-wave nonlinear ultrasound simulations in tissue-realistic media has been demonstrated by a number of groups. These models have since underpinned a range of interesting studies into the interaction of ultrasound fields with the human body. However, simulations have thus far been limited to domain sizes on the order of 1 billion grid points. This limitation usually means either the spatial area or highest frequency of interest is truncated. Looking toward the exascale era, where supercomputers integrating over 1M cores are predicted to appear before 2020, there is a significant opportunity to use models to gain new insight into ultrasound-tissue interaction with unprecedented detail. The challenge is to develop suitable numerical methods that map to these massively parallel architectures, in particular, that minimize data movement, allow the exploitation of co-processors, and minimize the accumulation of numerical errors. Here, we show a novel domain decomposition approach for the Fourier collocation spectral method that allows ultrasound simulations to be distributed across up to 32k CPU cores or hundreds of accelerators with reasonable efficiency. Using this model, we demonstrate for the first time the possibility of ultrasound simulations exceeding 70 billion grid points.

10:10–10:25 Break

10:25

4aBA6. Ultrasound imaging of the coefficient of nonlinearity. Libertario Demi, Ruud J. van Sloun (Lab. of Biomedical Diagnostics, Eindhoven Univ. of Technol., Den Dolech 2, Eindhoven 5612 AZ, Netherlands, l.demi@tue.nl), Caifeng Shan (Philips Res., Eindhoven, Netherlands), Martin D. Verweij (Lab. of Acoutical Wavelfield Imaging, Delft Univ. of Technol., Delft, Netherlands), and Massimo Mischi (Lab. of Biomedical Diagnostics, Eindhoven Univ. of Technol., Eindhoven, Netherlands)

Cardiac ablation (CA) is increasingly used to treat atrial fibrillation. However, long-term success is relatively low, and the procedure carries serious risks. To this end, we are developing a β (coefficient of nonlinearity) imaging method that may be employed to perform both tissue characterization and real time temperature estimation to respectively plan, monitor, and execute optimal CA. Starting from a one-dimensional generalized form of the Westervelt equation, we derived an analytical procedure for extracting β which is then further adapted to echo-mode. To evaluate the method performances, *in-silico* and *in-vitro*experiments were performed. First, one- to three-dimensional simulations including linear array scanning of three-dimensional objects were obtained with the INCS method. Next, the ULA-OP scanner was used with an Esaote LA332 linear-array to image a phantom consisting of 2-layers obtained as a mixture of oil, gelatin and water. Varying the percentage of oil (β similar to fat), different β values were obtained for the two layers. Both *in-silico* and *in-vitro* results show the capability of the method to estimate β variations. Compared to existing methods, the proposed approach provides more stable estimations (spatially) and does not require a special transducer or set-up, being more easily applicable in a clinical setting.

10:45

4aBA7. Characterization of medical ultrasound fields using modeling with a boundary condition obtained from measurements. Vera Khokhlova (Dept. of Acoust., Phys. Faculty, Moscow State Univ., 1013 NE 40th St., Seattle, Washington 98105, va.khokhlova@gmail.com), Petr Yuldashev, Pavel Rosnitskiy, Maria Karzova, Oleg Sapozhnikov (Dept. of Acoust., Phys. Faculty, Moscow State Univ., Moscow, Russian Federation), Adam Maxwell (Dept. of Urology, Univ. of Washington School of Medicine, Seattle, WA), Bryan Cunitz, Michael Bailey, Lawrence Crum, and Wayne Kreider (Ctr. for Industrial and Medical Ultrasound, Univ. of Washington, Seattle, WA)

Numerical modeling is becoming an important metrological tool for accurate characterization of ultrasound fields generated by medical transducers in water and in situ. While acoustic propagation equations are well established, setting a boundary condition relevant to the experiment still remains a critical problem. Two methods differing in complexity are described to address this problem. The first method comprises 3D simulations of the Westervelt equation with a boundary condition determined from acoustic holography measurements. The second, simpler method utilizes simulations based either on the KZK or Westervelt equation with an equivalent source boundary condition obtained by matching linear simulations to low-amplitude beam profiles along and transverse to the axis of the source. Calculations with both methods are compared to fiber optic hydrophone measurements of 2D phased therapeutic arrays, a 128-element diagnostic probe, and strongly focused single-element transducers. It is shown that the 3D Westervelt model with holographic boundary condition can accurately simulate the entire nonlinear ultrasound field. The simplified methods based on an equivalent source are shown to give accurate results in the focal zone of the transducers, even when shocks are present in the focal waveform. [Work supported by NIH EB7643, EB016118, DK043881, and RSF 14-12-00974.]

11:05

4aBA8. Angular spectrum methods for nonlinear ultrasound wave propagation: Mathematical development and implementation. François Varray, Olivier Basset, and Christian Cachard (Creatis, Université de Lyon, 7 av jean capelle, Villeurbanne 69621, France, francois.varray@creatis.insa-lyon.fr)

The use of simulation tools is required in many domains, especially in medical ultrasound (US). In US propagation, the nonlinear wave distortion is used in clinical application as harmonic or contrast imaging. Various methods exist to compute and evaluate the harmonic increase depending on the medium and the probe geometry. Among these methods, the angular spectrum method (ASM) proposes good compromise between accuracy and computation time. After expressing the wave propagation equation into Fourier domain, either a quasi-linear approximation or a slowly variation envelope approximation (SVEA) can be used to evaluate the harmonic increase which provide two tools. SVEA overcomes the quasi-linear limitation and evaluates the nonlinear frequency interaction during the propagation. A GPU implementation and optimization of both techniques allow a fast computation of the full nonlinear propagation. For both techniques, the computed fields are close and validate our proposed approach. Thanks to the GPU implementation, the computation is strongly reduced compare to Matlab or C version. Of course, some limitations in ASM are present, such as the one way computation of the US field, which did not allow the computation of reflection for media with inhomogeneities in density, celerity, attenuation, or coefficient of nonlinearity.

Contributed Papers

11:25

4aBA9. Computed tomography-based aberration correction for transskull acoustic focusing: Comparison of simulations and measurements with focused ultrasound brain systems. Ryan M. Jones (Dept. of Medical Biophys., Univ. of Toronto, 2075 Bayview Ave., Focused Ultrasound Lab (C713), Toronto, ON M4N 3M5, Canada, rmjones@sri.utoronto.ca), Yuexi Huang (Physical Sci. Platform, Sunnybrook Res. Inst., Toronto, ON, Canada), Daniel Pajek, and Kullervo Hynynen (Dept. of Medical Biophys., Univ. of Toronto, Toronto, ON, Canada)

Clinical focused ultrasound brain systems currently employ computed tomography (CT)-based phase and amplitude corrections to mitigate skull-induced distortions and restore an acoustic focus at the intended target. In this study, acoustic measurements were conducted using two transcranial magnetic resonance-guided focused ultrasound systems (ExAblate 4000, InSightec, Haifa, Israel) operating at 230 and 650 kHz. The acoustic fields generated by the devices within intact, water-filled *ex-vivo* human skulls near the geometric focus were mapped using a 0.5 mm diameter needle hydrophone. In addition, the signals transmitted from each individual array element were captured at various locations within the skull cavity. These measurements were repeated without the presence of the skull in order to determine the element-specific aberrations induced by the cranial bone for each target position. The experimental measurements were simulated using three previously developed transcranial ultrasound propagation models: an analytical method similar to that currently employed by clinical brain systems, a multi-layered ray-acoustic approach, and a three-dimensional full-wave propagation model based on the Westervelt equation. We will present a comparison of the models based on their computational complexity as well as their ability to both predict the phase and amplitude aberrations induced by the skull and reproduce the measured *in-situ* pressure field distributions.

11:40

4aBA10. Comparison between computational and experimental methods for the characterization of therapeutic ultrasound fields. Subha Maruvada, Yunbo Liu, Joshua E. Soneson, Bruce A. Herman, and Gerald R. Harris (U.S. Food and Drug Administration, 10903 New Hampshire Ave., Bldg. WO 62-2222, Silver Spring, MD 20993, subha.maruvada@fda.hhs.gov)

Analytical modeling of medical ultrasound fields has been developed by the FDA to aid in pre-clinical characterization of therapeutic ultrasound devices. In order to assess this publicly available software, called the HIFU Simulator, acoustic and thermal measurements of power, pressure/intensity and temperature distribution have been performed for comparison. Measurement and modeling issues include using hydrophones and radiation force balances at therapeutic power levels, validation of simulation models, and tissue-mimicking material (TMM) development for temperature measurements. To better understand these issues, a comparison study was undertaken between simulations and measurements of the HITU acoustic field distribution in water and TMM, and temperature rise in TMM. For the specific conditions of this study, the following results were obtained. In water, the simulated values for p+ and p- were 3% lower and 10% higher, respectively, than those measured by hydrophone. In TMM, the simulated values for p+ and p- were 2% and 10% higher, respectively, than those measured by hydrophone. The simulated spatial-peak temporal-average intensity values in both water and TMM were greater than those obtained by hydrophone by 3%. Simulated and measured end-of-sonication temperatures agreed to within their respective uncertainties (coefficients of variation of approximately 20% and 10%, respectively).

THURSDAY MORNING, 5 NOVEMBER 2015 — ORLANDO, 8:30 A.M. TO 11:40 A.M.

Session 4aEA

Engineering Acoustics: Acoustic Material Characterization Methods

Michael R. Haberman, Chair
Applied Research Laboratories, The University of Texas at Austin, 10000 Burnet Rd, Austin, TX 78758

Invited Papers

8:30

4aEA1. Ultrasonic interrogation of tissues and tissue-mimicking materials with the aid of wideband hydrophone-based transducer characterization. Keith A. Wear (Ctr. for Devices and Radiological Health, Food and Drug Administration, Bldg. 62, Rm. 2104, 10903 New Hampshire Ave., Silver Spring, MD 20993, keith.wear@fda.hhs.gov)

Ultrasonic measurements can provide useful information to characterize biologic tissues and tissue-mimicking materials. Quantitative accuracy may be improved by first characterizing spatial and frequency dependence of ultrasound transducer beams, which may be measured with hydrophones in a water tank. Accuracy of estimates of acoustic output detected by hydrophones may be enhanced by deconvolving hydrophone sensitivity from hydrophone voltage output. Full deconvolution requires knowledge of magnitude and phase of hydrophone sensitivity, which may be measured over frequency bands of 1–40 MHz using time delay spectrometry. Time-delay spectrometry may also be used to provide wideband through-transmission measurements of attenuation and sound speed in tissues and tissue-mimicking materials. Much information regarding tissue composition and structure may be obtained from (1) through-transmission mode measurements of attenuation, sound speed, and dispersion, (2) pulse-echo mode measurements of backscatter, and (3) pulse-echo tracking of shear waves induced by acoustic radiation force push beams.

8:50

4aEA2. Monitoring hardening of concrete using ultrasonic guided waves. Jinying Zhu and Hongbin Sun (Civil Eng., Univ. of Nebraska-Lincoln, 1110 S 67th St, PKI 204C, Omaha, NE 68182, jyzhu@unl.edu)

Evaluation of early age properties of concrete is critical for ensuring construction quality of concrete structures. Although ultrasonic wave based methods show potential for monitoring the hardening process of concrete in laboratory, there are many challenges related to sensor installation and data acquisition in practice. In this study, the authors present an ultrasonic guided wave method that uses guided waves in a steel rebar to monitor hardening of surrounding cement paste and mortar. The longitudinal L(0,1) mode guided wave in rebar is excited by an EMAT sensor and received by a piezoceramic P-wave ultrasonic transducer. Continuous measurements during cement hydration were used to monitor the longitudinal wave attenuation resulted from leakage from the rebar to the surrounding cement paste/mortar. Shear wave velocities in cement materials were also monitored at the same time. Experiments were performed on four cement paste samples and three mortar samples. Experimental results demonstrated a strong correlation between the guided wave leakage attenuation and shear wave velocity for all tested samples, and both parameters increase with the age of cement materials.

Contributed Papers

9:10

4aEA3. Characterization of visco-elastic material parameters by means of the ultrasonic polar scan method. Koen Van Den Abeele, Arvid Martens (Dept. of Phys., KU Leuven - Kulak, E. Sabbelaan 53, Kortrijk 8500, Belgium, koen.vandenabeele@kulak.be), Mathias Kersemans, Joris Degrieck (Dept. of Mater. Sci. and Eng., Ghent Univ., Gent, Belgium), Steven Delrue (Dept. of Phys., KU Leuven - Kulak, Kortrijk, Belgium), and Wim Van Paepegem (Dept. of Mater. Sci. and Eng., Ghent Univ., Gent, Belgium)

The Ultrasonic Polar Scan (UPS) is a non-destructive technique which insonifies a material spot on a sample using ultrasonic pulses from as many oblique incidence angles $\psi(\varphi,\theta)$ as possible. Mapping the transmitted time-of-flight (TOF) and/or amplitudes as function of the incidence angle $\psi(\varphi,\theta)$ in a polar representation yields a UPS image with intriguing patterns that represent a fingerprint of the local visco-elastic properties. The present paper reports on recent advances in the revival of the UPS technique, involving the construction of an automated high-precision UPS scanner, the implementation of advanced simulation models as well as the development of efficient inversion routines. On the experimental level, unprecedented high quality TOF and amplitude UPS landscapes for a range of orthotropic (fiber reinforced) materials have been obtained. Using numerical simulation models, it can be readily demonstrated that the TOF UPS landscapes are directly connected to the elastic properties of the material, while the signal magnitudes displayed in the amplitude UPS landscapes are merely determined by the viscosity. By developing a coupled inversion scheme using the two landscapes simultaneously, we can achieve a full determination of the visco-elastic tensor. The new advances and inversion scheme will be illustrated in the case of fiber-reinforced plastics.

9:25

4aEA4. Acoustical characterization of nano-porous carbons. Kirill Horoshenkov, Michael Pelegrinis (Mech. Eng., Univ. of Sheffield, Mappin St., Sheffield S1 3JD, United Kingdom, k.horoshenkov@sheffield.ac.uk), Marco Conte (Dept. of Chemistry, Univ. of Sheffield, Sheffield, United Kingdom), Rodolfo Venegas (Carbon Air Ltd, Salford, United Kingdom), and Olga Umnova (Univ. of Salford, Salford, United Kingdom)

The acoustical and related non-acoustical properties of activated carbon were studied to understand better the effect of added catalyst and acid treatment on the nano- and micro-structure of activated, porous nano-carbon. The acoustical impedance was measured in a 45 mm diameter impedance tube with a special adaptor to accommodate a very small material quantity available for this experiment. The presence of the adaptor was compensated using the procedure detailed in Dupont *et al*. [POMA 19, 065008 (2013); http://dx.doi.org/10.1121/1.4799701]. The acoustic impedance of activated carbon was predicted using the model proposed by Venegas [Section 6.1, Ph.D. thesis, University of Salford, 2011]. The micro- and nano-scale porosities and pore sizes were determined by fitting the model to the acoustic impedance data in the frequency range between 50 and 1000 Hz. It is shown that the acid treatment and addition of catalyst result in the reduced radius of nano-pores and reduced nano-porosity. These effects are small but measurable acoustically. These results are consistent with the results of the BET experiment. This work provides the foundation for the development of acoustical methods for nano-porous material characterization which are rapid and non-invasive.

9:40

4aEA5. Acoustic characterization of flexural metamaterial elements using an impedance tube. Matthew D. Guild, David Calvo, and Gregory Orris (NRC Res. Associateship Program, Naval Res. Lab, 4555 Overlook Ave. SW, Washington, DC 20375, matthew.guild.ctr@nrl.navy.mil)

Acoustic metamaterials have been a topic of interest in recent years and have enabled exotic effective fluid properties to be achieved, including those with negative or near-zero dynamic values. These extreme properties arise through the particular design of the microstructure, which make use of simple microscale acoustic elements to create the desired macroscopic characteristics. While many of these acoustic elements utilize the quasistatic motion of the surrounding fluid, recently thin elastic plates have been receiving more attention in the use of transmission-line acoustic metamaterials. Due to the high acoustic impedance and flexural coupling of the elastic material, the acoustic characterization of these elastic structures using traditional techniques such as an air-filled impedance tube presents a significant challenge. In this work, flexural elastic elements are examined in an acoustic impedance tube. Using theoretical formulations for elastic plates in conjunction with acoustic impedance tube measurements, information about both the effective and intrinsic material properties of the flexural metamaterial elements can be obtained. The results of this analysis and its implications on acoustic metamaterial design will be discussed. [Work supported by the Office of Naval Research and the National Research Council.]

9:55–10:15 Break

Invited Papers

10:15

4aEA6. Multimode nonlinear resonant ultrasound spectroscopy (NRUS): From one-dimensional to three-dimensional characterization of the hysteretic elastic nonlinearity. Timothy J. Ulrich, Marcel Remillieux, Pierre-Yves Le Bas (Geophys. Group, Los Alamos National Lab., MS D446, Los Alamos, NM 87545, tju@lanl.gov), and Cedric Payan (Laboratoire de Mecanique et d'Acoustique, Aix-Marseille Univ., Marseille, France)

Nonlinear Resonant Ultrasound Spectroscopy (NRUS) has been used extensively over the last two decades to quantify, through the nonlinear parameter α, the hysteretic nonlinearity of materials for geophysical, biomedical, and civil engineering applications. This technique relies on the variations of the damping and frequency of a resonance mode with the amplitude of this mode. A typical NRUS experiment is conducted on a long bar using its first longitudinal mode. In some experiments, higher order modes have been used because the nonlinearity was more pronounced but the type of motion involved has not been characterized. The parameter α measured from these experiments is then used to calibrate a 1D model of the non-classical nonlinearity. As a first step toward extending this model from 1D to 3D, experiments were conducted on long bar samples (assumed to be macroscopically isotropic) where modes are excited selectively and the type of motion involved in each mode is well characterized. In this simple isotropic case, longitudinal and torsional motions are decoupled in order to find the α_{11} and α_{44} parameters that correspond to the compression (C_{11}) and shear (C_{44}) moduli, respectively.

10:35

4aEA7. Laboratory measurements of compressional and shear wave properties in reconstituted mud and comparison to marine sediment acoustic propagation models. Kevin M. Lee, Megan S. Ballard, Thomas G. Muir (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, klee@arlut.utexas.edu), Gabriel R. Venegas, and Preston S. Wilson (Mech. Eng. Dept. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

Laboratory measurements were performed to characterize propagation of compressional and shear waves in mud, whose composition differs significantly from granular marine sediments like sand or silt. Muddy sediments are colloidal suspensions of thin, irregularly shaped platelets that carry surface charges linked to their cation exchange capacities. These suspensions result in flocculent structures, which cause mud to have high porosity and exhibit gel-like behavior. Samples of reconstituted mud were prepared by mixing kaolinite powder with distilled water and then placing them under vacuum to remove air bubbles entrained by the mixing process. Pairs of hydrophones and benders elements were immersed in the samples to measure compressional (50 kHz to 500 kHz) and shear (0.1 kHz to 1.5 kHz) wave speed and attenuation, respectively. A resonator tube technique was used to infer compressional wave speed at lower frequencies (1 kHz to 10 kHz). Wet-dry mass measurements characterized the density and porosity of different mud samples, and electron microscopy was used to estimate platelet size distributions. The measured material parameters were used as inputs to various sediment acoustic propagation models. The comparison between the predicted and measured wave speeds and attenuations will be described. [Work supported by ARL:UT and ONR.]

Contributed Papers

10:55

4aEA8. Modified resonator method for laboratory measurement of the low frequency compressional wave speed in granular sediments. Gabriel R. Venegas and Preston S. Wilson (Dept. of Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, 204 E Dean Keeton St., Austin, TX 78712-1591, gvenegas@utexas.edu)

Several models have been developed to describe dispersion in granular sediments. There is an abundance of measurements confirming these models above 10 kHz, obtained via time-of-flight measurements, but there is far less data for accurate model verification under 10 kHz. The present work focuses on laboratory compressional wave speed measurements of water-saturated glass beads in the frequency range 1 kHz to 10 kHz using a resonance tube technique that, in future work, will be scaled in size to attain results below 1 kHz. In previous versions of this technique, the granular sediment completely filled the resonator; however, grain interaction with the resonator walls and the development of grain-to-grain force chains yielded undesirable results. The present method isolates a cylindrical volume of sediment from the resonator wall by a tulle net membrane and a layer of water. The effective sediment wave speed is inferred through the measured system resonance frequencies and a finite-element model that relates the system resonance frequencies to the intrinsic sound speed of the sediment. Sound speed measurements in a model sediment composed of 1-mm-diameter glass beads and distilled water agreed with the Effective Density Fluid Model (EDFM). [Work supported by ONR.]

11:10

4aEA9. Numerical and experimental analysis for the construction of an *in situ* measurement system of the acoustic impedance to be used as an academic tool. Lucas C. Lobato and Eric B. Carneiro (Structures and Construction, Federal Univ. of Santa Maria, Av. Roraima n° 1000, Santa Maria, Rio Grande do Sul 97105-900, Brazil, lucascostalobato@gmail.com)

Due to the need for a non-destructive method to measure sound absorption of acoustic materials, *in situ* methods have achieved significance in research on acoustic. So, the goal of this work is to build and test a system of *in situ* measurement of acoustic impedance to be used as an academic tool for undergraduate students of Acoustical Engineering on the Federal University of Santa Maria, Brazil. At first, two measurement techniques are introduced in this article: PP system and PU probe. Then, numerical analyses through a FEM model for both techniques are presented. The choice of FEM to model the measurement system occurs as an alternative to the BEM model, widely used in the literature. Thus, through the numerical model, errors were observed due to the finite size of sample. So, a strategy based on techniques proposed by the literature was applied, obtaining better precision. All numerical analysis were referenced on the Johnson-Champoux-Allard model of porous material. At last, a prototype of the measurement system is presented, followed by experimental results, using two microphones (PP system). There results were compared to measurement performed with an Impedance Tube.

11:25

4aEA10. Preliminary evaluation of the sound absorption coefficient of a thin coconut coir fiber panel for automotive applications. Key F. Lima, Nilson Barbieri, Fernando J. Terashima, Victor P. Rosa (Mech. Eng., PUCPR, Imaculada Conceição, 1155, Curitiba, Paraná 80215901, Brazil, keyflima@gmail.com), and Renato Barbieri (Mech. Eng., UDESC, Joinville, SC, Brazil)

Absorbent materials are fibrous or porous and must have the property of being good acoustic dissipaters. The noise reduction process occurs due to transformation of a part of the sound energy into heat. This process occurs when a sound wave propagates through pores or irregular arrangement of fibers. Sound propagation causes multiple reflections and friction of the air present in the absorbent medium transforming sound energy into heat. The acoustic surface treatment with absorbent material are widely used to reduce the reverberation in enclosed spaces or to increase the sound transmission loss of acoustic panels. In addition, these materials can also be applied to acoustic filters with the purpose to increase their efficiencies. The sound absorption depends on the excitation frequency of the sound and it is more effective at high frequencies. Natural fibers such as coconut coir fiber have a high potential to be used as sound absorbing material. Natural fibers are agriculture waste, manufacturing this fiber is a natural product, therefore an economic and interesting option. This work compares the sound absorption coefficient of a thin coconut coir fiber panel in relation to a composite panel made of fiberglass and expanded polyurethane foam used in the automotive industry. The evaluation of sound absorption coefficient was carried out with the impedance tube technique.

THURSDAY MORNING, 5 NOVEMBER 2015 GRAND BALLROOM 2, 9:00 A.M. TO 11:50 A.M.

Session 4aMU

Musical Acoustics: Stick-Slip Processes in Musical Instruments

Thomas R. Moore, Chair

Department of Physics, Rollins College, 1000 Holt Ave., Winter Park, FL 32789

Chair's Introduction—9:00

Invited Papers

9:05

4aMU1. Playability of a bowed string physical model including finite-width thermal friction and hair dynamics. Esteban Maestre (McGill Univ., Roc Boronat 138, Barcelona 08018, Spain, esteban@ccrma.stanford.edu), Carlos Spa (Univ Federico Santa Maria, Santiago, Chile), Quim Llimona, Gary P. Scavone (McGill Univ., Montreal, QC, Canada), and Julius O. Smith (Stanford Univ., Stanford, CA)

We report on the playability of a bowed-string physical model that combines digital waveguide and finite-difference time-domain frameworks. We extend previous approaches by combining a finite-width bow-string interaction model with a dynamic friction model based on simulating heat diffusion along the width of the bow. Bow hair dynamics are incorporated in the bow-string interaction, which includes two transversal string polarizations. The bridge termination is realized using a digital reflectance matrix model obtained from fitting two-dimensional driving-point admittance measurements. We present preliminary results from a playability study in which we explore the establishment of Helmholtz motion in static and dynamic bowing conditions.

9:25

4aMU2. Exploring the bowed string dynamical behavior using a linearized model approach. Vincent Debut (Instituto de Etnomusicologia - Centro de Estudo de Musica e Dança, Faculdade de Ciencias Sociais e Humanas, Universidade Nova de Lisboa, Lisbon 1069-061, Portugal, vincentdebut@ctn.ist.utl.pt), Octávio Inácio (Musical Acoust. Lab., NIMAE, School of Music and Performing Arts of the Polytechnic Inst. of Porto, Porto, Portugal), and José Antunes (Centro de Ciências e Tecnologias Nucleares, Instituto Superior Técnico, Universidade de Lisboa, Lisbon, Portugal)

For several decades bowed-strings have captured the attention of many researchers aiming for a thorough understanding of this system. Different approaches have been adopted particularly in the time-domain numerical simulations of the self-excited nonlinear regimes. Recently, the authors have been exploring the advantages of using a linearized approach to this problem, with or without the body coupling influence. Despite the highly non-linear bow/string friction force, the problem can be linearized about the average sliding velocity, as usually done in break-squeal noise, and an eigenvalue analysis can offer interesting information. For example, this approach allowed exploring the modal dynamics of bowed-string/body coupled system, studying the prediction of modes instabilities and the possible emergence of a strongly coupled mode responsible for the wolf-note, among other features. Here, using the linearized modal dynamics of bowed-strings we look in detail to the behavior of the string modes as a function of the bowing parameters. We start from a modal formulation of the string acted by the nonlinear bowing forces and develop the corresponding linearized formulation, which enables computation of the complex eigenvalues and eigenvectors as a function of the bowing velocity and normal force as well as location of the bow on the string.

9:45

4aMU3. Bifurcations in cello bowing using bow force below or above the Helmholtz regime. Rolf Bader (Inst. of Musicology, Univ. of Hamburg, Neue Rabenstr. 13, Hamburg 20354, Germany, R_Bader@t-online.de) and Mobert Mores (Dept. of Media Technol., Univ. of Appl. Sci., Hamburg, Germany)

Two bifurcation regimes with bowing forces above and below the thresholds for regular Helmholtz or sawtooth motion are investigated on a cello. During operation, string acceleration is measured at the bridge while bow force and speed are defined and measured on a bowing machine. High bow force causes subharmonics where the pitch depends on bowing pressure and velocity. Here sudden pitch changes were recorded at playing parameter thresholds while with slightly different parameters quasi-random scratchy sounds occurred. At very low bow forces, bifurcations appeared in the higher harmonics with two- and four-fold subdivision of periodicity, while the fundamental pitch remained stable. The reason for these bifurcations are bistable periodicities of the time series. To account for this, the sounds were analyzed using a Finite-Difference cochlea model which results in Interspike Intervals (ISI) with precise temporal resolution of these periodicities.

10:05

4aMU4. Maximum bow force revisited for the cello—Instrumentation with a precision pendulum. Robert Mores (Design Media Information, Univ. of Appl. Sci. Hamburg, Finkenau 35, 104, Hamburg 22081, Germany, robert.mores@haw-hamburg.de)

Schelleng (1971), Askenfelt (1989), Schumacher (1993), and Schoonderwaldt *et al.* (2008) formulated—in slightly different ways—how the maximum bow force relates to bow velocity, bow-bridge distance, string impedance, and friction coefficients. Related measurements at the respective transitions between Helmholtz und bifurcation regimes cover a diverse scenario of bowing machines and stringed instruments. So far, the empirical data does not clearly support either of the theories in a general way. A bowing pendulum of virtually infinite radius has been constructed to allow precise measurement of relevant bowing parameters. Two cellos are measured across all strings for three different bow-bridge distances. The empirical data suggest that linear relations predict the maximum bow force sufficiently well and a more distinct general model can be drawn. Furthermore, the pendulum employs an adaptive bow driving mechanism instead of a motor or engine. Such adaptive bowing discloses that mentioned regimes are stable and transitions between them sometimes require a hysteresis on force and speed variations. This explains some of the uncertainties in earlier studies and in this study. To confirm the findings the friction coefficients are measured separately by means of the same pendulum construction.

10:25–10:40 Break

10:40

4aMU5. Temporal phase evaluation of transients in the Brazilian cuíca drum. Tatiana Statsenko and Wilfried Kausel (Inst. of Music Acoust. (Wiener Klangstil), Univ. of Music and Performing Arts, Anton-von-Webern-Platz, 1, Vienna 1030, Austria, statsenko@mdw.ac.at)

Transient deformations can be observed by means of time-resolved electronic speckle pattern interferometry (trESPI), which implements a high-speed camera in the experimental setup and allows measurements of non-reproducible transients in real time. The stick-slip phenomena in musical instruments leads to transient nonlinear effects in the surface deformation, which can be resolved and characterized using trESPI. In this work, transient effects during the stick-slip motion of the Brazilian cuíca drum are investigated. Measurements of the cuíca under harmonic excitation and in playing conditions using trESPI are presented leading to a quantitative analysis of the motion.

11:00

4aMU6. Measurements of friction instruments with high-speed camera and subpixel tracking. Rolf Bader, Florian Pfeifle, Niko Plath, and Christian Koehn (Inst. of Musicology, Univ. of Hamburg, Neue Rabenstr. 13, Hamburg 20354, Germany, R_Bader@t-online.de)

Instruments working with stick-slip interactions not using a string were popular in the West since the invention of the glass harmonica by Benjamin Franklin in 1751. The Terpodion investigated was built by Buschmann in the mid-19th century as a friction instrument with a keyboard, where bars are pressed against a wooden rotating cylinder producing a sound. Using high-speed camera recordings, the only playable instrument today at the Viadrina museum in Frankfurt a.O. shows sinusoidal vibrations of the bars determining the played pitch while the radiated sound is highly complex. Therefore the instrument shows a fundamentally different stick-slip action compared to bowed instruments. Friction instruments of the East, like the singing bowls also show very sinusoidal-like sounds again caused by the stick-slip interaction. The New Ireland Lounuet, a finger-rubbed wooden block has both, a sinusoidal motion as well as a highly complex one which is to imitate bird and frog sounds. A systematic view on the different stick-slip interactions is suggested.

Contributed Papers

11:20

4aMU7. The Etiology of chatter in the Himalayan singing bowl. Chloe Keefer, Samantha Collin, and Thomas R. Moore (Dept. of Phys., Rollins College, Winter Park, FL 32789, ckeefer@rollins.edu)

The Himalayan singing bowl is a nearly symmetric idiophone played by rotating a wooden stick called a puja around the outer rim of the bowl. The vibrations of the bowl are excited by a stick-slip mechanism, which produces a radial motion of the bowl with a deflection shape similar to the (2,0) mode. We present experimental evidence that the position of the puja coincides with the point of minimum displacement on the bowl, indicating that it imposes a node in the deflection shape that rotates around the bowl with the puja. However, in many cases the puja is forced off of the bowl and an audible chatter is produced as the puja repeatedly strikes the bowl several times per second. This indicates that the position of the puja is not a node, but rather merely a point of minimum deflection. Examination of high-speed electronic speckle pattern interferograms and time-resolve acoustic spectra provide insight into the mechanics of the singing bowl and the origin of the chatter.

11:35

4aMU8. An unconditionally stable scheme for simulation of stick-slip processes. Vasileios Chatziioannou and Wilfried Kausel (Inst. of Music Acoust., Univ. of Music and performing Arts Vienna, Anton-von-Webern-Platz 1, Bldg. M, Vienna 1030, Austria, kausel@mdw.ac.at)

Stick-slip processes are often encountered in musical instruments, the most notable example being that of a bowed violin string. Modeling such nonlinear interactions has been often attempted in the field of Musical Acoustics, using a variety of time-stepping algorithms. The nonlinear nature of such problems requires careful analysis of the stability properties of the developed numerical schemes. This is usually carried out using energy based methods, in which case one needs to consider the continuous exchange of energy between the bow and the bowed object. In this paper an unconditionally stable scheme is proposed for the simulation of a bowed, lumped mass. The stability of the algorithm is ensured due to the presence of an invariant quantity. The numerical results are in agreement with well established algorithms, with the proposed methodology possessing no stability constraints and being extensible to a wide range of (lumped and distributed) acoustic systems.

THURSDAY MORNING, 5 NOVEMBER 2015 GRAND BALLROOM 1, 8:30 A.M. TO 11:40 A.M.

Session 4aNS

Noise, ASA Committee on Standards, and Psychological and Physiological Acoustics: Thoughts on the Next Generation of ANSI Loudness Standards

Patricia Davies, Chair
Ray W. Herrick Labs., School of Mechanical Engineering, Purdue University, 177 South Russell Street, West Lafayette, IN 47907-2099

Invited Papers

8:30

4aNS1. How should we move forward with the next version of the ANSI S3.4 loudness standard? Patricia Davies (Ray W. Herrick Labs., School of Mech. Eng., Purdue Univ., 177 South Russell St., West Lafayette, IN 47907-2099, daviesp@purdue.edu)

Recent updates to the international loudness standard ISO 532 reflect both increased understanding of loudness and the use of loudness standards by the acoustics and engineering communities. Both are inspiration to consider revising ANSI S3.4-2007. The current ANSI standard is based on Moore and Glasberg's model of loudness for stationary sounds. The new ISO 532-2 is a revision of this. The new ISO 532-1 is based on Zwicker's time-varying loudness, and will replace the current ISO 532B, also based on Zwicker's loudness model but for stationary sounds. For people who are not developing loudness models but are using them to assess sounds or noise, there are a number of other issues, e.g., loudness of very low frequency sounds, and exposure assessment based on loudness. These will be discussed and possible directions in updating the current ANSI standard are presented.

8:50

4aNS2. Differences between ANSI S3.4-2007 and the proposed ISO532-2. Brian C. Moore (Experimental Psych., Univ. of Cambridge, Downing St., Cambridge CB3 9LG, United Kingdom, bcjm@cam.ac.uk)

The ANSI standard for the calculation of loudness (ANSI S3.4-2007) is based on the model developed by Moore, Glasberg and co-workers, and uses the assumption that a diotic sound is twice as loud as that same sound presented monaurally. However, recent data suggest that loudness summation across ears is less than assumed in the ANSI standard. This was taken into account in the revised loudness model of Moore and Glasberg [J. Acoust. Soc. Am., **121**, 1604–1612, 2007] using the concept of "binaural inhibition", whereby the signal at one ear inhibits the internal response to a signal at the other ear. This model predicts that a diotic sound is 1.5 times as loud as that same sound presented monaurally, and it forms the basis for the proposed ISO532-2 standard. The revised model also gives reasonably accurate predictions of loudness in cases where the sounds differ across the two ears, for example, sounds recorded via a dummy head. An extension of the model to deal with time-varying sounds has been shown to give accurate predictions of loudness for a variety of technical and musical sounds. It is proposed that the revised and extended model be used as the basis for a new ANSI standard.

9:10

4aNS3. Determining binaural summation for stationary signals across the frequency spectrum. Colin J. Novak and Jeremy Charbonneau (Mech., Automotive and Mater. Eng., Univ. of Windsor, 401 Sunset Ave., Dept. of Mech. Eng., Windsor, ON N9B 3P4, Canada, novak1@uwindsor.ca)

The concept of binaural summation has been widely accepted; however, the value of summation, and the manner in which it is applied, is still under a great deal of debate. This research is an investigation of the binaural summation mechanisms through the use of a loudness comparison experiment for pure tones. Preliminary results have concluded that for pure tones presented at 40 dB the amount of summation increases with frequency from approximately 2 dB at 40 Hz up to 8 dB at 10 kHz. Due to the low hearing threshold at 40 dB, the experiment was modified such that the hearing threshold of each individual is collected in order to presented each signal at a fixed amplitude above the threshold of hearing for each frequency (i.e., a sensation level of 20 dB(SL)). However, similar outcomes have been found using this approach, thus supporting the conclusions that binaural summation does exist and is found to increase at defined increments with increasing frequency for pure tones. The eventual outcome of this work is to improve the existing loudness metrics to allow for the input of binaural measurements taken using binaural mannequins.

9:30

4aNS4. Status quo of standardizing loudness of time-varying sounds. Roland Sottek (HEAD Acoust. GmbH, Ebertstr. 30a, Herzogenrath 52134, Germany, roland.sottek@head-acoustics.de)

Recently, a new ISO standard for loudness of arbitrary sounds ISO 532-1 (Zwicker method) was proposed for the revision of ISO 532:1975 (method B). The new standard is based on DIN 45631/A1:2010, which includes the widely used standard DIN 45631:1991 for stationary sounds as a special case. DIN 45631:1991 differs slightly from ISO 532:1975 (method B) by specifying corrections for low frequencies and by restricting the description of the approach to numerical instructions only, thus allowing a unique software description. ISO 532-1 eliminates uncertainties of existing standards by strictly defining the complete procedure of loudness calculation starting with the waveform of the time signal and ending with specific and total loudness vs. time functions. The strict definition of the complete procedure, given not only by formulae and tables but also by program code, is a step forward to comparability of calculated loudness results. ISO 532-1 shall update the previous ISO 532:1975 (method B) and adapt it to proven new practice while preserving procedural and database continuity. The method according to Moore/Glasberg based on the American standard ANSI S3.4-2007, for stationary sounds only, shall replace ISO 532:1975 (method A) and will be named as ISO 532-2 in the updated version.

9:50

4aNS5. Loudness of temporally varying environmental sounds. Jesko L. Verhey, Jan Hots (Dept. of Experimental Audiol., Otto von Guericke Univ. Madgeburg, Leipziger Str. 44, Magdeburg 39120, Germany, jesko.verhey@med.ovgu.de), Moritz Wächtler, and Jan Rennies (Cluster of Excellence Hearing4all Project Group Hearing, Speech and Audio Technol., Fraunhofer Inst. for Digital Media Technol. IDMT, Oldenburg, Germany)

Loudness of speech, speech-like signals, and other dynamic environmental sounds were measured and compared to predictions of current loudness models and predictions on the basis of the relevant standards. Loudness was assessed experimentally by using a loudness matching procedure and categorical loudness scaling. The scaling method used in our study is in agreement with the requirements of the international standard on categorical loudness scaling. Categorical scaling allows for a fast assessment of loudness over a large level range but has the disadvantage that loudness is not measured in sones, as commonly used in loudness models. The data of the two methods are compared by deriving levels at equal loudness from the categorical loudness data. In addition, the present study discusses to what extend the scaling data can be compared to loudness predictions by using recently proposed equations relating categorical units to sones. The comparison of the measured levels at equal loudness and simulations revealed that for speech and speech-like signals the long-term spectrum largely determines the loudness of the sound. Dynamic models with short time constants tend to overestimate loudness. In general, this is also true for the other environmental sounds although for some technical signals discrepancies remain.

10:10–10:25 Break

10:25

4aNS6. Some factors affecting loudness measurement and prediction. Robert S. Schlauch, Edward Carney, Tzu-Ling J. Yu, and Heekyung J. Han (Speech Lang. Hearing Sci., Univ. of Minnesota, 164 Pillsbury Dr. SE, 115 Shevlin Hall, Minneapolis, MN 55455, schla001@umn.edu)

The current ANSI standard (ANSI S3.4, 2007) estimates the loudness of sustained sound, but many naturally occurring sounds have time-varying levels. Glasberg and Moore (2002) published a model for the prediction of time-varying sounds and this aspect should be considered as part of a revision to the ANSI standard. To assess the predictions of the dynamic model, loudness magnitude estimation functions were obtained for 24 listeners using pure tones (0.5 and 1.0 kHz), vowels, spondees, and speech-shaped noise (SSN) presented at levels from 40 to 90 dB SPL. Inferred equal-loudness levels from the fitted loudness functions were compared to the model predictions. The loudness model was qualitatively consistent with the behavioral data. The model predicted SSN to be louder than vowels and spondees which would be louder than tones; however, the model over-predicted the loudness differences. Possible explanations for the over-prediction include biases in loudness judgments, cognitive factors regarding learned expectations for loudness, assumptions regarding the free-field-to-headphone transfer function, and the model's over-prediction of spectral, loudness summation, a result found recently (Schlittenlacher *et al*., 2014) for the ANSI standard and DIN 45631.

10:45

4aNS7. On time-varying loudness and nonlinearly propagated sound. Andrew Marshall, Karl Oelschlaeger, and Emily Belzer (Southwest Res. Inst., 6220 Culebra Rd., San Antonio, TX 78238-5166, andrew.marshall@swri.org)

Time varying loudness models have been found to be highly correlated with subject judgments for many sources of noise. Among these sources are impulsive sounds such as sonic booms, blast noise, and thunder. Aside from the short temporal duration, all of these sound sources are distinctive in that each is of high enough amplitude to have nonlinear propagation effects occur, which change the temporal and spectral character of a signal over distance. Because of this, it is possible that differences in how a model handles spectral content may result in differences in loudness predictions over distance. A series of simulated nonlinear sources were propagated and its time-varying loudness calculated from both the Moore & Glasberg and Zwicker model. These predictions will be compared to examine how these models differ when the spectral content of the signal changes. Other parameters, including historical level metrics and those derived from loudness time histories, such as the loudness time derivative, will also be discussed.

Contributed Paper

11:05

4aNS8. Loudness and noise rating: How may standardization bridge the gap between science and engineering? Florian Völk (Bio-Inspired Information Processing, Technische Universität München, Boltzmannstraße 11, Garching 85748, Germany, florian.voelk@mytum.de)

Current noise regulations and standards, in Europe and to some extent also elsewhere, are primarily based on A-weighted sound levels. Most perceptual aspects are—if at all—taken into account by additional correction levels, for example, punishing tonality or impulsive content. On the contrary, instrumental loudness-prediction methods have been available for more than half a century and were first standardized 40 years ago. Against this background, the question arises why loudness—regardless of being calculated according to Zwicker or Moore and Glasberg—has not yet been accepted more widely in the field of noise control and related standardization. This contribution attempts to raise some possibly influential aspects, and to discuss them in the light of next-generation loudness standards. Advantages of stationary and time-varying loudness predictions over corrected weighted levels will be discussed and contrasted with potentially contradicting arguments. The discussion is primarily intended to provide arguments for the selection of future loudness-prediction standards. However, also a broader perspective is taken, looking at interrelations with other standards, algorithmically and regarding limiting values.

11:20–11:40 Panel Discussion

THURSDAY MORNING, 5 NOVEMBER 2015 ST. JOHNS, 8:30 A.M. TO 11:40 A.M.

Session 4aPA

Physical Acoustics and Noise: Launch Vehicle Acoustics I: Acoustics of Launch Vehicles and Supersonic Jets

Kent L. Gee, Cochair
Brigham Young University, N243 ESC, Provo, UT 84602

Tracianne B. Neilsen, Cochair
Brigham Young University, N311 ESC, Provo, UT 84602

Chair's Introduction—8:30

Invited Papers

8:35

4aPA1. Near-field acoustical array measurements of an impinging supersonic jet. Kent L. Gee, Tracianne B. Neilsen, Darren K. Torrie (Dept. of Phys. and Astronomy, Brigham Young Univ., N243 ESC, Provo, UT 84602, kentgee@byu.edu), Masahito Akamine (Graduate School of Frontier Sci., Univ. of Tokyo, Chiba, Japan), Koji Okamoto (Graduate School of Frontier Sci., Univ. of Tokyo, Kashiwa, Japan), Susumu Teramoto, Takeo Okunuki (Graduate School of Eng., Univ. of Tokyo, Bunkyo-ku, Tokyo, Japan), and Seiji Tsutsumi (Japanese Aerosp. Exploration Agency, Sagamihara, Kanagawa, Japan)

Impingement significantly alters the rocket plume in the near-launch pad environment, which in turn affects the acoustic radiation. Prior laboratory measurements of an unheated, Mach 1.8 ideally expanded jet impinging on 45-degree inclined flat plate were carried out using a microphone that was moved within a relatively dense grid [Akamine *et al.*, AIAA J. **53,** 2061–2067 (2015)]. A multi-institution collaboration by the authors was begun in order to conduct array measurements of the acoustic radiation from both impinging and free jets. Measurements comprised a total of 42 measurement channels located within 40 nozzle diameters. An array of two-dimensional microphone probes was placed so as to examine the transition from the hydrodynamic near field to the acoustic radiation. A scanning linear array of microphones and a stationary polar array were also designed to enable beamforming, cross correlation, and partial field decomposition. This paper describes the jet facility, experiment design, and initial analyses conducted on the data collected. [Measurements supported by the Japanese Society for the Promotion of Science.]

4a THU. AM

8:55

4aPA2. Effects of time-varying grain shape on combustion instability of a solid rocket motor. Taeyoung Park, Hunki Lee, Won-Suk Ohm (School of Mech. Eng., Yonsei Univ., Eng. Bldg. A391, 50 Yonsei-ro, Seodaemun-gu, Seoul 120-749, South Korea, pty0948@yonsei.ac.kr), and Dohyung Lee (Agency for Defense Development, Seoul, South Korea)

A common approach to modeling combustion instability in a rocket propulsion system is to express the pressure oscillation in a combustor as a superposition of acoustic modes with time-varying amplitudes. Here, the tacit assumption is that the acoustic modes themselves remain more or less the same over the entire burning time. However, in the case of a solid rocket motor, the grain shape changes significantly as combustion progresses. This can gradually alter the shapes and frequencies of the acoustic modes, the influence of which on combustion instability has rarely been discussed in the existing literature. In this study, the effects of time-varying grain shape are modeled by introducing a slow time scale associated with the progressive burning of the grain. The resulting model equation accounts for the evolution of acoustic modes as well as their growth/decay in amplitude. Predictions with and without the use of the slow scale are compared with respect to measurements of a static firing test.

9:15

4aPA3. Far-field acoustical measurements during a Space Launch System solid rocket motor static firing. Blaine M. Harker, Brent O. Reichman, Trevor A. Stout, Eric B. Whiting, Kent L. Gee, and Tracianne B. Neilsen (Dept. of Phys. and Astronomy, Brigham Young Univ., N283 ESC, Provo, UT 84602, blaineharker@byu.net)

Acoustical measurements were made in the very far field during a recent test firing of the five-segment QM-1 Space Launch System solid rocket motor at Orbital ATK. Data were taken using 6.35 mm and 12.7 mm type-1 microphones at three far-field locations to the sideline and aft of the nozzle at a range of 650–800 nozzle diameters. The experiment setup, including the appreciable terrain changes, is first discussed. Spectral and autocorrelation analyses highlight the variation of the noise with respect to observation angle. In addition, high-frequency spectral characteristics and waveform statistics are evidence of the significant nonlinear propagation over the propagation range. Effects of microphone size, terrain effects, and data stationarity during the firing are discussed. This dataset is compared to measurements of other solid rocket motors at closer and farther ranges, including the GEM-60 and the four-segment Shuttle Reusable Solid Rocket Motor.

9:35

4aPA4. Multiresolution non-stationary techniques for the characterization of transient pressure signals embedded in broadband plume noise. David Alvord and Alessio Medda (Aerosp. & Acoust. Technologies Div., Georgia Tech Res. Inst., 7220 Richardson Rd., Smyrna, GA 30080, david.alvord@gtri.gatech.edu)

The acoustic environment generated by the Space Shuttle propulsion system historically was considered one of the most complex rocket noise environments to predict of NASA's heritage rockets. After the Space Shuttle was retired in 2011, the Space Launch System (SLS) architecture chosen to succeed the shuttle will generate an even more complex acoustic environment than Space Shuttle. Both Space Shuttle and SLS have dual first stage propulsion systems consisting of two Solid Rocket Boosters (SRBs) and three (Space Shuttle) or four (SLS) RS-25 LH2/LOX engines. The additional complexity for the SLS acoustic environment arises due to the in-line propulsion configuration (as opposed to the offset configuration for Space Shuttle) firing both SRB and liquid engine plume into the same exhaust duct at liftoff. For Space Shuttle, the offset configuration meant each acoustic phenomena was separable and could be predicted and analyzed individually. For SLS, the in-line configuration means the IOP waves (one per SRB) are injected directly into the steady state plume meaning the events can no longer be assumed separable. This paper discusses data from a scaled down configuration and applies wavelet analysis in attempt to extract and characterize an injected transient blast wave from steady state plume noise.

9:55–10:15 Break

10:15

4aPA5. Spatiotemporal correlation of high-performance military aircraft jet noise. Blaine M. Harker, Tracianne B. Neilsen, Kent L. Gee (Dept. of Phys. and Astronomy, Brigham Young Univ., N283 ESC, Provo, UT 84602, blaineharker@byu.net), Alan T. Wall (Battlespace Acoust. Branch, Air Force Res. Lab., Wright-Patterson AFB, OH), and Michael M. James (Blue Ridge Res. and Consulting, LLC, Asheville, NC)

Correlation analyses of pressure measurements on a ground-based array of microphones of noise from a tethered high-performance military aircraft provide insights into the sound field variation with position and engine conditions which are fundamental in the continued development of more complete jet noise models. Time-scaled, single-point (auto)correlation functions confirm that to the side of the nozzle exit, the temporal correlation envelope decays very rapidly, whereas the envelope decays more slowly in the maximum radiation region and farther downstream. Two-point space-time (cross) correlation functions confirm that noise from a single engine operating at intermediate power is more similar to that from heated, laboratory-scale jets, whereas additional features seen at military power and afterburner are unique, including a feature which is likely related to the dual directivity lobe observed in the far field of military aircraft. A complementary coherence analysis provides estimates of spatial coherence lengths as a function of frequency and location. Field coherence lengths are utilized in analyzing coherence lengths of equivalent source distributions obtained from applying DAMAS-C to the ground-based array data. The cumulative results of these investigations provide a full-scale military jet noise benchmark that should be considered when evaluating laboratory-scale jet studies and simulations of jet noise.

10:35

4aPA6. Inclusion of a ground-reflecting plane in wavepacket modeling of military jet noise. Tracianne B. Neilsen, Kent L. Gee (Brigham Young Univ., N311 ESC, Provo, UT 84602, tbn@byu.edu), Michael M. James (Blue Ridge Res. & Consulting, Asheville, NC), and Blaine M. Harker (Brigham Young Univ., Provo, UT)

Equivalent source models for high-speed jet noise are intended to represent the acoustic properties of the turbulent mixing noise. A wavepacket ansatz previously applied to obtain equivalent sources from anechoic laboratory measurements is modified to include both a direct and image complex pressure representation of the source to model the presence of a ground-reflecting plane. This is important for modeling the sound field at maintainer and flight deck personnel positions because the interference effects caused by ground-reflected propagation paths significantly influence the sound levels and vary with frequency and location. The ability of the direct-plus-image wavepacket model to yield the interference patterns observed across large planes of data (2 m tall by 23 m long) measured near a high-performance military aircraft is evaluated. In particular, the variation in the nulls of the sound pressure level across the planes as a function of frequency predicted by this wavepacket model are compared to the measurements between 4 and 25 m from the engine nozzle exit. Comparisons with previous work based on a Rayleigh-distributed source amplitudes are also provided. [Work supported by the Office of Naval Research.]

Contributed Papers

10:55

4aPA7. Quantitative nonlinearity in subsonic and supersonic model-scale jet noise. Kyle G. Miller (Dept. of Phys. and Astronomy, Brigham Young Univ., 323 East 1910 South, Orem, UT 84058, kglenmiller@gmail.com), Brent O. Reichman, Kent L. Gee, Tracianne B. Neilsen (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT), and Anthony A. Atchley (Gradate Program in Acoust., Penn State Univ., Provo, UT)

Understanding the impact of jet noise, including annoyance due to crackle, can be improved by quantifying the nonlinearity in a signal with a single-microphone measurement. An ensemble-averaged, frequency-domain version of the generalized Burgers equation has been used to find a quantitative expression for the change in sound pressure level spectrum, L_p, with distance, r, due to the separate effects of geometric spreading, absorption, and nonlinearity. The nonlinear term, based on the dimensionless nonlinearity indicator known as "Q/S," has been used to characterize the frequency-dependent nonlinearity as a function of angle and distance in subsonic (Mach-0.85), overexpanded (Mach-1.8), and ideally expanded (Mach-2.0) model-scale jet data. Analyses show that nonlinear effects in the Mach-2.0 data are about twice as strong as those in the Mach-1.8 data, but such effects are completely absent in the Mach-0.85 data. [Work supported by the AFRL SBIR program.]

11:10

4aPA8. Comparison of measured and predicted statistical measures in military jet noise propagation. Brent O. Reichman (Brigham Young Univ., 453 E 1980 N, #B, Provo, UT 84604, brent.reichman@byu.edu), Alan T. Wall (Air Force Res. Labs, Wright-Patterson Air Force Base, OH), Kent L. Gee, and Tracianne B. Neilsen (Brigham Young Univ., Provo, UT)

Crackle, an annoying component of jet noise, has been associated with acoustic shocks that form as a waveform experiences nonlinear steepening. The skewness of the first time derivative of the pressure waveform, or derivative skewness, is a metric that is sensitive to acoustic shocks, and has shown that shocks, and therefore crackle, are found in the far field of full-scale military jets. Recent extensive measurements on F-35 aircraft enable a comparison of linear and nonlinear predictions and measured waveforms over a large spatial area. Waveforms measured at 76.2 m from the aircraft are numerically propagated using linear and nonlinear methods to several distances up to 305 m, and the derivative skewness calculations from these numerically propagated waveforms are compared with those of measured signals. Comparisons are made for both the A and B variants of the F-35, in changing meteorological conditions, and with engine conditions ranging from intermediate power to full afterburner. [Work supported by USAFRL through ORISE.]

11:25

4aPA9. Estimating aircraft noise levels and spectra from aircraft flyover and ground operations. Alan T. Wall and Richard L. McKinley (Battlespace Acoust. Branch, Air Force Res. Lab., Bldg. 441, Wright-Patterson AFB, OH 45433, alantwall@gmail.com)

The noise from military aircraft operations can adversely affect people within the civilian community near an airbase and within the military community living and operating within the confines of the airbase. The aircraft noise levels and spectra should be estimated to determine the building noise attenuation treatments to achieve the desired interior noise levels. This presentation describes a method to estimate the noise levels and spectra from both ground operations and flyover operations of high performance military jet aircraft for the community near the airbase and the airbase proper. Sound levels are predicted over a typical airbase region using an equivalent source model for a fighter jet aircraft and atmospheric propagation modeling. Overall level contours are generated over the entire area, and spectral levels are predicted at desired points corresponding to noise sensitive building locations. [Work supported by USAFRL through ORISE.]

Session 4aSC

Speech Communication: Development of Speech Production and Perception Across the Lifespan

Mary E. Beckman, Cochair
Linguistics, Ohio State University, 222 Oxley Hall, 1712 Neil Ave, Columbus, OH 43210-1298

Valerie Hazan, Cochair
Speech, Hearing and Phonetic Sciences, UCL, Chandler House, 2, Wakefield Street, London WC1N 1PF, United Kingdom

Invited Papers

8:50

4aSC1. Clear speech adaptations across the lifespan. Valerie Hazan (Speech, Hearing and Phonetic Sci., UCL, Chandler House, 2, Wakefield St., London WC1N 1PF, United Kingdom, v.hazan@ucl.ac.uk)

Talkers need to be able to alter their speech production in order to communicate effectively in conditions in which there are acoustic or linguistic barriers to communication. The clear speaking styles that result from such adaptations have been well documented, but mostly for young adult talkers. As a skilled aspect of speech production, the ability to make such adaptations may develop late during acquisition and be affected by reduced motor or cognitive control in older talkers. In a series of studies, we are investigating how speech adaptations in challenging communicative conditions change across the lifespan: in children aged 9 to 15 years, in young adults, in older adults aged 65 and above. In all three studies, recordings were made while talker pairs complete a cooperative problem-solving task (diapix) when communication was easy or when a communication barrier was placed on one talker. Acoustic analyses have shown adaptations to speaking rate, intensity, fundamental frequency characteristics, vowel formant space. While young adults seem adept at making adaptations that are well suited to overcome the specific interference experienced by their interlocutors, younger talkers appear to make less nuanced adaptations. We will discuss factors affecting such speech adaptations across the lifespan.

9:10

4aSC2. Speech perception and spoken word recognition in young children. Jan Edwards and Tristan Mahr (Commun. Sci. and Disord., Univ. of Wisconsin-Madison, Goodnight Hall, 1975 Willow Dr., Madison, WI 53706, jedwards2@wisc.edu)

Recent studies have found that expressive vocabulary size predicts spoken word recognition in young children. Children with larger vocabularies recognize even highly familiar words more efficiently than their peers with smaller vocabularies (Fernald *et al*., 2006). Furthermore, lexical processing efficiency at 18 months predicts future vocabulary size up to 8 years of age (Fernald *et al*., 2013; Marchman & Fernald, 2008). However, these studies did not include any measures of speech/language development beyond vocabulary size. We discuss the results of several studies that included multiple measures of speech and language development as well as measures of the home linguistic environment (using LENA, Ford *et al*., 2008). One experimental task assessed children's online responses to correct productions of familiar words, to mispronunciations of these words, and to nonwords using eye-tracking. We found that speech perception was a better predictor of lexical processing efficiency than expressive vocabulary size. Furthermore, the number of conversational turns between children and their caregivers also predicted how quickly children looked at unfamiliar objects when they heard nonwords. Implications of these results are discussed. [Research supported by NIDCD grant 02932.]

9:30

4aSC3. Speech perception and production in sequential bilingual children: A longitudinal study. Kathleen McCarthy (Speech, Hearing and Phonetic Sci., Univ. College London, 2 Wakefield St., London WC1N 1PF, United Kingdom, kathleen.mccarthy@ucl.ac.uk)

The majority of bilingual speech development research has focused on children who acquire both languages simultaneously from birth. Yet, for many bilingual children growing up in dense immigrant communities, their language experience is typically very different. These children, often referred to as sequential bilinguals, are initially exposed to their heritage language (L1) and in many cases are only immersed in the host country's language (L2) when they start school at around 4-years-old. To date, little is known about the developmental trajectory of sequential bilinguals. The current study tracked the acquisition of the English voicing contrast and monophthongal vowels by Sylheti-English sequential bilingual children from the London-Bengali community. Children were tested during preschool (mean age: 54 months old) and again one year later. The sequential bilinguals perception and production was initially driven by their L1 experience, resulting in less refined English phonemic categories than their monolingual peers. With L2 experience, children acquired the target phonemic contrasts, however they displayed a preference Sylheti phonotactics in a non-word repetition task. These findings have implications for our understanding of language development in complex multilingual settings, and will be discussed in light of the children's language input and broader phonological development.

9:50

4aSC4. Cognitive and linguistic influences on speech motor development. Ignatius Nip (School of Speech, Lang., & Hearing Sci., San Diego State Univ., 5500 Campanile Dr., San Diego, CA 92182-1518, inip@mail.sdsu.edu)

Speech development is affected by the interaction of the development of various domains, including motor control and language. Infants demonstrate strong associations between speed and range of movement of the lips and jaw with cognitive (attention, memory) and language (number of words understood/expressed, number of gestures used) skills during language acquisition (Nip *et al*., 2011). In addition, goal-directed vocal behaviors (e.g., words) are produced with faster lip and jaw speeds in infants (Nip *et al*., 2009). Similar task-related changes can be observed in older children. Tasks requiring greater language formulation demands (e.g., re-telling stories) are produced with greater movement speeds and oral excursions than tasks requiring fewer demands (e.g., repeating syllables) (Nip & Green, 2013). Kinematic descriptions of speech production may provide insight into the acoustic consequences of adapting movement strategies for different tasks; increasing oral excursions may be a strategy to increase articulatory precision and may account for the reduced speaking rate during speaking tasks requiring greater demands (Nip & Green, 2013). In addition, coordination of articulatory movements is highly associated with intelligibility in children with and without speech disorders (Nip, in press). Understanding the interactions among motor control and language may explain patterns of speech development.

10:10–10:25 Break

10:25

4aSC5. The influence of gender identity on children's production of sibilant fricatives. Benjamin Munson (Univ. of Minnesota, 115 Shevlin hall, 164 Pillsbury Dr., SE, Minneapolis, MN 55455, benjamin.ray.munson.jr@gmail.com)

Phonetic differences between men and women's speech are the consequence of sex-related variation in the anatomy of the speech-production system, as well as learned behaviors specific to particular languages and to specific social and cultural contexts. Children learn socially and culturally specific gendered phonetic variants concurrent with developmental changes in the vocal tract which give the child an increasing capacity to produce systematic phonetic variation. The research in this presentation is part of a larger project examining phonetic variation in 5–13 year old boys with gender dysphoria (GD). This presentation focuses on the acoustic characteristics of /s/, which was chosen because it is the locus of a great deal of gender-related phonetic variation that appears not to be the consequence of sex differences in vocal-tract size and shape. Preliminary analyses of a subset ($n = 30$) of talkers from a corpus of 5 to 13 year old children ($n = 104$, including boys with GD and boys and girls without GD), found that boys with GD produce more [θ]-like /s/ tokens than boys without GD in single words. This talk presents the results from the entire corpus, which allows for a more robust analysis of the influence of gender identity on children's fricative production.

10:45

4aSC6. Phonetic and lexical influences on changes across the lifespan. Jonathan Harrington and Johann U. Reubold (Inst. of Phonet. and Speech Processing (IPS), Univ. of Munich, Schellingstrasse 3, Munich 80799, Germany, jmh@phonetik.uni-muenchen.de)

The study considers longitudinal studies over several decades within the same individual in order to determine whether phonetic sound change takes place initially in more frequent words. The focus of the analysis was on the vowels of the Christmas Broadcasts by Queen Elisabeth II over several decades and in Alistair Cooke's Letter from America broadcasts. For the first of these, a re-analysis of the phonetic lowering of the vowel in the lexical set TRAP and of tensing of final lax vowel in HAPPY showed no effect of lexical frequency on sound change. The focus of analysis in the second was during a period in which the speaker was shown to acquire General American characteristics after emigrating to the United States from Britain, but then in later life to revert over a 5–10 year period back toward characteristics of British English Received Pronunciation. This reversion is shown to take place at a faster rate in lexically frequent words. While the evidence overall for lexically gradual changes is equivocal, the changes in both speakers are most appropriately modeled as a leveling due to variation in dialect contact: increasingly with middle class speakers for the Queen; decreasingly with American English speakers for Cooke.

11:05

4aSC7. Effects of age-related hearing loss on verbal processing and short-term memory. Esther Janse (Ctr. for Lang. Studies, Radboud Univ. Nijmegen, PO Box 310, Nijmegen 6500 AH, Netherlands, e.janse@let.ru.nl) and Elise D. Bree (Res. Inst. of Child Education and Development, Univ. of Amsterdam, Amsterdam, Netherlands)

Adult aging is frequently accompanied by hearing loss, as well as by cognitive decline. Both play a role in listeners' understanding of noisy, foreign-accented speech and fast conversational speech. Hearing loss leads to increased perceptual effort during listening, which affects memory encoding of the spoken message. Furthermore, some studies have suggested that acquired hearing loss also has long-term effects by degrading the quality or accessibility of phonological representations in long-term memory. This was investigated further by looking into non-word reading, thus bypassing immediate effects of hearing loss. A sample of 29 older adults, varying in degree of high-frequency hearing loss and visual digit span performance, saw 72 multisyllabic nonwords varying in phonotactic frequency (i.e., the phoneme-co-occurrence statistics of the language). They saw the nonwords for 5 seconds and were prompted to produce them from memory after another 3 seconds. As expected, response accuracy was influenced by phonotactic frequency of the nonword and digit span performance. Crucially, response accuracy was also higher if the participant had better hearing, supporting the claim that hearing loss degrades phonological representations in long-term memory. These results emphasize the broad consequences hearing loss has on language processing beyond its immediate effect on speech audibility.

11:25–11:45 Panel Discussion

Session 4aUW

Underwater Acoustics and Signal Processing in Acoustics: Environmental Variability Impact on Shallow Water Acoustics I

Brian T. Hefner, Cochair
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Anthony L. Bonomo, Cochair
Applied Research Laboratories, The University of Texas at Austin, 10000 Burnet Road, Austin, TX 78713

Chair's Introduction—8:00

Contributed Papers

8:05

4aUW1. A traditional forward scattering theory view of shallow water acoustics. Timothy F. Duda (Woods Hole Oceanographic Inst., WHOI AOPE Dept. MS 11, Woods Hole, MA 02543, tduda@whoi.edu)

Theories of propagation through a random medium (forward scattering theories) identify the scales of feature structure within the medium that are most effective at creating field fluctuations. These scales are near the Fresnel scale for straight-line and refracted propagation. The Fresnel scale (or Fresnel radius), the square root of the wavelength times the propagated distance, is therefore useful as a diagnostic. Shallow-water propagation of sound below 1 kHz often shows a dispersed modal propagation character, with modes having distinct wavelengths. Consideration of individual modes reduces the problem physical dimension by one. Waveguide anomaly features that are known to cause fluctuations, such as small-scale internal waves and frontal intrusions, have characteristic scales that overlap with the Fresnel scales for frequencies in the range of 0.1 to 1 kHz and distances of 1.0 to 25 km (40 to 600 m). This means that established theories can be applied, as long as other requirements are met. The waveguide thickness (water depth) can also exhibit variability on these scales, causing a similar fluctuation effect, although in this case the fluctuation represents a departure from acoustic behavior with a simple planar seabed model. This scaling analysis is potentially most useful for modeling shallow-water ocean propagation, or for processing ocean data to emphasize features relevant to acoustic scattering.

8:20

4aUW2. Forward and back scattering in a shallow inhomogeneous environment. Steven A. Stotts, David P. Knobles, and Robert A. Koch (Environ. Sci. Lab., Appl. Res. Labs/The Univ. of Texas at Austin, 10000 Burnet Rd, Austin, TX 78759, stotts@arlut.utexas.edu)

An experiment in 20 m of water off Panama City, FL, motivates the simulation with several propagation models of the time series produced from rough seabed scattering. Both the forward and backward scattered time series are simulated on vertical line arrays in the 2–4 kHz band to ranges out to 5 km. Results from several Born approximation approaches are compared to the results of a full numerical two-way coupled-mode solution to analyze the measured data. The results from a method of separating the forward and backward scattered amplitudes are also presented. Competing effects, such as the interplay between modal attenuation and the scattering of the higher order modes, are examined within the context of both the full coupled-mode and Born approximation solutions.

8:35

4aUW3. Time-domain Helmholtz-Kirchhoff integral for forward surface scattering in a refractive medium. Youngmin Choo, Heechun Song (Scripps Inst. of Oceanogr., 9263 Regents Rd., La Jolla, San Diego, CA 92037, ymchu@ucsd.edu), and Woojae Seong (Seoul National Univ., Seoul, South Korea)

Time-domain Helmholtz-Kirchhoff integral (H-K integral) for forward surface scattering is extended to a refractive medium. Ray theory is applied to obtain the Green's function, while the normal derivative in the integral is evaluated analytically using a ray geometry along with the ray-based Green's function. This approach allows for stationary phase approximation which reduces the surface integration to a line integration. For high-frequency signals with a narrow bandwidth, an asymptotic form of the time-domain H-K integral then can be derived using the Fourier transform. The pressure field scattered from a sinusoidal surface wave in a constant-gradient sound speed profile is evaluated and compared to the result based on a conventional ray model.

8:50

4aUW4. A modeling approach to scattering and reverberation in shallow water. Anatoliy Ivakin (Appl. Phys. Lab, Univ. of Washington, 1013 NE 40th, Seattle, WA 98105, aniv@uw.edu)

A physics-based modeling approach is described that allows prediction of reverberation in complex shallow water environments. An integral expression is presented for the backscatter intensity with a factorized integrand comprised of two kernels, the two-way propagator and the scattering kernel. The propagator is defined by Green's function, describes the local intensity, and can be calculated using available models, such as PE, normal modes, or ray approximations. An expression for the scattering kernel is obtained using a unified approach to volume and roughness scattering [Ivakin, JASA Feb. 1998], and represents a sum of volume and roughness scattering coefficients. They are specified for sea-water column and rough heterogeneous seabed with continuous spatial fluctuations of compressibility and density, and/or discrete randomly distributed targets, such as bubbles, fish, shells, and others. Results of numerical simulations for shallow water reverberation time/range series, based on a PE propagation model, are presented, and potential contributions of different mechanisms of scattering are compared and discussed. The approach is applied to consider reverberation in a complex shelly sand/mud environment, such as one at the Target and Reverberation Experiment 2013 (TREX), and results in model/data comparisons based on analysis of TREX acoustic scattering data and environmental ground truth measurements. [Work supported by ONR-OA.]

9:05

4aUW5. Simulating the lateral line with low-frequency nearfield acoustic holography based on a vector hydrophone array for short-range navigation in littoral waters. Tim Ziemer (Inst. of Systematic Musicology, Univ. of Hamburg, Neue Rabenstr. 13, Hamburg 20354, Germany, tim.ziemer@uni-hamburg.de)

Fish use the lateral line system to detect swimming objects within a range of a few body lengths to avoid collision. This is achieved by detecting particle velocity and acceleration with receptors which are distributed all over the skin. In this work, this principle is imitated by means of a hydrophone array detecting particle accelerations. In a two-dimensional simulation setup, a dipole source is detected in presence of high-level noise and a disturbing source. This is achieved by a hull mounted vector hydrophone array applying low-frequency nearfield acoustic holography and an adaption of minimum energy method which simulate the lateral line. Noise is added to simulate decorrelated signals—like ambient noise and reverberation—whereas a disturbing dipole source creates highly correlated disturbing signals at the hydrophone positions which is typical for early reflections and sound sources outside the detection area. At the hydrophone array the noise is up to almost 70 dB louder than the radiated source signal. Despite these difficult conditions the proposed method localizes the source reliably within the range of a few array lengths. The system could be implemented in vessels for short-range navigation in coastal and littoral areas or underwater vehicles for mine deployment and harbor construction works.

9:20

4aUW6. Model/data comparisons of high-frequency backscattering from well-characterized sand sediment. Brian T. Hefner (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, hefner@apl.washington.edu)

In the Spring of 2010, acoustic backscattering from an artificially-smoothed sand sediment was measured in the NSWC PCD test pond. The measurements were made at frequencies from 200 to 500 kHz as a function of grazing angle. The residual roughness of the smoothed surface was measured using a laser line scanner while the sediment parameters were determined through diver core analysis and sound speed and attenuation measurements. At the time of the experiment, it was not possible to characterize the volume heterogeneities within the sediment. Subsequent analysis of historic conductivity probe measurements collected in the same test pond under similar conditions has been used to fill this gap. With this additional dataset, the environmental characterization of the sediment places significant constraints on models of scattering from both the sediment roughness and the subsurface volume heterogeneities. Using the sediment parameters, the predictions of small perturbation roughness and volume scattering theories are compared to the data when the sediment is modeled as a fluid and as a poroelastic medium. The effect of the measured porosity fluctuations on sound propagation within the sediment is also examined using a recently developed scattering loss model. [Work supported by ONR.]

9:35

4aUW7. Reverberation modeling approximately accounting for three-dimensional forward scattering effects. Eric I. Thorsos, Jie Yang, and Frank S. Henyey (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, eit@apl.washington.edu)

A transport theory approach has been developed for modeling shallow water propagation and reverberation at mid-frequencies with emphasis at 1–3 kHz. With this approach, sea surface forward scattering can be taken into account in a 2-D (range-depth) approximation. While the effects of surface forward scattering on transmission loss are found to be modest (1–2 dB), the corresponding effects on reverberation level for typical conditions can be significant (~10 dB), even though bottom backscatter dominates reverberation in shallow water. An attempt to make data/model comparisons brought out the issue of the proper way of using the 2-D (in horizontal plane) surface roughness spectrum to model forward scattering with a 2-D (range-depth) propagation model. For reverberation modeling, we are investigating the approximation of preserving the distribution in vertical scattering angles in reducing the full 3-D problem to a 2-D (range-depth) problem when using the traditional N ’ 2-D approach. A description of the method will be given, followed by data-model comparisons for TREX-13. [Work supported by ONR Ocean Acoustics.]

9:50

4aUW8. Coherent, very high frequency underwater acoustic communications under wind-driven seas: Experiments in an ocean simulator. James Preisig (JPAnalytics LLC, 638 Brick Kiln Rd., Falmouth, MA 02540, jpreisig@jpanalytics.com) and Grant Deane (Scripps Inst. of Oceanogr., La Jolla, CA)

Very high frequency, underwater water acoustic communications (VHF UWAC) operated near the sea surface are subject to interference from surface-scattered energy and, under wind driven seas, noise from breaking waves. Here, we present results from experiments to study these effects conducted in an ocean simulator. Acoustic transmissions in the frequency range (400–750) kHz were made using vertical arrays of transducers in the 44 m long, 2.4 m wide, and 1.5 m deep wind-wave channel at the Hydraulics Laboratory at Scripps Institution of Oceanography. This facility simulates oceanic conditions with a piston wave maker and a fan driving airflows up to 15 m/s. A variety of communications and probe signals were sent under calm and wind-driven conditions. Surface scattering was characterized in terms of arrival delay and gravity wave focusing, along with measurements of breaking wave noise. These data will be discussed in terms of VHF UWAC performance.

10:05–10:20 Break

Invited Papers

10:20

4aUW9. Environmental factors that contribute to high frequency bottom loss variability. Jacob George, David W. Harvey, Allen Lowrie (NP53, NAVOCEANO, 1002 Balch Blvd., Stennis Space Ctr., MS 39522, jacob.george1@navy.mil), and Lori S. Conner (NP64, NAVOCEANO, Stennis Space Ctr., MS)

We discuss three factors: short range sediment variability, bio-attenuation, and seafloor roughness. In a previous presentation (ASA 2013), we have shown that statistical distributions of bottom loss values derived from measured transmission loss (TL) are nearly invariant to measured sediment properties such as sound speed, density, and porosity. To test if this surprising result was caused by under-sampling of sediment cores, three TL runs were done in 2014 when cores were collected at 1 km intervals. The results of these showing short range variations will be discussed. Among the stations in our HFBL database we have found possible evidence for bio-attenuation due to fish (anchovies) swim bladder resonance (Diachok-Wales, JASA 2005). Finite element modeling of a TL station has shown that seafloor roughness can explain the observed frequency dependence of HFBL values (Isakson, report to NAVO). All these will be discussed.

10:40

4aUW10. Finite element modeling of propagation and reverberation shallow water waveguide with a variable environment. Marcia J. Isakson, Nicholas P. Chotiros, and James Piper (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78713, misakson@arlut.utexas.edu)

Shallow water waveguides can exhibit environmental variability in the water column, on the bottom interface and in the sediment. Finite element models provide a method to capture the effects of this variability since every element can be described by a different sound speed and density. However, fully three-dimensional finite element models are often computationally inaccessible due to extreme memory requirements. In this study, a longitudinally invariant finite element model is used to predict the reverberation from a shallow water waveguide described by environmental measurements at the Target and Reverberation Experiment 2013 conducted off the coast of the Florida panhandle. Longitudinally invariant models retain all of the fidelity of a three-dimensional model with the requirement that one geometric dimension must be invariant. Therefore, it is an ideal model for wedges and ridges. In this case, the longitudinally invariant direction describes the sand ripples. The reverberation with and without variations in sediment sound speed and density will be compared for the same bathymetry to determine the role of sediment variability in reverberation. Reverberation from along and across the sand ridges will also be examined. [Work supported by ONR, Ocean Acoustics.]

11:00

4aUW11. Initial considerations on effects of braided river beds on long-range acoustic propagation in shallow water. Charles W. Holland (Appl. Res. Lab., The Penn State Univ., P.O. Box 30, State College, PA 16804, cwh10@psu.edu), Allen Lowrie, and Rhett Hamiter (Naval Oceanographic Office, Stennis Space Ctr., MS)

A braided river consists of multiple small channels that divide and recombine numerous times. They form when the sediment load and transporting energies are such that coarser sediment can be deposited as shifting islands or bars between the channels. Braided rivers formed on continental shelves during sea level lowstands during the last 700,000 years when the shelf was sub-aerially exposed to greater than 125 m below present day sea level. A salient feature is that the stream bed (which can be many kilometers wide) is expected to exhibit extremely high geoacoustic variability, given that the islands and bars exhibit much coarser material than that in the channels. The variability is expected both laterally (highest variability or smallest scales perpendicular to the stream flow) and also vertically inasmuch as the river channels experience different avulsion patterns over time. Chirp sonar and core data provide some insight into the underlying geologic processes and the associated scales of variability. Modeling gives some initial clues about the effects of braided river beds on acoustic propagation. [Work supported by the Naval Oceanographic Office and the Office of Naval Research, Ocean Acoustics program.]

11:20

4aUW12. Model-data comparisons of range-dependent shallow-water reverberation including both boundary and volume scattering. Dale D. Ellis (Phys. Dept., Mount Allison Univ., 18 Hugh Allen Dr., Dartmouth, NS B2W 2K8, Canada, daledellis@gmail.com) and Sean Pecknold (Atlantic Res. Ctr., Defence Res. and Development Canada, Dartmouth, NS, Canada)

A range-dependent shallow-water reverberation model using adiabatic normal modes has been previously developed [Ellis *et al.*, ISURC Conference, La Spezia, 2008] to handle bottom scattering and clutter echoes in a range-dependent environment. It has now been extended to handle volume scattering from the water column and volume scattering from the sub-bottom. Beam time series similar to that from a horizontal line array can be produced. Comparisons can then be made directly with data, and area scattering maps created. Of particular interest will be data obtained on the triplet line array during the 2013 TREX experiment in the Gulf of Mexico off Panama City, Florida. Predictions will be compared with data for average reverberation from a number of pings, as well as ping-to-ping variability on a particular run. [Work supported in part by U.S. Office of Naval Research, Code 32.]

Session 4pAAa

Architectural Acoustics and Noise: Acoustic Comfort in Building Indoor Environmental Quality (IEQ) Performance II

Kenneth P. Roy, Cochair
Building Products Technology Lab, Armstrong World Industries, 2500 Columbia Ave, Lancaster, PA 17603

Donna A. Ellis, Cochair
The Division of Architecture and Engineering, The Social Security Administration, 415 Riggs Ave., Severna Park, MD 21146

Invited Papers

1:00

4pAAa1. Annoyance perception of complex multi-tone noise signals in both harmonic and inharmonic structures within the built environment. Joonhee Lee and Lily M. Wang (Durham School of Architectural Eng. & Construction, Univ. of Nebraska - Lincoln, 1110 S. 67th St., Omaha, NE 68182-0816, joonhee.lee@huskers.unl.edu)

Assorted building mechanical systems generate tonal components within the background noise of built environments. In most cases, this type of noise includes multiple tones in harmonic or inharmonic structures rather than a single tone. However, there is limited information on the comprehensive annoyance caused by multiple tones as perceived by human occupants. Two current standards, ISO 1996-2 and ANSI S1.13, propose calculation methods to address tones in noise, but those methods only analyze the tones individually. This paper aims to investigate how each tone contributes to overall annoyance perception when complex tones are present in background noise. Noise stimuli with five-tone complexes between 125 Hz to 2 kHz were artificially generated for subjective testing. The levels of each tone were randomly adjusted for every trial, and both harmonic and inharmonic structured tone complexes were utilized. Ten musically trained subjects participated in the subjective test involving paired comparisons. Each participant was asked to choose which noise stimulus is more annoying between two noise signals. Perceptual weighting analysis is applied to the results to compute a spectral weighting function for overall annoyance. The performance of the derived spectral weighting function is examined against annoyance ratings of actual building mechanical noises.

1:20

4pAAa2. Innovative approaches to structure borne vibration reduction. Bonnie Schnitta (SoundSense, LLC, 46 Newtown Ln., Ste. One, East Hampton, NY 11937, bonnie@soundsense.com)

People feel vibrations depending mostly on the vibration direction. One has to distinguish between vertical vibration and horizontal vibration, the latter often called lateral vibration, in order to address the homeowners' concerns. The difference in perception is described in ISO 8041:2005. Above 20 Hz and up to 1000 Hz, vibration may be called structure-borne noise. Such vibrations may generate audible sound. In private or multi-family residence, structure-borne noise can become prevalent due to elevators, mechanical equipment, subways, nearby train tracks, etc. Often times, areas exhibiting these problems do not have the space in floor or wall configurations in order to take a traditional approach toward the abatement of this structure-borne noise. The purpose of this paper is to present the results from a controlled experiment of different combinations of floor and wall configurations under both horizontal and lateral vibration in order to simulate vibration found in the field due causing structure-borne noise. Floor and/or wall configurations will include combinations of various resilient layers in order to present configurations to successfully abate vibrations originating from these sources while conserving and minimizing space in floor and/or wall configurations to optimize the acoustic efficacy of the installation.

1:40

4pAAa3. Acoustic comfort in closed rooms often means expectations of speech privacy. Kenneth W. Good and Kenneth P. Roy (Armstrong, 2500 Columbia Ave., Lancaster, PA 17601, kwgoodjr@armstrong.com)

The design of privacy areas is very similar in offices, healthcare, and other building segments, but the focus is squarely on healthcare due to the HIPAA regulations involving personal medical information. Given the degree of enclosure provided by the walls, ceiling, doors, etc., common expectations by occupants of those closed spaces is for confidential speech privacy. So, how do we design and evaluate the speech privacy provided by medical offices, treatment rooms, conference rooms, etc. ASTM laboratory testing of components and systems including STC, CAC, and NRC are part of the answer. But the bottom line is in the field with actual performance testing of the systems as they are built and operated. Two questions to be answered are how do we make informed decisions about product and system choices, and how do we maximize speech privacy using those architectural choices.

Contributed Papers

2:00

4pAAa4. Acoustic comfort in living environment. Made Samantha Wiratha and Lucky Tsaih (Architecture, National Taiwan Univ. of Sci. and Technol., RB 807, No.43, Sec. 4, Keelung Rd., Da'an Dist., Taipei City 10607, Taiwan, sasawiratha@gmail.com)

As population of aging people growing fast, the need of long-term care facility is also increased. In the living environment of elder people, acoustic comfort has the same importance as thermal comfort and visual comfort. This research is to study the preference of acoustic comfort through the listening evaluation with normal hearing people. The listening evaluation is based on 20 live recorded sound samples from two Taiwanese long term care facilities and a university dormitory. The listening evaluation was participated by 66 architecture students. A semantic differential scale questionnaire with 11 pairs of sound qualities for each sound sample were used. The results shown that normal hearing people have negative impression for the current acoustic comfort condition of these Taiwanese long term care facilities. All of the respondents gave negative impression to water pump machine sound due to the "noisy" and "harsh" sound qualities. A live percussive music rehearsed event was associated with "noisy" and "agitating" impression by 97% of the respondents. Speech, TV, telephone ringing, snoring, and footsteps were also identified as discomfort sound quality. Bird chirping and quite dormitory room were recognized by 91% of the respondents as the most comfortable sound with the "natural" and "quiet" impression.

2:15

4pAAa5. Evaluation of a real-time convolution system for perception of self-generated speech in simulated rooms. Jennifer K. Whiting, Timothy W. Leishman, Nathan G. Eyring, Mark L. Berardi, and Michael K. Rollins (Dept. of Phys. and Astronomy, Brigham Young Univ., C110 ESC, Provo, UT 84606, lundjenny@comcast.net)

A real-time convolution system has been developed to quickly manipulate the auditory experiences of human subjects. The system is used to study perceptions of self-generated speech and music, and responses of talkers and musicians to varying acoustical conditions. It allows talkers in an anechoic environment to experience simulated room responses excited by their own voices. While their direct sound travels directly to their ears, they hear convolved room responses via specialized headphones spaced away from their heads. This presentation discusses the system's development, as well as its objective and subjective validations. Several existing rooms were modeled using EASE. Oral-binaural room impulse responses (OBRIRs) from these models were generated and implemented with the convolution system. Binaural recordings and measurements from the rooms were also made using a G.R.A.S. KEMAR mannequin. Objective comparisons of the OBRIRs from the measurements and simulations were explored in the investigation. Subjective evaluations of auralizations made from the OBRIR measurements and simulations, and binaural recordings, followed from A/B listening and speaking tests. In the latter, participants spoke in the various simulated acoustical environments and compared and rated the effects of each experience.

2:30

4pAAa6. Effects of noise flanking paths on ceiling attenuation class ratings of ceiling systems and inter-room speech privacy. Gary Madaras (ROCKFON, 4849 S. Austin Ave., Chicago, IL 60638, gary.madaras@rockfon.com) and Andrew E. Heuer (NGC Testing Services, Buffalo, NY)

Continuous plenums above suspended, modular ceilings and partial-height walls in buildings can result in inter-room speech privacy and annoyance problems, especially when noise flanking paths via air diffusers, grilles, and lights exist. However, testing of the effects of ceiling system noise flanking paths is limited in the industry. Multiple ceiling systems comprised of various noise flanking paths through air diffusers, grilles, and lights were tested in an independent, accredited, acoustics laboratory according to ASTM International (ASTM) E-1414 and E-413. Additionally, recorded speech was played back in the test chamber source room and binaurally recorded in the test chamber receiver room. The results show that wideband ceiling attenuation class (CAC) decreases by 10 decibels (dB) and 1/3 octave band normalized ceiling attenuation (Dn,c) decreases by 15 to 22 dB in the higher frequency bands when common noise flanking paths are introduced into a ceiling system with CAC-37 ceiling panels. Subjective listening during the course of these tests shows that a ceiling system comprised of CAC-37 panels and typical noise flanking paths (that drop the system rating down to CAC-27) did not provide speech privacy. Intelligibility of recorded speech transmitting into the receiver room was high.

2:45

4pAAa7. A case study investigation of the indoor environmental noise in four urban South African hospitals. Coralie A. van Reenen (Built Environment, Council for Sci. and Industrial Res., PO Box 395, Pretoira, Gauteng 0001, South Africa, cvreenen@csir.co.za)

This multiple case study was designed to investigate acoustics in multi-bed general wards in four South African hospitals. Evidence-based research shows that a quiet indoor environment has positive outcomes for hospital patients and staff. Though international guidelines define noise limits in hospitals, numerous studies world-wide reveal that few hospitals, if any, comply. The goal of this research was to determine whether hospital design paradigms in South Africa should be changed to improve the acoustic environment based on the findings of an acoustic assessment. The acoustic conditions in wards were assessed in terms of sound levels, user opinions, and architecture. The objectives were to determine whether the sites comply with guidelines, to determine user perceptions of noise, and to determine whether design factors influence the noise. It was found that the average sound levels exceeded the guidelines, yet the overall user perception was that noise is not disturbing. Layout and workflow have a likely influence on noise, requiring further research with particular reference to the difference found between patient and staff perceptions of sound. Further discussion pertains to the interpretation, application, and relevance of noise guidelines in an operational hospital environment, recommending further extensive research of human responses to noise exposure.

Session 4pAAb

Architectural Acoustics: Architectural Acoustics Potpourri

Ana M. Jaramillo, Chair
Ahnert Feistel Media Group, 8717 Humboldt ave. N, Brooklyn Park, MN 55444

Contributed Papers

3:20

4pAAb1. The status of classroom acoustics in Colombia. Ana M. Jaramillo (AFMG Services North America, LLC, 8717 Humboldt ave. N, Brooklyn Park, MN 55444, ana.jaramillo@afmg.eu), Bruce C. Olson, and Milton A. Salcedo (Olson Sound Design, LLC, Brooklyn Park, MN)

With the lack of standards to regulate classrooms in Colombia, we decided to find out what the current status was in terms of acoustical comfort. Due to the mild weather, classrooms usually do not have mechanical heating or cooling. For this reason, noise levels are mandated by proximity to busy roads or other noise sources. Their construction is typically done using heavy materials such as brick and concrete, which results in very low noise transmission, but Reverberation Times tend to be higher than recommended due to the little use of absorptive materials. Measurements and visual inspections were performed in several schools to evaluate these parameters against the current S12.60 ANSI standard.

3:35

4pAAb2. Acoustic response of a multipurpose auditorium at Grand Valley State University. Bailey Groendyke and Karen Gipson (Phys. and Music, Grand Valley State Univ., Allendale, MI 49401, groendba@mail.gvsu.edu)

Like many multipurpose auditoriums, the Louis Armstrong Theatre (LAT) at Grand Valley State University has been reported by a considerable number of students and faculty to have unsatisfactory acoustics for music performance. This study focused on physical measurements and simulated changes to LAT. Reverberation time (RT) was measured by filling LAT with sound and measuring the decay for select frequencies as per ASTM E2235 protocol, and the initial time delay gap (ITDG) was determined using slapsticks as an impulsive sound source. A model of LAT was also constructed from blueprints and physical measurements; simulations using this model were conducted using Odeon. Data from the physical measurements as well as the simulation confirmed that the RTs over a wide range of frequencies were smaller than desired for music, whereas ITDG measurements showed prevalent spurious reflections. Modifications to the model were made to increase reverberation time and reduce undesirable reflections in order to improve LAT for musical performance without compromising its functionality for speech.

3:50

4pAAb3. Acoustical characterization of touristic caves in Portugal. Antonio P. Carvalho and Joana I. Sousa (Lab. of Acoust., Univ. of Porto, FEUP (NIF501413197), R. Dr. Roberto Frias, Porto 4200-465, Portugal, carvalho@fe.up.pt)

Since the Palaeolithic, Mankind has been taking advantage of the acoustical characteristics of natural caves to perform its rituals. Nowadays, many of these spaces are used as touristic attractions or even as stages for musical performances. This study characterizes three touristic caves in Portugal where *in situ* measurements were done, of background noise sound pressure levels, RASTI, and Reverberation Time. The average RT values were 1.3 s to 1.7 s, and the RASTI average values revealed good intelligibility (from 0.50 to 0.57). The sound absorption coefficient of the stone that constitutes the interior of those caves was also measured in a standing wave apparatus.

4:05

4pAAb4. Acoustical study of historical large room for a contemporary use: The oratory of Albergo dei Poveri in Genova. Anna Chiari (DSA, Università Degli Studi di Genova, Genova, Italy), Ilaria Pittaluga, Corrado Schenone, and Davide Borelli (DIME - Sez. TEC, Università Degli Studi di Genova, Via all'Opera Pia 15/A, Genova, Italy, corrado.schenone@unige.it)

The conversion of deconsecrated religious buildings in conference halls or music auditoria is nowadays frequent, especially throughout Italy and Europe, although in most cases these rooms are under artistic and architectural preservation due to their historical heritage value. In this way, acoustic control interventions in such spaces, generally characterized by high reverberation time values, tend to be highly complex. The Oratory of "Albergo dei Poveri" building in Genova, transformed during the 90s in the Auditorium of the Departments of Law and Political Sciences of the University of Genova, faced these needs. After several interventions regarding the installation of sound systems and acoustical curtains, nowadays the acoustical features of the room are not still sufficient for the achievement of the adequate acoustic comfort. In this paper, the acoustic analysis of the hall has been evaluated and effective correction interventions at the same time compliant and non-invasive with the architectural constraints have been designed and investigated. A numerical model of the room was implemented through computer simulations and the model was validated by *in situ* measurement campaigns. Then, the numerical model was used to develop the subsequent design-scenarios calculations.

4:20

4pAAb5. A study on the analysis of hanok (the Korean traditional houses) acoustic characteristics. Won-Hee Lee, Myung-Sook Kim, and Myung-Jin Bae (Hyungnam Eng. Buliding 1212, Soongsil Univ., 369 Sangdo-Ro, Dongjak-Gu, Seoul, Korea, kimm@ssu.ac.kr)

Hanok is a type of Korean traditional houses, which is built based on science. In order to build Hanok, necessary materials are wood, rock, and clay, which can be easily acquired in nature. Due to the natural building materials, it is known that hanok is efficient to keep it warm and cool. Its heating system, Ondol, uses thermal conduction and convection of heat from the fireplace. Doors and windows are located on south and east sides because cold wind comes from northwest side in winter and cool breeze blows from southeast side in summer. On the other hand, the genre of music that goes well with the Hanok is the unique melody of Gukak (Korean classic music). The floors and walls of the Hanok, the rafters, highlights Gukak's uniqueness and plays the role of a sound box. This study was conducted to analyze the harmony between Hanok features and Gukak's acoustic characteristics.

4:35

4pAAb6. Assessing the range of spatial impression metrics from varying source position across a stage in assorted performing arts venues. Sungbeen Cho and Lily M. Wang (Durham School of Architectural Eng. and Construction, Univ. of Nebraska-Lincoln, 1110 S. 67th St., Omaha, NE 68182-0816, sungbeen@huskers.unl.edu)

A listener's perception of sound in the built environment is determined by various physical factors such as the size and form of a space, and shape and finishing of materials. Spatial impression is a term used to describe the spatial perception of the auditory field by listeners, and a number of metrics have been proposed to quantify this, including Interaural Cross-Correlation Coefficient (IACC) and Lateral Energy Fraction (LEF). While values for these metrics have been presented for a number of performing arts venues, few investigations have studied the range of these metrics within a specific venue, particularly as a source moves across the stage. This paper assesses data from both computer-simulated and measured rooms on how IACC, LEF, and Interaural Level Differences (ILD) vary within a venue, due to early reflections, as the source location varies across the stage. How greatly do the ranges of these metrics differ with source location? The range and values of these spatial impression metrics due to source movement may be another way in which a venue's spatial impression can be quantified and compared, correlating to how a listener at a fixed location spatially perceives sound sources within that room.

THURSDAY AFTERNOON, 5 NOVEMBER 2015 CITY TERRACE 9, 1:00 P.M. TO 5:35 P.M.

Session 4pAB

Animal Bioacoustics and Acoustical Oceanography: Bioacoustics Research in Latin America

Juliana R. Moron, Cochair
Instituto de Ciências Biológicas, Universidade Federal de Juiz de Fora, Rua Batista de Oliveira 1110 apto 404 B, Juiz de Fora 36010520, Brazil

Marie Trone, Cochair
Math and Science, Valencia College, 1800 Denn John Lane, Kissimee, FL 34744

Chair's Introduction—1:00

Invited Papers

1:05

4pAB1. New perspectives in Amazonian bird vocal behavior. Maria Luisa Silva (Instituto de Ciências Biológicas, Universidade Federal do Pará, Rua Augusto Correa, 1, Belém, Pará 66075150, Brazil, silva.malu@uol.com.br)

Neotropical region are known for its huge avian diversity and for the dominance of Sub-oscine or Tyranni passerines species. These species diversity are reflected in vocal communication and we will present here two examples of Brazilian Amazon forest of Tyranni species. Although the Tyranni are known for presenting a stereotyped song compared to Oscine passerines, we have found a species with a complex calls repertoire and other with a distinguished individual variation. The study area is the Ecological Park of Gunma, Santa Bárbara, Pará, Brazil, 50 km north of Belém. We performed playback experiments to define the vocalizations of Rusty-margined flycatcher *Myiozetetes cayanensis*, species that presents a complex repertoire of 15 different vocalizations, including a complex duet song. We also studied *Lipaugus vociferans,* a lekking bird, in which males perform conspicuous vocalizations in aggregations. We have analyzed songs of 19 individuals from six leks, considering the physical parameters (frequency bandwidth and temporal parameters) of the species-specific song. The results showed that it is possible to differentiate the individuals by song. These differences can represent an important function in individual recognition inside or between leks. These results show the necessity to investigate the high diversity of behavior in tropical habitats.

1:25

4pAB2. Does vocal learning accelerate acoustic diversification? Evolution of contact calls in Neotropical parrots. Marcelo Araya-Salas, Angela Medina-Garcia, and Timotthy Wright (Biology, New Mexico State Univ., 625 E University Ave., Apt. 26, Las Cruces, NM 88005, maraya@nmsu.edu)

Learning has been traditionally though to accelerate the evolutionary change of behavioral traits. We evaluated the evolutionary rate of learned vocalizations and the interplay of morphology and ecology in the evolution of these signals. We examined contact calls of 51 species of Neotropical parrots of the tribe Arini from recordings obtained in Central and South America. Parrots are ideal subjects due to their wide range of body size and habitats and their open-ended vocal learning that allows them to modify their calls throughout life. We estimated the evolutionary rate of acoustic parameters of parrot contact calls and directly compared them to those of morphological traits and habitat. We also evaluated the effect of body mass, bill length and vegetation density on acoustic parameters of contact calls while controlling for phylogeny. Evolutionary rates of acoustic parameters did not differ from those of our predictor variables except for spectral entropy, which had a significantly slower rate of evolution. We found support for correlated evolution of call duration, and fundamental and peak frequencies with body mass; and of fundamental frequency with bill length. We demonstrate that parrot contact calls, which are learned acoustic signals, show similar evolutionary rates to morphological traits. This is the first study to our knowledge to provide evidence that change through cultural evolution does not necessarily accelerate the evolutionary rate of traits acquired through life-long learning.

1:45

4pAB3. Transmission properties of vocalizations in a year-round territorial bird. Luis Sandoval (Escuela de Biologia, Universidad de Costa Rica, Escuela de Biologia, Universidad de Costa Rica, Montes de Oca, San Jose 11501-2060, Costa Rica, biosandoval@hotmail.com)

Acoustic Adaptation Hypothesis (AAH) predicts that acoustic signals using in long-distance communication should be optimized for transmission through its natural environment. To test if White-eared Ground-sparrows (*Melozone leucotis*) vocalizations are adapted to transmit long distances, I conducted two sound transmission experiments where I broadcast and re-recorded different calls, songs, and duets. This ground-sparrows use vocal communication year round for territory defense and mate attraction. I conducted the experiments inside ground-sparrows territories in Costa Rica, broadcasting natural vocalizations at different combination of distances, speaker and microphone heights to quantified the signal-to-noise ratio, tail-to-signal ratio, blur ratio, and excess attenuation. Songs and duets of White-eared Ground-sparrows showed similar patterns of degradation with distance and with proximity to the ground, suggesting that vocalizations facilitate communication with receivers at similar shorter distances (in less than a typical territory's diameter). *Chip* calls showed higher degradation in comparison to *tseet* calls with the distance, suggesting that *tseet* calls are design for longer distance communication. To my surprise, *chip* calls, songs, and duets has not experienced strong selection for long distance communication, because results do not support the AHH, and probably these vocalization characteristics are under other selective forces as sexual selection or phylogenetic constrain.

2:05

4pAB4. Studying dolphin whistles in Mexico. Carmen Bazúa-Durán, Julieta E. Sarmiento-Ponce, Brenda P. González-Leal, and Elena Montejano-Zea (Física, UNAM, Facultad de Ciencias, Circuito Exterior s/n, Ciudad Universitaria, México, D.F. 04510, Mexico, bazua@unam.mx)

Dolphin whistles are emitted especially during social interactions and feeding activities involving group cohesion, individual recognition, and recruitment. Here, at the Faculty of Sciences of UNAM we are studying dolphin whistles, mainly those of wild and captive bottlenose dolphins, *Tursiops truncatus*, to learn about their social structure and how whistles may be used to study dolphin ecology. We have developed a new methodology to describe and compare the whistle repertoire, which consists of whistle contour extraction to classify whistles into whistle types (using Matlab BELUGA and ArtWARP), then classifying whistle types into four general categories (high complexity, low complexity, linear long, and linear short), and finally computing a complexity index and a proportional variability of the whistle repertoire. Results obtained showed that this very simple method is useful to describe the whistle repertoire and to compare it according to the general behavioral state of dolphins, and between species. It is necessary to implement new methodologies like this one to better understand how dolphins are using whistles, since acoustic communication is the most important sense in dolphin species. [Work supported by PAPIIT-UNAM.]

2:25

4pAB5. Cetacean acoustic survey using towed array in the western South Atlantic shelf break. Artur Andriolo, Franciele R. Castro, Thiago Amorim (Laboratório de Ecologia Comportamental e Bioacústica, Departamento de Zoologia, Universidade Federal de Juiz de Fora, ICB, Rua José Lourenço Kelmer, S/n - Martelos, Juiz de Fora 36036-330, Brazil, artur.andriolo@ufjf.edu.br), Eduardo R. Secchi, Juliana Di Tulio (Instituto de Oceanografia, Universidade Federal de Rio Grande, Rio Grande, Brazil), Juliana Moron, Gabriela Ramos, Bruna Ribeiro (Laboratório de Ecologia Comportamental e Bioacústica, Departamento de Zoologia, Universidade Federal de Juiz de Fora, Juiz de Fora, Brazil), Alexandre N. Zerbini (National Marine Mammal Lab., Alaska Fisheries Sci. Ctr., NOAA, Seattle, WA), Luciano Dalla Rosa (Instituto de Oceanografia, Universidade Federal de Rio Grande, Rio Grande, Brazil), Raíssa R. Mendes, and Fábio B. Palácio (Laboratório de Ecologia Comportamental e Bioacústica, Departamento de Zoologia, Universidade Federal de Juiz de Fora, Juiz de Fora, Brazil)

Passive acoustic towed array during ship surveys has being applied to increase the knowledge about cetacean. From 2012 to 2015, towed arrays were used to investigate cetacean distribution along the western South Atlantic shelf break. Research cruises were performed between 26°S and 38°S over the continental shelf break and slope. Acoustic tracklines comprised an average of 780 nm of effort per survey. Hydrophone arrays (Auset®) were towed 150 and 300 m behind the vessel. The system was configured to give a variable frequency response from 1592 (High Pass Filter) to 100,000 Hz. Acoustic data were recorded as .wav files. Concurrent environmental and GPS data were logged automatically using WinCruz software. Visual positive identifications were associated to the acoustic recordings. The .wav files were analyzed using partially automated detections tools complemented with visual and acoustical searched for species confirmation whenever possible. A total of nine cetacean species were acoustically detected. The most frequently species were *Physeter macrocephalus*, *Delphinus delphis*, *Stenella longirostris*, *Orcinus orca*, and *Globicephala sp.*. Additional studies are needed to describe acoustic parameters of the various species present in this region in order to improve automated detection systems. [This study was supported by Instituto Aqualie and was funded by BG Group and Chevron Brasil Upstream Frade LTDA.]

Contributed Paper

2:45

4pAB6. Enhanced feature extraction using the Morlet transform on 1 MHz recordings reveals the complex nature of Amazon River dolphin (*Inia geoffrensis*) clicks. Marie Trone (Math and Sci., Valencia College, 1800 Denn John Ln., Kissimee, FL 34744, mtronedolphin@yahoo.com), Hervé Glotin (Comput. Sci., LSIS UMR 7296, DYNI, Aix Marseille Université, CNRS, ENSAM, Université de Toulon, Institut Universitaire de France (IUF), La Garde, France), Randall Balestriero (Comput. Sci., LSIS UMR 7296, DYNI, Université de Toulon, La Garde, France), and David E. Bonnett (LCDR USN (Ret), Silverdale, WA)

The Amazon River dolphin lives exclusively in freshwater throughout the Amazon River watershed, a dynamic and acoustically complex habitat. Although generally considered a relatively non-vocal species, recent evidence suggests that these animals are acoustically active, producing tremendous quantities of high-frequency, pulsed signals. Moreover, these pulsed signals appear to be considerably more complex than previously believed. This study explored the high-frequency pulsed emanations produced by Amazon River dolphins in Peru. Audio recordings were made using a two hydrophone array, one of which was sampled at 1 MHz, in August of 2015. Digitized recordings were analyzed using FFT and Morlet wavelets. Subsequently, unsupervised machine learning attempted to delineate various click categories based upon inter-click intervals, the frequency bandwidth of each click, and the formants contained within each click. Although the Morlet transform is much more robust and accurate for higher frequencies than the FFT, its performance was not constant for all frequencies. Thus, the Morlet transform and the FFT produced different click categories. Thus, formant results above 230 kHz most likely were skewed. These results are the first to clearly demonstrate the heterogeneity of the high-frequency pulsed emanations of the Amazon River dolphin.

Invited Paper

3:00

4pAB7. Steps towards promoting bioacoustics research in Mexico. Eduardo Vivas (CIBNOR, Av Instituto Politecnico Nacional 195, Playa Palo de Santa Rita Sur, La Paz, Baja California Sur 23096, Mexico, evivas@cibnor.mx)

Bioacoustics has played a major role in animal research in the past decade. Unfortunately, in Mexico, it is still seen as a novel approach, and few research groups incorporate it into their studies. To overcome this lag, work needs to be done in three major areas: Collaboration among experts, preparing young students, and providing specialized equipment. In Mexico, there are many institutions devoted to research in biology, but there are only a few researchers in acoustics because there are no graduate research programs in this area. Collaboration among acousticians interested in biology and biologists interested in acoustics becomes crucial, and future researchers, coming from both specialties, need to be prepared. Once this synergy is created, emerging research groups often see the cost and availability of specialized equipment as a limitation, so it is also necessary to focus on providing access to research-grade, low-cost hardware. In this talk, I will describe my efforts toward promoting bioacoustics research in Mexico through collaboration with other research centers, with examples of ongoing projects, graduate program support, and equipment presented in major peer-reviewed conferences that we have developed with low-cost, open source materials that are comparable with standard, high quality research equipment.

3:20–3:35 Break

Contributed Papers

3:35

4pAB8. Preliminary evidence for signature and copied whistles among spinner dolphins in the Southwest Atlantic Ocean: Beacon purpose? Juliana R. Moron (Instituto Aqualie, Universidade Federal de Juiz de Fora, Juiz de Fora 36010520, Brazil, julianamoron@hotmail.com) and Artur Andriolo (Laboratório de Ecologia Comportamental e Bioacústica, Instituto de Ciências Biológicas, Universidade Federal de Juiz de Fora, Juiz de Fora, Brazil)

In order to evaluate strategies of cohesion in a fission-fusion society, the occurrence of signature and copied whistles were investigated in free-ranging spinner dolphins, *Stenella longirostris*. Through an one-element hydrophone array towed over the Brazilian continental shelf break, a group of approximate 400 dolphins were recorded at 96 kHz/24 bits while navigating. The preliminary results demonstrated 218 similar signals that fit into previous definitions of signature or copied whistles. These whistles were produced in bouts with an inter-whistle interval of 0.066—11.56 s (mean $\pm$ SD: 2.66 $\pm$ 2.72) that varied from 2 to 32 repetitions comprising six different contour shapes. Thus, these data support previous hypothesis that these signals are important units in the dolphin's repertoire. It may also suggests a potential use on individual direction and localization, where repeated contours could be acting as a beacon to direct and locate the animals within the group. Additional research to ascertain the natural function of these vocalizations may clarify the basis for acoustic badges of membership and group organization of this cosmopolitan species.

3:50

4pAB9. Geographic variation assessment of Bryde's whale (*Balaenoptera edeni*) Be4 call in the Gulf of California. Eduardo Vivas (CIBNOR, La Paz, Baja California Sur, Mexico), Violeta C. Vera Cuevas (CIBNOR, Centro de Investigaciones Biológicas del Noroeste, S.C., Av. IPN 195. Col Playa Palo de Santa Rita Sur, La Paz 23096, Mexico, cassandra_vera@hotmail.com), Jorge Urbán Ramírez, Lorena Viloria Gómora (Programa de Investigación de Mamíferos Marinos. Departamento de Biología Marina, Universidad Autónoma de Baja California Sur, La Paz, Mexico), and Patricia Cortés Calva (CIBNOR, La Paz, Mexico)

Mysticeti whales modify their calls in response to an increase in noise level within their environment. This is particularly important since environmental noise has been increasing continuously in the last decades. In the case of Bryde's whale (*Balaenoptera edeni*), previous studies of its predominant call, Be4, have exposed differences in the duration of its main component (f0) for northern and southern areas of the Gulf of California (GC). The objective of this research is to determine if the differences found among Be4 calls (f0, duration, and energy distribution) are influenced by the noise levels in those areas. Results based on the characterization of Be4 Call and noise show that northern noise level is intense in the octave frequency band centered around 30 Hz, and could mask the 30 Hz component. This kind of noise is rare in the south (5%) and when it was present, the 30 Hz component was not registered, which might be an adaptation to noise. Predominant

moderate noise in the south within the band 20 to 70 Hz overlaps the signals and higher components are present, especially the 165 Hz component. Results suggest that the difference between call from north and south areas might also be related to the noise present.

4:05

4pAB10. An underwater acoustic camera for marine mammal vocalization interaction studies. Eduardo Vivas (CIBNOR, La Paz, Baja California Sur, Mexico), Omar Bustamante, and Sergio Beristain (Academia de Acustica, ESIME, ESIME Academia de Acustica, Instituto Politécnico Nacional, Instituto Politécnico Nacional s/n, Unidad Profesional Adolfo Lopez Mateos, Gustavo A. Madero, Mexico City, Distrito Federal 07738, Mexico, omarb.p@hotmail.com)

Acoustic Cams, promoted as "a way to listening with your eyes," are widely used in the industry to visually pinpoint particular sound sources in a clouded sound environment. A video camera and complex multiple microphone array configurations are used to find the direction of arrival of sound, and the results are color coded and superimposed in the video image. This technique has a great potential to be used in marine mammal studies where underwater vocal interactions among a close group needs to be determined. This is particularly important for *Otariids* and *Pinnipeds* for which the function of in-water calls involving no bubble emission needs to be studied. A prototype of an underwater acoustic camera built around a low cost open source linear array of hydrophones and a fishing underwater camera is presented. Results of the prototype test, first under controlled conditions, and then recording vocal interactions during normal swimming behavior of a colony sea lions (*Zalophus californianus*) in the Gulf of California, are presented.

4:20

4pAB11. Bioacoustic characterization of a coastal marine soundscape in Quintana Roo, Mexico. Heather R. Spence (City Univ. of New York Graduate Ctr., Psych. Dept., Hunter College, 695 Park Ave., New York, NY 10065, info@heatherspence.net) and Dorian S. Houser (National Marine Mammal Foundation, San Diego, CA)

Wild dolphins are exposed to a variety of biological and other natural and anthropogenic sounds. Several species of dolphins, including bottlenose dolphins (*Tursiops truncatus*), have been documented in the MesoAmerican Reef (MAR) region; however, there is little to no scientific monitoring of these populations. Characterizing the soundscape of wild dolphins on a section of the MAR would facilitate understanding the acoustic nature of this dolphin habitat. Passive Acoustic Monitoring using an Ecological Acoustic Recorder was conducted for one year off the coast of Quintana Roo, Mexico, just north of Isla Mujeres, a site where dolphins are known to be frequent. The soundscape was typified by natural sounds, however there was important periodic contribution by anthropogenic sources. Boat motor noise contributed to higher SPL_{rms} during the day than in the night and contributed to noise between 500 Hz and 25 kHz, which is consistent with small vessels. While boat motor noise was not as frequent as fish sounds nor as pervasive as snapping shrimp sounds, when present it potentially overwhelms the natural soundscape. Anthropogenic characteristics of this soundscape have implications for dolphin welfare and regulations in the nearby marine protected areas.

Invited Papers

4:35

4pAB12. Assessing the structure of a Neotropical bat community using acoustic monitoring techniques. Sergio Estrada Villegas (Smithsonian Tropical Res. Inst., Dept. of Biological Sci., Marquette Univ., P.O. Box 1881, Milwaukee, WI 53201-1881, estradavillegassergio@gmail.com), Christoph F. Meyer (Universidade de Lisboa, Lisboa, Portugal), Brian McGill (Univ. of Maine, Orono, ME), and Elisabeth K. Kalko (Univ. of Ulm, Ulm, Germany)

Determining the structure and composition of tropical communities is challenging because some species are rare or hard to detect. Within Neotropical bats, aerial insectivores have been systematically undersampled because they avoid mist nets, the traditional sampling tool. Advances in bioacoustic monitoring techniques have allowed the study aerial insectivorous bat (AIB) communities across various spatial scales and habitats. We present two studies that assessed the underlying mechanisms that structure an AIB community across the Isthmus of Panama. First, we evaluated how habitat fragmentation affected two guilds of AIBs and found higher species richness in islands than in continuous forests. Background clutter aerial insectivores showed compositional differences due to effects of isolation, area, and forest complexity, whereas open space bats were not affected by fragmentation. Second, we determined how climate and forest complexity affected AIB community structure at different spatial scales. We found that most of the variation in bat richness, abundance, and feeding activity occurred at the smallest spatial scale (10×10m) and was explained by habitat structure. In contrast, at large scales, climatic differences explained most of the variation in individual species' abundances. Interestingly, species richness peaked at intermediate levels of precipitation, while total abundance was very similar across sites.

4:55

4pAB13. Acoustic description and synchronization of the duetting species *Pezopetes capitalis*, Costa Rica. Carla V. Trejos (Biology, Universidad de Costa Rica, Santa Clara, San Carlos, Santa Clara, Alajuela 21001, Costa Rica, ktrejos07@gmail.com) and Gilbert Barrantes (Biology, Universidad de Costa Rica, Tres Rios, Cartago, Costa Rica)

Duetting has been widely studied, and its function vary across species, though in all cases temporal synchrony of the elements sang by each sex in the duet seems to play an important role. My goals in this study were to describe the structure of the duet of *Pezopetes capitalis*, and the degree of temporal synchronization of each individual of the pair during duetting. Duets of *P. capitalis* consist of overlapping elements between both members of a mated pair and can be initiated by either sex. Synchronization was similar between both individuals. For 11 pairs, both, females and males, reduce their silence intervals before singing a new element as a response to an increase in the duration of its partner's elements. Furthermore, both mates lower the high frequency of their elements of the second section, as the duet continues. Birds showed this pattern even in those cases where the female joins the male after he had began to sing its duet's part. This suggests a temporal coordination in frequencies (high frequency decreases with time). Indirect evidence suggest that duets in *P. capitalis* serve as a joint defense of territory and as a way to recognized mates when a pair reunited.

5:15

4pAB14. Scaled mining of environmental acoustic data from temperate to tropical forests, from ocean to tropical rivers: A convolutional feature learning approach. Herve Glotin (LSIS UMR 7296, DYNI, Comput. Sci., Aix Marseille Université, CNRS, ENSAM, Université de Toulon, Institut universitaire de France (IUF), USTV, Ave. Université, BP20132, La Garde 83957, France, glotin@univ-tln.fr)

Scaled passive acoustic monitoring has recently been developed to assess changes in biodiversity. It is applied to monitor changes in fauna composition associated with anthropogenic impacts and to improve the management of species and habitat conservation. Under the auspices of the "Scaled Acoustic BIODiversity" SABIOD project (CNRS MI/GDR MADICS/JASON UTLN), interdisciplinary teams collaborate to develop new joint machine learning and signal processing bioacoustic analyses. Currently, terabytes of sounds are recorded monthly using the innovative open SABIOD autonomous sensor arrays, positioned within forests or deep in the ocean. The challenges associated with tropical biodiversity as the variety of sound sources and the complexity of acoustic paths are being addressed by optimizing strategies that couple features learning, convolutional neural net, and advanced passive acoustic localization/filtering. Applications are many, including the classification of one thousand Amazon bird species, whale song classification, automatic indexing forest soundscapes, passive acoustic 3D tracking of bats and cetaceans with our patented system, and offshore cetacean monitoring. We discuss the main difficulties encountered and summarize the promising steps and strategies that we intend to pursue in the future coupling scaled CNN with Deep Scattering Spectrum. Demo/details: http://sabiod.univ-tln.fr.

THURSDAY AFTERNOON, 5 NOVEMBER 2015 — CLEARWATER, 1:30 P.M. TO 4:55 P.M.

Session 4pBA

Biomedical Acoustics and Physical Acoustics: Numerical and Analytical Modeling of Medical Ultrasound II

Martin D. Verweij, Cochair
Laboratory of Acoustical Wavefield Imaging, Faculty of Applied Sciences, Delft University of Technology, Lorentzweg 1, Delft 2628CJ, Netherlands

Robert McGough, Cochair
Department of Electrical and Computer Engineering, Michigan State University, 2120 Engineering Building, East Lansing, MI 48824

Chair's Introduction—1:30

Invited Papers

1:35

4pBA1. Simulation of wave propagation in anisotropic and viscoelastic tissues. Matthew W. Urban, Bo Qiang, Sara Aristizabal, Ivan Nenadic, Pengfei Song, Shigao Chen (Dept. of Physiol. and Biomedical Eng., Mayo Clinic College of Medicine, 200 First St. SW, Rochester, MN 55905, urban.matthew@mayo.edu), Robert J. McGough (Dept. of Elec. and Comput. Eng., Michigan State Univ., East Lansing, MI), and James F. Greenleaf (Dept. of Physiol. and Biomedical Eng., Mayo Clinic College of Medicine, Rochester, MN)

Many methods for characterization of the mechanical properties of soft tissues using propagating shear waves have been developed over the past two decades. Most of these methods assume that the shear wave is traveling in an elastic, isotropic tissue. However, many soft tissues are viscoelastic and have material properties that are directionally dependent or anisotropic. We have been developing methods to measure waves propagating in soft tissues to estimate the anisotropic viscoelastic material properties. To refine our measurement methods, we have also developed techniques and models for simulating the wave propagation in these types of materials. We have developed specialized finite element and pseudo-spectral models that can simulate viscoelastic transversely isotropic materials and compared our results with measurements in *ex vivo* porcine muscle. Additionally, we have developed a finite element model that simulates wave propagation in the myocardium by creating a model of layered transverse isotropic media oriented at different angles. We examined the effect of frequency of the waves for estimating of the angle of propagation. We compared these simulation results with experimental results from an *ex vivo* porcine left ventricular wall. These simulation methods provide insight for optimizing our measurement methods for *in vivo* characterization of material properties.

1:55

4pBA2. Numerical simulation of transcranial focused ultrasound therapy. Aki T. Pulkkinen (Dept. of Appl. Phys., Univ. of Eastern Finland, PO Box 1627, Kuopio 70211, Finland, Aki.Pulkkinen@uef.fi), Beat Werner, Ernst Martin (Ctr. for MR-Res., Univ. Children's Hospital, Zürich, Switzerland), and Kullervo Hynynen (Physical Sci. Platform, Sunnybrook Res. Inst., Toronto, ON, Canada)

Transcranial focused ultrasound therapy is becoming a viable tool for treatment of brain related disorders. The technique has been used for treatment of essential tremors and chronic neuropathic pain with good results. In addition, the technique has a broad scope of potential applications including: tumor ablation, localized drug delivery, thrombolysis, and neurostimulation. Performing an efficient focused ultrasound treatment delivery requires planning. Accurate computational models could, in principle, be utilized to provide auxiliary guidelines for such planning. In this work, a computational model is presented for simulating the propagation of ultrasound in transcranial setting. The effects of ultrasound propagation through soft tissue and bone are modeled by coupling wave equations of fluid and solid. The coupled model is numerically evaluated by using a hybrid simulation technique that couples finite difference method with a grid method. The resulting computational model is utilized to simulate focused ultrasound field in clinical patient treatment setting. Effect of absorption of heat into the brain tissue is further simulated by using the bioheat equation. Simulation results are compared with magnetic resonance thermometry performed during the clinical treatments for evaluation of validity of the computational model.

2:15

4pBA3. Viscosity-compensated shear speed imaging with acoustic radiation force. Yiqun Yang (Dept. of Elec. and Comput. Eng., Michigan State Univ., 2120 Eng. Bldg., East Lansing, MI 48824), Matthew W. Urban (Mayo Clinic College of Medicine, Rochester, MN), and Robert J. McGough (Dept. of Elec. and Comput. Eng., Michigan State Univ., East Lansing, MI, mcgough@egr.msu.edu)

Shear wave elastography imaging (SWEI) with ultrasound uses an applied acoustic radiation force to obtain quantitative images of liver, breast, prostate, and other soft tissues. In SWEI, shear velocity waveforms are collected along a line, and then, the shear speed is estimated either with a k-space or a correlation-based method. Although these yield effective estimates in elastic media, both methods consistently overestimate the shear speed in viscoelastic soft tissue because neither considers the shear viscosity. To address this problem, we have created a new shear speed estimation approach that also accounts for the effect of shear viscosity. The new approach is evaluated in a computational model that calculates the three-dimensional (3D) intensity field generated by a linear array transducer in FOCUS (http://www.egr.msu.edu/~fultras-web) and then simulates the shear wave velocities in a viscoelastic medium with a 3D finite difference model. In the viscoelastic shear wave simulation model, the shear speed constant is 1.5 m/s, and the shear viscosity is 1 Pa·s. Correlation-based shear speed estimates are compared to the viscosity-compensated estimates, and the results show that the viscosity-compensated approach consistently achieves improved shear speed estimates relative to the correlation-based method. [Work supported in part by NIH Grant Nos. R01EB012079 and R01DK092255.]

2:35

4pBA4. Simulation of shear wave elasticity imaging including speckle and refraction effects. Stephen A. McAleavey and Jonathan H. Langdon (Biomedical Eng., Univ. of Rochester, 309 Goergen BME/Optics Bldg., Rochester, NY 14627, stephen.mcaleavey@rochester.edu)

We present an overview of, and results from, a GPU-accelerated, 3D finite-difference time-domain acoustic radiation force impulse (ARFI) shear wave elasticity imaging (SWEI) simulator recently developed in our laboratory. The simulator allows modeling of the ultrasound "push" beam used to generate the shear wave, propagation of the shear wave in a viscoelastic, inhomogeneous medium, and simulation of ultrasound tracking of the shear wave. Spatial variations in both the ultrasound and shear wave speeds can be included to model ultrasound beam refraction errors and shear wave reflection and refraction by inclusions. Speckle has been shown experimentally to induce distortions in the apparent shape of a tracked shear wave, as well as biases in shear wave arrival time estimates used to generate shear wave speed images. A complete simulation of fully developed speckle captures this speckle bias effect but is time consuming. We present a dominant-speckle simulation approach that allows realistic modeling of the speckle-noise induced noise observed in videos of shear wave propagation in distributed speckle targets, and of artifacts in shear wave speed images. Matched simulation and phantom images of uniform and spherical inclusion targets are presented to demonstrate the ability of the simulation to capture key effects.

Contributed Paper

2:55

4pBA5. Numerical simulations of ultrasound-induced deformation of the lung surface. Brandon Patterson and Eric Johnsen (Univ. of Michigan, 1231 Beal Ave., Rm. 2016, Ann Arbor, MI 48109, awesome@umich.edu)

Ultrasound-induced lung hemorrhage remains the only known bioeffect of non-contrast, diagnostic ultrasound (DUS) found to occur in mammals. However, a fundamental understanding of the ultrasound-lung interaction is still lacking. In this study, we numerically simulate the deformation of the lung surface when subjected to clinically relevant pressure waves. We model the lung as air and the surrounding tissue as water. The two-dimensional, compressible Navier-Stokes equations are solved using a high-order accurate finite volume scheme. We first consider an air-water interface with a small sinusoidal perturbation and study the growth of the perturbations when subjected to a planar waveforms consisting of a step change in pressure. Scaling laws relating the waveform properties to the interface deformation rate are proposed for this simple case. Simulations with DUS waveforms indicate that the interface deformation depends on the unsteady nature of the wave. Preliminary analysis suggests that the initial interface deformation is driven by baroclinic vorticity deposited along the perturbed interface, and that the resulting stresses and strains are a possible candidate causing bioeffects.

3:10–3:30 Break

Invited Papers

3:30

4pBA6. Acoustics of finite Bessel tractor beams. Farid G. Mitri (Area 52 Technol., Chevron, 5 Bisbee Court, Santa Fe, NM 87508, f.g.mitri@ieee.org)

Surprises and counterintuitive effects occurring in physical phenomena rank high on the list of things that make science discoveries a source of perpetual excitement. Some examples illustrated here include, **(i)** the effects of attracting (i.e., pulling) a spherical particle placed in the field of a Bessel acoustical beam of progressive waves back towards the radiator's surface [Mitri, "*Single Bessel tractor-beam tweezers*," Wave Motion **51**, no. 6, pp. 986–993, 2014], **(ii)** suppressing some of the sphere's resonances [Mitri, "*Near-field acoustic resonance scattering of a finite Bessel beam by an elastic sphere*," IEEE Trans. Ultrason. Ferroelectr. Freq. Control, **61**, no. 4, pp. 696–704, 2014], **(iii)** and the emergence of a "negative radiation torque," meaning that an incoming high-order Bessel vortex beam with a positive topological charge (known also as the order of the beam) would rotate a viscoelastic sphere in the opposite sense of the beam handedness [Silva *et al*., "*Radiation torque produced by an arbitrary acoustic wave*," Europhys. Lett., **97**, art. no. 54003, 2012]. Results and numerical predictions illustrate the analysis and extensions to other geometries, such as rigid oblate and prolate spheroids [Mitri, "*Axisymmetric scattering of an acoustical Bessel beam by a rigid fixed spheroid*," http://arxiv.org/abs/1505.06754v3, *ibid*. "*Acoustic radiation force on oblate and prolate spheroids in Bessel beams*," Wave Motion, **57**, pp. 231–238, 2015] are also mentioned.

3:50

4pBA7. Acoustic fields and ultrasound contrast agents. John S. Allen (Mech. Eng., Univ. of Hawaii, 2540 Dole St., Holmes Hall 302, Honolulu, HI 96822, alleniii@hawaii.edu)

Ultrasound contrast agents are encapsulated gas bubbles which oscillate nonlinearly upon acoustic excitation. Recent developments include targeted agents for molecular imaging applications and ultrasound contrast agents such those with polymer shells specifically designed for high frequency ultrasound applications. Though a variety of models have been proposed since the early development of the agents, many outstanding questions exist on the specifications of the shell and most appropriate model for the different materials. Moreover, only recently have independent mechanical measurements been attempted to advance and facilitate modeling efforts. The various shell models are reviewed and discussed. Validation of models of polymer shell agents are given with respect high frequency acoustic microscopy quantification of the shell elastic properties. Recent work on the modeling and design of agents for high frequency subharmonic excitation based on adaptive signal processing and nonlinear time series methodologies is overviewed. Novel analytical method for treating hydrodynamic interactions with respect to unsteady drag and secondary acoustic radiation forces are outlined with respect to drug delivery and sonoporation applications. The direction and magnitude of coupled oscillations are investigated with respect evolution of the amplitude and phase and the transfer entropy.

Contributed Papers

4:10

4pBA8. A two-criterion model for microvascular bioeffects induced *in vivo* by contrast microbubbles exposed to medical ultrasound. Charles C. Church (NCPA, Univ. of MS, 1 Coliseum Dr., University, MS 38677, cchurch@olemiss.edu) and Douglas L. Miller (Radiology, Univ. of Michigan Health System, Ann Arbor, MI)

The mechanical index (MI) assumes that bubbles of all relevant sizes exist in tissue, yet that assumption is approximated only in studies that include use of a microbubble contrast agent. If the MI is taken to be the key dosimetric parameter, then it should allow science-based safety guidance for contrast-enhanced diagnostic ultrasound. However, theoretical predictions based on the MI typically do not concur with the frequency dependence of experimentally measured thresholds for bioeffects. For example, experimental measurements of thresholds for glomerular capillary hemorrhage in rats infused with contrast microbubbles (Miller *et al*. UMB 2008;34:1678) increase approximately linearly with frequency while the MI assumes a square-root dependence. Here, thresholds for inertial cavitation were computed for linear versions of the acoustic pulses used in that study assuming bubbles containing either air, C_3F_8, or a 1:1 mixture of the two and surrounded by either blood or kidney tissue. While no single threshold criterion was successful, combining results for one criterion that maximized circumferential stress in the capillary wall with another that ensured an inertial collapse, produced thresholds that were consistent with experimental data. This suggests that development of a contrast-specific safety metric may be achieved by further testing and confirmation in different tissues.

4:25

4pBA9. Myocardial cavitation-enabled therapy modeling. Yiying I. Zhu (Biomedical Eng., Univ. of Michigan, 1301 Catherine St., Med Sci I 3218A, Ann Arbor, MI 48109-5667, zhuyiy@umich.edu), Douglas L. Miller, and Oliver D. Kripfgans (Radiology, Univ. of Michigan, Ann Arbor, MI)

A therapeutic method named Myocardial Cavitation Enabled Therapy aiming at noninvasive cardiac tissue reduction is modeled here. Sparsely distributed microlesions induced by ultrasound cavitation of contrast agents are hypothesized to cause myocardial shrinkage. The objective is to model lesion formation based on the acoustic field and plan treatments accordingly. An ultrasound field simulation was established in Field II, an acoustics toolbox. It simulates the acoustic field of a 1.5 MHz ultrasound burst of five cycles at 4.0 MPa peak rarefactional pressure amplitude (PRPA) by use of a F# 2 single element therapeutic transducer, as used in concomitant animal studies. Medium parameters, including speed of sound, density, absorption, and B/A were set to water path and heart tissue along the acoustic path. Lesions were masked as region exceeding 2 MPa PRPA, the threshold of microlesions occurrence. The lesion volume is 143 μL compared to *in vivo* rodent study of approximately 100 μL. To reach a larger lesion in pigs, sweeping of the ultrasound beam is needed using a phased array. Nonlinear acoustic simulation from k-Wave software will assist transitioning from single element to imaging array apertures in terms of interpreting possibly different lesion formation resulting from nonlinearities.

4:40

4pBA10. Frequency dependence of thresholds for lethal cardiomyocyte injury in myocardial contrast echocardiography. Douglas L. Miller, Xiaofang Lu, and Chunyan Dou (Radiology, Univ Michigan, 3240A Medical Science I, 1301 Katherine St., Ann Arbor, MI 48109-5667, douglm@umich.edu)

Contrast enhanced diagnostic ultrasound employs microbubble activation for microvascular imaging; however, the on-screen Mechanical Index is a poor parameter for safety guidance. More research is needed on microvascular bioeffects, particularly their variation with frequency. A GE Vivid 7 with an S3 probe operated at 1.6 MHz, and an S5 probe operated at 2.5 and 3.5 MHz was used for myocardial contrast echocardiography of rats mounted in a water bath. Power settings were varied in 2 dB steps for determination of the thresholds for cardiomyocyte injury. The contrast agent was made to duplicate the properties of the clinical agent Definity. The scans were intermittently triggered each 4 heartbeats from the ECG signal. The cardiomyocyte death was assessed using Evans blue vital staining. Thresholds were defined as the mean of the lowest exposure with a statistically significant cardiomyocyte death and the next lower exposure level. Thresholds were 1.2 MPa, 1.7 MPa, and 2.7 MPa peak rarefactional pressure amplitude (derated for 1 dB/cm/MHz attenuation) for 1.6, 2.5, and 3.5 MHz, respectively. Linear regression showed that the thresholds were essentially proportional to frequency ($0.72\ \boldsymbol{f^{1}.02}$, $r^2 = 0.97$). Theoretical analysis is under development to explain this dependence and develop a contrast-specific safety parameter.

THURSDAY AFTERNOON, 5 NOVEMBER 2015 ORLANDO, 1:00 P.M. TO 2:15 P.M.

Session 4pEA

Engineering Acoustics and Structural Acoustics and Vibration: Layered Media

Elizabeth A. Magliula, Cochair
Division Newport, Naval Undersea Warfare Center, 1176 Howell Street, Bldg. 1302, Newport, RI 02841

Andrew J. Hull, Cochair
Naval Undersea Warfare Center, 1176 Howell St., Newport, RI 02841

Invited Papers

1:00

4pEA1. Acoustic scattering from rib-stiffened finite bilaminar composite cylindrical shells- 3-D solution. Sabih I. Hayek (Eng Sci., Penn State, 953 McCormick Ave., State College, PA 16801-6530, sihesm@engr.psu.edu) and Jeffrey E. Boisvert (NUWC, Newport, RI)

The acoustic scattering from an insonified finite bilaminar cylindrical shell stiffened by a thin circular rib-stiffener is analyzed. The two cylindrical shell laminates are perfectly bonded having the same lateral dimension but have different radii and material properties. The bilaminar shell is analyzed using the exact theory of three-dimensional elasticity. The thin rib-stiffener has rectangular cross-section and is perfectly bonded to the inside of the inner shell. It is analyzed as a thin elastic ring. The finite shell has shear-diaphragm supports at ends $z = 0$ and L and is terminated by two semi-infinite rigid cylindrical baffles. The shell is insonified by an incident plane wave at an oblique incidence angle. The scattered acoustic farfield is evaluated for various incident wave frequencies and stiffener location and dimensions. A uniform steel stiffened shell in water was initially analyzed to study the influence of stiffened-shell geometries on the scattered acoustic farfield. A second shell made up of an outer elastomer shell bonded to an inner stiffened steel shell was also analyzed to study the influence of elastomeric properties on acoustic scattering. [Work supported by NAVSEA Division Newport under ONR Summer Faculty Program.]

1:20

4pEA2. Imaging a barely visible impact damage in a laminated composite without contact using air-coupled nonlinear elastic wave spectroscopy. Marcel C. Remillieux (Geophys. Group (EES-17), Los Alamos National Lab., MS D446, Los Alamos, NM 87545, mcr1@lanl.gov), Łukasz Pieczonka (Dept. of Robotics and Mechatronics, AGH Univ. of Sci. and Technol., Krakow, Poland), Pierre-Yves Le Bas, Brian E. Anderson, and TJ Ulrich (Geophys. Group (EES-17), Los Alamos National Lab., Los Alamos, NM)

We present the first set of experiments in which air-coupled ultrasonic emission is used for exhaustive and rapid imaging of a Barely Visible Impact Damage (BVID) introduced by an impact test on a laminated composite. Such experiments have been limited thus far by the impedance mismatch of nearly five orders of magnitude between the air and an elastic solid. This limitation was overcome by designing an ultrasonic source capable of generating a focused elastic wave field in the sample with an amplitude sufficiently large to allow the use of nonlinear elastic wave spectroscopy. The images of the damaged area obtained with the proposed apparatus and signal processing technique reveal the same features, namely a delamination and a crack, as with vibro-thermography and more features than with a C-scan, but remove the need to be in direct contact with the composite, thus reducing the test time by orders of magnitude. [This work was supported by the U.S. Department of Energy through the LANL/LDRD Program and the Fuel Cycle R&D, Used Fuel Disposition (Storage) campaign. The visit of Dr. Pieczonka at LANL was supported by the Foundation for Polish Science (FNP) within the scope of the WELCOME Program—project no. 2010-3/2.]

1:40

4pEA3. A stochastic inverse solution for functionally graded acoustic layered metamaterial validation. Heather Reed (Weidlinger Assoc., Inc., 40 Wall St 18th Fl., New York, NY 10005, heather.reed@wai.com), Jeffrey Cipolla (Weidlinger Assoc., Inc., New York, New York), and Patrick Murray (Weidlinger Assoc., Inc., New York, NY)

Functionally graded acoustic metamaterials (FGAMs) can be designed to have specific waveguide properties. In sonar applications, FGAMs can be tailored to resist incident wave reflection. As these materials do not exist naturally, they must be fabricated by gradually layering manufactured, resulting in a (usually smooth) variation of properties. Validating material properties of FGAMs is difficult with conventional tests, as the distribution of material properties over the layered structure results in a non-unique solution if typical data (e.g., compressive strain) is measured. This talk will demonstrate an approach to characterize the functionally graded material properties by parameterizing how the functionally graded material changes throughout the specimen. Experiments designed to minimize the uncertainty surrounding the FGM model parameters are formulated and evaluated numerically. The FGM model parameters are estimated by Markov chain Monte Carlo so that a probability distribution surrounding each parameter is recovered. Probability distributions enable uncertainty quantification (UQ) surround the validated material parameters. UQ is important as uncertainty surrounding the parameter results directly translates into uncertainty surround the FGM performance. The approach is verified by bootstrap analyses of known FGAM distributions.

Contributed Paper

2:00

4pEA4. Seismic surface wave method for subsurface layered soil exploration. Zhiqu Lu (National Ctr. for Physical Acoust., Univ. of MS, 1 Chucky Mullins, University, MS 38677, zhiqulu@olemiss.edu)

Subsurface soils in the vadose zone are layered structures. Typically, there are three distinctive layers at the depths of a few meters, featuring as a top rigid layer (due to surface crusting and sealing), a middle soft zone (due to soil moisture), and a region with stiffness increasing with depth (due to increased overburden pressure). The properties of subsurface soils in the vadose zone are often altered by natural events (weather and chemical reactions) or cultural activities (compaction). The exploration of the subsurface soils is required for agricultural, environmental, civil engineering, and military applications. In this talk, a seismic surface wave technique is developed, a so-called high frequency multi-channel analysis of surface wave method (HF-MASW). In the method, an electromagnetic shaker is placed on the ground and serves as a seismic source, a vibration sensor (either a laser Doppler vibrometer or an accelerometer) is used to record surface vibrations at multiple locations alone a straight line. Rayleigh waves propagation theory, layered structural modeling, and spectral analysis are applied for processing received signals and conducting inversion. A soil profile in terms of shear wave velocity is determined from the method. In this talk, several applications using MASW method will be reported, including layering delineation with cross-section imaging, weather effects monitoring, fragipan detection, and surface crusting and sealing evaluation.

Session 4pMU

Musical Acoustics: Acoustical Evolution of Musical Instruments

Whitney L. Coyle, Chair
The Pennsylvania State University, 201 Applied Science Building, University Park, PA 16802

Chair's Introduction—1:00

Invited Papers

1:05

4pMU1. Acoustical innovations in nineteenth-century violinmaking. Sarah Gilbert (Musicology, Florida State Univ., 122 N. Copeland St., Tallahassee, FL 32304, smg11b@my.fsu.edu)

Tensions between innovation and tradition in violinmaking have impeded the acceptance of most attempts to improve or alter the structure of the instrument. The nineteenth century, however, saw a proliferation of innovative violins as luthiers responded to musical developments and changing social and economic environments during the Industrial Revolution. As nineteenth-century composers called for greater range and diversity in timbre, chromaticism, dynamics, range, and key, instruments were developed to accommodate these demands. But perhaps more important than the purely musical considerations was the interdisciplinary collaboration between musicians and scientists in the pursuit of acoustic perfection. Many luthiers, such as François Chanot and Jean-Baptiste Vuillaume, viewed themselves as scientists and engineers, experimenting with acoustic properties and new materials in order to improve upon the existing form of the violin. In a reciprocal relationship, acousticians recognized musical instruments as rich sources for the study of acoustic principles, and luthiers consulted with acousticians and engineers about the technical construction of experimental forms. In this paper, I examine the technical construction of several of these instruments for insight into their novel construction techniques and acoustical properties, relating this experimental trend to the alliance of the sciences and arts during the Industrial Revolution.

1:25

4pMU2. Air resonance power efficiency evolved in the violin and its ancestors. Hadi T. Nia, Ankita D. Jain, Yuming Liu, Mohammad-Reza Alam (Dept. of Mech. Eng., Massachusetts Inst. of Technol., 77 Massachusetts Ave., 5-212, Cambridge, MA 02139), Roman Barnas (Violin Making, North Bennet St. School, Boston, MA), and Nicholas C. Makris (Dept. of Mech. Eng., Massachusetts Inst. of Technol., Cambridge, MA, makris@mit.edu)

Acoustic radiation from a violin can be explained by clear physics at its lowest frequency resonance (air resonance), which also helps to explain key aspects of violin design evolution. It is found that inefficient, inactive void area was decimated and air-resonance power doubled as sound-hole geometry of the violin's ancestors slowly evolved over eight centuries from simple circles of Medieval 10th century fitheles to complex f-holes of the late Renaissance and Baroque period. F-hole length then increased across centuries in the renowned Amati, Stradivari, and Guarneri workshops, favoring correspondingly higher air-resonance power, by processes consistent with random craftsmanship mutations and subsequent selection.

1:45

4pMU3. Acoustical evolution of the viola da gamba. D. Murray Campbell (School of Phys. and Astronomy, Univ. of Edinburgh, James Clerk Maxwell Bldg., Mayfield Rd., Edinburgh EH9 3JZ, United Kingdom, d.m.campbell@ed.ac.uk), J. Patricia Campbell (Edinburgh College of Art, Univ. of Edinburgh, Edinburgh, United Kingdom), and Jim Woodhouse (Dept. of Eng., Univ. of Cambridge, Cambridge, United Kingdom)

The viola da gamba (or viol) emerged as a distinctive type of bowed string instrument during the fifteenth century. Its characteristics include a fingerboard with gut frets and between five and seven strings tuned in fourths with a central major third. By the end of the fifteenth century, the instrument was being constructed in a range of sizes. To investigate the acoustical significance of design changes during the evolution of the viol family in the sixteenth and seventeenth centuries admittance measurements at the bridge have been made on a wide variety of reproductions of viols and other early bowed-string instruments. A comparison of the results with existing data on modern violin-family instruments, and other stringed instruments such as guitars, reveals that the acoustic behavior of viols is more disparate than that of violins or guitars. Models that have been measured include examples with and without a soundpost, and a correspondingly wide range of frequency responses is seen. The influence of the viol bridge and the significance of the adoption of overwound bass strings are also discussed.

2:05

4pMU4. Early history of the European free reed instruments. James P. Cottingham (Phys. Dept., Coe College, Cedar Rapids, IA 52402, jcotting@coe.edu)

Free reed instruments have been known in Asia for thousands of years, but the Western free reed instruments such as the harmonica, accordion, and reed organ were only invented and developed during the last two centuries. In 1780, Kratzenstein published a paper in St. Petersburg describing a speaking machine, which produced vowel sounds using free reeds with resonators of various shapes. This event marks a convenient, if arbitrary, starting point for the history of the free reed musical instruments of European origin. These instruments developed rapidly, and by 1850, the accordion, concertina, harmonica, reed organ, and harmonium all had been invented and developed into more or less final form. This paper presents some episodes in the development of these instruments. Also addressed is the question of the influence of the Asian free reed mouth organs on the origin and development of the European free reeds.

2:25

4pMU5. Evolution of the piano. Nicholas Giordano (Phys., Auburn Univ., 246 Sci. Ctr. ClassRm. Bldg., Auburn, AL 36849-5319, njg0003@auburn.edu)

The first pianos were built around 1700 and unlike the case with many other instruments, we know precisely what the first pianos were like since several of them still exist. These early pianos were patterned after Italian harpsichords of the time, with the plucking mechanism of the harpsichord replaced by an “action” mechanism that enabled hammers to strike the strings. The first pianos had a compass of four octaves with brass and iron strings, and a frame composed of wood. The design of those instruments along with the various stages of their evolution into the modern piano with its 88 notes, steel strings and a cast iron plate will be described. The manner in which this evolution was driven by the needs of composers and musicians along with advances in technology will also be discussed.

2:45–3:00 Break

3:00

4pMU6. The acoustic complexity of keyboard percussion instruments. Part I: (What we know and what we wish we knew) Leigh Howard Stevens. Leigh H. Stevens (21 South Riverside Dr., Neptune, NJ 07753, Drsplatbar@Gmail.com)

Part I Introduction to tuned bars and resonators Keyboard Percussion Instruments include innumerable variations of marimbas, xylophones, vibraphones, and glockenspiels. The tone bars, made of wood, metal, or other materials, are often incapable of producing sufficient volume without the amplifying assistance of a resonating tube or chamber. Unfortunately, these two vibrating systems (bar and resonator) react in opposite directions to changes in temperature, decoupling from their ideal relationship with any rise or fall in temperature. Also, unlike string, brass, or woodwind instruments, where the overtones are produced “naturally” and “are what they are,” the overtones of the bars of keyboard percussion instruments can be manipulated and controlled in the design and manufacturing process. For example, while both instruments have tone bars made of rosewood, the xylophone has the second partial tuned to an octave and a perfect fifth above the fundamental, while a marimba has the second partial tuned to a frequency two octaves above the fundamental. While we can identify and lower the frequency of many of the prominent partials contained in tone bars, their interdependence within the bar, and their coupling with a resonator provides a level of complexity for the designer and manufacturer, perhaps unrivaled in musical instrument manufacturing.

3:20

4pMU7. The acoustic complexity of keyboard percussion instruments. Part II: What we know and what we wish we knew Leigh Howard Stevens. Leigh H. Stevens (21 South Riverside Dr., Neptune, NJ 07753, Drsplatbar@Gmail.com)

Part II Resonating chamber shapes, their Q factor, their resultant musical character Resonating Chambers come in many shapes and sizes and largely determine the power, focus, and sustain of whatever the associated tone bar is coupled with it. The most basic box resonators are found on elementary education musical instruments such as Carl Orff tuned percussion instruments: demonstration of various tone bars resonating over box, rectangular, square and round tube resonators; demonstration of high and low Q systems responding to the same tone bar; demonstration of well-tuned, miss-tuned, and deliberately de-tuned partials on various tone bars; and questions from (and challenges to!) the audience.

3:40

4pMU8. A short acoustical history of flutes from the Paleolithic period to nowadays. Benoit FABRE and Camille Vauthrin (Sorbonne Universités, UPMC Univ Paris 06, UMR 7190, Institut Jean le Rond d’Alembert, 75015, Paris, France, CNRS, UMR 7190, Institut Jean le Rond d’Alembert, 11, rue de Lourmel, 75015 Paris, France, UPMC-LAM d’Alembert, LAM d’Alembert, Sorbonne Universités, UPMC Univ Paris 06, UMR 7190, Institut Jean le Rond d’Alembert, 75015, Paris, France, vauthrin@lam.jussieu.fr)

Flutes appear at different time periods and places over the world. The instrument making adapts to the cultural and musical context. Studying the instruments from the acoustical point of view allows to understand some evolutions in instrument making, focusing on different aspects such as the geometry of the bore of the instrument and its consequences on the passive resonances, the playing conditions and aeroacoustical consequences, the sound characteristics such as intonation, spectral content, and sound intensity. The talk will focus on a few instruments and periods in order to illustrate some evolutions and relations between these different aspects.

4:00

4pMU9. The clarinet: Past, present, and future. Whitney L. Coyle (The Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, wlc5061@psu.edu)

The modern clarinet is the result of several hundreds of years of research and craftsmanship. This paper will discuss where the clarinet has been acoustically by studying some input impedance spectrum characteristics of historical instruments (the Baroque Chalumeau, 13-key, etc.). The paper will then present where we are at present—focusing on the same measurement information for a full Boehm system (multiple models of Bb clarinets) and a comparison of the modern French vs. German system clarinet. Finally, a few words will be given on the possible future of the instrument—where we are going, what are we doing to continue improving the instrument's quality and playability.

4:20

4pMU10. Acoustical and metalworking techniques study of cornua, roman brass instruments, and their sound reproduction by physical modeling sound synthesis. René E. Caussé (Ircam - UMR STMS CNRS - UPMC, 1 Pl. Igor Stravinsky, Paris 75004, France, Rene.Causse@ircam.fr), Benoît Mille (Ctr. de Recherche et de Restauration des Musées de France (C2RMF), Paris, France), and Margaux Tansu (Ecole nationale supérieure de chimie de Rennes (ENSCR), Rennes, France)

Under the musical instrument analysis project of ancient societies, combining musical archaeologists and research laboratories on materials and acoustic, a first technological study was conducted in the storage rooms of the Naples Archaeological museum. The five instruments were incomplete and consist of several pieces assembled on a plexiglass tube that does not allow the direct measurement of the bore. Two mouthpieces were associated with these instruments. Morphometric study was conducted and detached fragments of the Cornua were collected. The corresponding samples were then analyzed under a Bright Field Optical Microscope (BFOM) and a Scanning Electron Microscope (SEM) in order to determine the metalworking techniques. To determine the elemental composition of the Cornua (alloying and trace elements), analyses by means of Particle Induced X-ray Emission (PIXE) using the Ion beam facility AGLAE at C2RMF. From the morphometric data, the bore of the different cornua was reconstruct and their resonant frequencies calculated. Several possibilities of reconstituting the bore were tested and compared. Finally, the resonances obtained from the reconstructions of these instruments of the past will be compared with those of present instruments like French horn and trombone. The various reconstructions will also be compared from sound examples obtained through physical modeling synthesis.

Contributed Paper

4:40

4pMU11. The Tintignac carnyx: An acoustical study of an early brass-wind instrument. Michael Newton (Acoust. and Audio Group, Univ. of Edinburgh, James Clerk Maxwell Bldg., Edinburgh EH9 3FD, United Kingdom, michael.newton@ed.ac.uk), John Kenny (Royal Conservatoire of Scotland, Glasgow, United Kingdom), John Chick, Amaya Lopez-Carromero, D. Murray Campbell (Acoust. and Audio Group, Univ. of Edinburgh, Edinburgh, United Kingdom), and Joel Gilbert (Laboratoire d'Acoustique de l'Université du Maine, Le Mans, France)

The carnyx was a metal wind instrument used by Celtic peoples around two thousand years ago. It was approximately two meters long with a bell in the shape of an animal head. In 2004, an excavation at Tintignac in the Corrèze district of France uncovered a horde of bronze instruments, including parts of several carnyxes. It proved possible to assemble an almost complete carnyx from these parts, and in 2011, a copy of this carnyx was made in brass by Jean Boisserie. The acoustical behavior of the brass copy was studied by Joel Gilbert and colleagues at the Université du Maine in Le Mans; this work led to the proposal that the musical performance of the instrument would be improved if an additional section of tubing was included in the reconstruction. Later, Boisserie made a second copy in bronze, together with several optional extension pieces. The musical performing possibilities of the bronze copy have been studied by the musician John Kenny, and its acoustical behavior has been studied at the University of Edinburgh. The results of these studies, including measurements of sound radiation by the large bronze ears attached to the head, are presented and discussed.

4p THU. PM

Session 4pNS

Noise, Physical Acoustics, and Animal Bioacoustics: Wind Turbine Noise

Nancy S. Timmerman, Cochair
Nancy S. Timmerman, P.E., 25 Upton Street, Boston, MA 02118

Paul D. Schomer, Cochair
Schomer and Associates Inc., 2117 Robert Drive, Champaign, IL 61821

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

Robert D. Hellweg, Cochair
Hellweg Acoustics, Wellesley, MA 02482

Chair's Introduction—1:15

Invited Papers

1:20

4pNS1. A proposed test of some people's ability to sense wind turbines without hearing or seeing them. Paul D. Schomer (Schomer and Assoc. Inc., 2117 Robert Dr., Champaign, IL 61821, schomer@SchomerAndAssociates.com)

Around the world, there are people that allege they sense times when significant power is being generated by a turbine from when little to no power is being generated. In contrast, the American Wind Energy Association (AWEA) and its Canadian counterpart (CanWEA) maintain that this is not possible, without these people hearing the turbines or seeing them move. A simple test can quickly clarify these opposing views. Notionally, people who live in residences situated beyond the threshold of hearing (indoors) would be tested in the area that they can sense the wind turbines. They would be told to tell the proctor when they sensed the turbine at power. They would be told the turbine might or might not be turned on in the next X hours. The area of the house they were in would have to be free of any visible or audible clues. No communication would be permitted between the subject and the outside world. The proctor also would not know when or if the turbine would come on. These and other controls like these would make for a quick, simple, unbiased test of some people's ability to sense wind turbines without hearing or seeing them.

1:40

4pNS2. Advanced signal processing techniques and optimized measurements to extract wind turbine signatures in nearby homes. Andy Metelka (Sound and Vib. Solutions Canada, Inc., 13652 4th Line, Acton, ON L7J 2L8, Canada, ametelka@cogeco.ca)

Previous measurements in homes near wind turbines indicate higher pressure levels below 10 Hz than audible pressure levels measured at the same time and location (ASA, vol. 20, 2013 Dooley & Metelka). Blade pass harmonic pressures were proven to be higher relative to basement ground-borne vibration and seismic vibration in floors throughout the home are reexamined simultaneous to broadband pressure. Although the pressure vs. frequency distribution appear to be the same at different locations in one room, they differ when compared to other rooms due to room dimensions. Further multichannel signal processing using a variety of different sensors at various locations inside a home identify areas in a home that are least affected by wind turbine low frequency discrete harmonics as well as acoustic metrics. Wind direction, wind speed, and other factors are precisely measured simultaneously. Sound level metrics inside homes are also compared to narrowband FFT data in an attempt to quantify annoyance and derive alternate metrics.

Contributed Papers

2:00

4pNS3. Radial mode analysis of broadband noise in flow ducts using azimuthal sensor array. Kunbo Xu (School of Power and Energy, Northwestern PolyTech. Univ., No. 127 Youyi Rd., Beilin District, Xi'an, Shaanxi 710072, China, 364398100@qq.com)

For the evaluation and improvement of noise reduction notions and the verification of broadband sound power measurement in flow ducts, it is interesting to obtain the detailed information of the in-duct acoustic mode spectrum and subsequent broadband noise sources separation. A new broadband noise mode analysis method, which uses full wall-mounted sensor array, was experimentally applied on broadband sound fields at three operation conditions, which were generated by means of high loaded single-stage axial flow fan test rig. Two axial sensor arrays were mounted wall-flush upstream of the fan. Measurements were made at operating conditions from 60% to 100% rotor design speed. On the whole, the new method behaves robustly. It delivers physically meaningful broadband mode amplitudes. Mode coherence functions are calculated between all pairs of propagating modes. This feature enables the detailed comparison of different sound fields with characteristically coupled mode pairs. For tonal noise, mode coherence results show that modes are correlated with the source and with each other, especially at blade passing frequencies. The experimental outcome proves the usefulness of the analysis technique for interpreting and understanding broadband sound propagation in turbomachinery flow ducts.

2:15–2:30 Break

2:30–5:00 Panel Discussion

THURSDAY AFTERNOON, 5 NOVEMBER 2015 — ST. JOHNS, 1:35 P.M. TO 4:25 P.M.

Session 4pPA

Physical Acoustics and Noise: Launch Vehicle Acoustics II: Analysis and Modeling of Noise from Supersonic Jets

Alan T. Wall, Cochair
Battlespace Acoustics Branch, Air Force Research Laboratory, Bldg. 441, Wright-Patterson AFB, OH 45433

Richard L. McKinley, Cochair
Air Force Research Lab., Wright-Patterson AFB, OH 45433-7901

Chair's Introduction—1:35

Invited Papers

1:40

4pPA1. Acoustic field of a shielded rectangular supersonic jet. Ephraim Gutmark, Pablo Mora, and Florian Baier (Aerosp. Eng., Univ. of Cincinnati, Rhodes Hall 799, Cincinnati, OH 45221-0070, ephraim.gutmark@uc.edu)

The impact of a flat surface installed parallel to a supersonic rectangular jet of AR = 2 on the far field noise is studied. Far-field cold jet results at design, over-expanded and under-expanded conditions are compared between the free-field jet, the nozzle with the surface matching with the exit lip ($h = 0D_e$), and the surface at different stand-off positions away from the jet axis, h = 1, 2, 3 D_e. Results are shown for all jet azimuthal angles. When the surface is installed at $h = 0D_e$, broadband shock-associated noise intensity was decreased and its peak frequency shifted. At NPR = 2.5, shock noise appears to be entirely mitigated. Also, a low frequency noise component is observe below St = 0.15, and assumed to be related to the trailing edge/jet plume interaction. At NPR = 3.0, strong screech tones are mitigated with the surface installed at $h = 0D_e$. In the shielded region, noise levels are significantly lower for all plate positions. Mixing noise and Mach wave radiation are affected by the plat stand-off location at the $\phi = 90^o$. Heated jet measurements up TR = 3.0, near-field and high-speed shadowgraph visualization are performed and reported.

2:00

4pPA2. Modeling of fighter jet cockpit noise. Alan T. Wall (Battlespace Acoust. Branch, Air Force Res. Lab., Bldg. 441, Wright-Patterson AFB, OH 45433, alantwall@gmail.com), Hilary L. Gallagher, and Richard L. McKinley (Battlespace Acoust. Branch, Air Force Res. Lab., Wright-Patterson Air Force Base, OH)

Military fighter jet cockpit noise may pose a hearing loss risk to pilots and can disrupt communications. The current study is an attempt to model the noise levels in the cockpit of F-35 aircraft as a function of aircraft operation parameters. Levels are based on a complex interplay of parameters such as airspeed, altitude, dynamic pressure, and the flow rate of the in-cockpit environmental control system. Full population of an operational envelope from point-by-point measurements would require an excessive number of flights. In the current work, in-cockpit noise levels were measured during sampled flight conditions. Correlations between levels and flight parameters were investigated. A preliminary parameter-based model is presented, and the ability to predict levels within the operational flight envelope is addressed. [Work supported by USAFRL through ORISE.]

2:20

4pPA3. Development of hearing protection device calculators for the F-35A and F-35B tactical aircraft. Anthony R. Pilon (Adv. Dev. Programs, Lockheed Martin Aeronautics Co., 1011 Lockheed Way, Palmdale, CA 93599-1100, tony.pilon@lmco.com)

Simple spreadsheet-based near field noise prediction tools for the F-35A (conventional takeoff and landing) and F-35B (short takeoff and vertical landing) aircraft have been developed recently. Near field noise predictions are based on empirical data collected during static engine runs conducted at Edwards AFB, and during hover/vertical landing operations at MCAS Yuma in September, 2013. The tools are programmed to calculate near field one-third octave band sound pressure levels as a function of engine power setting and listener location. The calculated near field levels can be used to determine the *Total Daily Exposure (TDE)* of personnel required to work in the vicinity of powered aircraft. Alternatively, the calculators can be used to determine the amount of hearing protection required so that the personnel TDEs remain below 1.0. The calculators are adjustable so that they can be employed for differing domestic and international noise exposure regulations. [*Work supported by the Air Force Research Laboratory and Ball Aerospace*.]

2:40

4pPA4. Jet noise of high-performance aircraft at afterburner. Christopher Tam (Mathematics, Florida State Univ., 1017 Academic Way, Tallahassee, FL 323064510, tam@math.fsu.edu)

The jet noise from a high-performance aircraft at afterburner is investigated. The main objective is to determine whether the dominant noise components are the same or similar to those of a hot supersonic laboratory jet. For this purpose, measured noise data from F-22A Raptors are analyzed. It is found, based on both spectral and directivity data, that there is a new dominant noise component in addition to the usual turbulent mixing noise. The characteristic features of the new noise component are identified. Measured data indicates that the new noise component is observed only when the rate of fuel burn of the engine is increased significantly above that of the intermediate power setting. This suggests that the new noise component is combustion related. The possibility that it is indirect combustion noise generated by the passage of hot spots from the afterburner through the nozzle of the jet is investigated.

3:00–3:20 Break

3:20

4pPA5. Source characterization of full-scale jet noise using vector intensity. Trevor A. Stout, Kent L. Gee, Tracianne B. Neilsen (Phys. and Astronomy, Brigham Young Univ., 688 North 500 East, Provo, UT 84606, titorep@gmail.com), Alan T. Wall (Battlespace Acoust. Branch, Wright-Patterson AFB, Air Force Res. Lab., Dayton, OH), and Michael M. James (Blue Ridge Res. and Consulting, LLC, Asheville, NC)

Vector acoustic intensity has the benefit over pressure measurements in that both the direction and magnitude of energy flow are represented. However, this important quantity has seen little application previously in aeroacoustics. In the present work, an intensity probe captured the radiated field to the sideline and aft of a tethered, full-scale military jet aircraft as one engine was operated at multiple engine conditions. Intensity data from each probe location provide a frequency-dependent map of the sound flow near the aircraft. The vector acoustic intensity is estimated using a recently developed processing technique that extends the traditional upper-frequency limit on estimation accuracy. The intensity vectors are traced back to the jet centerline as a method of approximating the extent and location of the source region as a function of frequency. This method is validated by numerical simulation. As expected for jet mixing noise sources, the resulting source region estimates contract and move upstream with increasing frequency. In addition, the source region at afterburner compared to full-throttle is located about 1 m farther downstream at frequencies above about 300 Hz, and the intensity tends to point more to the sideline by up to 10°. [Work supported by ONR.]

3:40

4pPA6. Analytical intensity calculated from a wavepacket model and comparison to intensity measurements near a high-performance military aircraft. Eric B. Whiting, Trevor A. Stout, Kent L. Gee, Tracianne B. Neilsen (Dept. of Phys. and Astronomy, Brigham Young Univ., N201 ESC, Provo, UT 84602, ericbenwhiting@gmail.com), Alan T. Wall (Air Force Res. Lab., Dayton, OH), and Michael M. James (Blue Ridge Res. and Consulting, LLC, Asheville, NC)

To create an equivalent source description of jet noise, a wavepacket model is used to determine the acoustic vector intensity away from the source. This paper represents an initial investigation into using the measured vector acoustic intensity to define wavepacket parameters. The complex pressure of a line source is defined according to an analytical hyperbolic tangent wavepacket model and Rayleigh integration is used to find the pressure, particle velocity, and time-averaged intensity at observer locations. The parameters that define the shape of the wavepacket are initially based on prior pressure level-based optimization carried out for ground-based microphones. The resulting calculated acoustic intensity vectors are compared to vector intensity measurements previously taken near a tethered high-performance military aircraft at military and afterburner engine conditions. The wavepacket parameters are then varied to provide an optimal fit using the intensity, rather than pressure level, as the cost function, and the differences between the two models are described. [Work supported by the Office of Naval Research.]

3:55

4pPA7. Comparison of holography, beamforming, and intensity-based inverse measurements on full-scale jet noise sources. Part I: Numerical investigations. Alan T. Wall (Battlespace Acoust. Branch, Air Force Res. Lab., Bldg. 441, Wright-Patterson AFB, OH 45433, alantwall@gmail.com), Blaine M. Harker, Trevor A. Stout, Tracianne B. Neilsen, and Kent L. Gee (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT)

The reduction of high-performance fighter aircraft noise, which is of particular concern due to its potential negative impact on communities near air bases and damage to military personnel hearing, is facilitated by improved understanding of the jet noise sources. To this end, near-field acoustical holography, beamforming, and acoustic vector intensity methods have been investigated for the source imaging of full-scale jets. In the first of a two-part investigation, a numerical experiment is performed to test the accuracy of each of these methods, using a simulated measurement array similar to a physical array from an actual full-scale measurement. Extended, partially correlated sources are generated over a range of frequencies using a wavepacket ansatz, and array measurements are simulated in the geometric near field. Source reconstructions are obtained using each of the methods. These equivalent sources are then propagated to the mid and far field. The various reconstructions are compared to simulated benchmarks in order to highlight the advantages and disadvantages of each method, and to evaluate the frequency ranges over which they provide accurate results at the source and in the mid and far fields. [Work supported by USAFRL through ORISE.]

4:10

4pPA8. Comparison of holography, beamforming, and intensity-based inverse measurements on full-scale jet noise sources. Part II: Experimental results. Blaine M. Harker (Dept. of Phys. and Astronomy, Brigham Young Univ., N283 ESC, Provo, UT 84602, blaineharker@byu.net), Alan T. Wall (Battlespace Acoust. Branch, Air Force Res. Lab., Wright-Patterson AFB, OH), Trevor A. Stout, Tracianne B. Neilsen, Kent L. Gee (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT), and Michael M. James (Blue Ridge Res. and Consulting, LLC, Asheville, NC)

Targeted reduction of high-amplitude jet noise is facilitated by accurate source imaging and sound field prediction models. To provide guidance for future jet noise reduction efforts, near-field acoustical holography, beamforming, and acoustic vector intensity-based inverse methods (AVIBIM) have been implemented in efforts to measure and predict full-scale high-performance aircraft noise sources. Patch-and-scan array measurements were taken in the near-field of a tethered tactical aircraft. From these data, equivalent source reconstructions are calculated using each method at multiple frequencies, and the source results are then propagated outward and compared with benchmark locations in the mid and far fields. The frequency ranges where each method's reconstructions predict the measured sound pressure levels are identified and guidelines are provided for the design of measurements to optimize source reconstructions. These results are compared to those obtained in the numerical study (Part I) of this two-part investigation. [Work supported by USAFRL through ORISE.]

Session 4pPP

Psychological and Physiological Acoustics: Pitch, Loudness, and Other Perceptual Phenomena

William M. Hartmann, Chair
Physics and Astronomy, Michigan State University, Physics-Astronomy, 567 Wilson Rd., East Lansing, MI 48824

Contributed Papers

1:30

4pPP1. The perception of vocal-tract length in cochlear implant users: Can it be improved? Etienne Gaudrain (Cognition Auditive et Psychoacoustique, Ctr. de Recherche en NeuroSci. de Lyon, CNRS UMR 5292, Université Lyon 1, UMCG, KNO, Huispostcode BB20, PO box 30.001, Groningen 9700 RB, Netherlands, etienne.gaudrain@cnrs.fr), Nawal El Boghdady, and Deniz Başkent (Dept. of Otorhinolaryngology, Univ. Medical Ctr. Groningen, Groningen, Netherlands)

When trying to understand a speaker among competing speakers, normal-hearing listeners take advantage of differences in the voices of the talkers. However, cochlear implant (CI) users do not seem to benefit from such voice differences. Roy D. Patterson's group in Cambridge has applied psychophysical methods to voice perception in order to isolate the acoustic cues that allow listeners to identify and segregate voices [e.g. Smith, Patterson, Turner, Kawahara and Irino, 2005, J. Acoust. Soc. Am., vol. **117**, pp. 305–318]. In our group, we have extended these methods to CI listeners to fully characterize the limitations CI users experience in utilizing voice characteristics. More specifically, we have shown that while voice pitch (F0) is sufficiently salient through the implant to allow gender categorization, the perception of vocal-tract length (VTL) is severely degraded, leading to erroneous gender perception. With the help of acoustic simulations mimicking the CI processing, the potential factors explaining this lack of sensitivity for VTL were explored. Poor spectral resolution, spectral quantization, and spectral distortion were all found to affect VTL just-noticeable-differences. These results have yielded a number of potential solutions to improve VTL representation — and hence concurrent speech perception — in the implant.

1:45

4pPP2. Probing monaural edge pitch. William M. Hartmann (Phys. and Astronomy, Michigan State Univ., 567 Wilson Rd., East Lansing, MI 48824, hartmann@pa.msu.edu), Xiaochen Li (Dept. of Phys., Xian Jiaotong Univ., Xian, China), and Peter Cariani (Biomedical Eng., Boston Univ., Boston, MA)

Noise with a sharp spectral edge, occurring at an edge frequency f_e, leads to a tonal sensation with a timbre like that of a sine tone and a pitch near the edge frequency. For lowpass noise the pitch is shifted below f_e and for high pass noise the pitch is shifted above. Typical shifts are less than 10% of the edge frequency. To probe this effect, pitch matching experiments were done using 32 different lowpass and highpass noises with edges at 500 and 600 Hz and with durations of about 50 ms. Pitch matches were made in an indefinitely repeating cycle of noises and half-second matching tones. After each match, listeners rated the difficulty of the match. The ratings were ultimately reduced to a five-point scale. The experimental data were compared with the predictions of a model based on the peaks of an all-order population interval distribution as determined by the Zilany-Bruce-Carney model of the auditory periphery. Matching frequencies were predicted from the periodicities of the best fitting subharmonic template and matching difficulty was predicted by the relative goodness of fit. Correlations between experiment and prediction were often high, approaching 0.9. [Work supported by the USAFOSR].

2:00

4pPP3. Acoustic and auditory sketches: Recognition of severely simplified natural sounds by human listeners. Vincent Isnard (Département Action et cognition en situation opérationnelle, Institut de recherche biomédicale des armées, 1 Pl. Igor-Stravinsky, Paris 75004, France, vincent.isnard@ircam.fr), Marine Taffou, Isabelle Viaud-Delmon (Espaces acoustiques et cognitifs, Institut de recherche et coordination acoustique/musique, Paris, France), and Clara Suied (Département Action et cognition en situation opérationnelle, Institut de recherche biomédicale des armées, Brétigny-sur-Orge, France)

Sounds in our environment like voices, animal calls, or musical instruments can be recognized on the basis of timbre alone. The objective of this study was to unravel the features underlying this robust sound recognition. We investigated how severely sounds can be simplified while still being recognizable. A large number of natural sounds were simplified into "auditory or acoustic sketches:" energy peaks were selected on the auditory or acoustic spectrogram, at three levels of simplification (high, medium, and low). The remaining non-sparse information was removed. Listeners were asked to classify the simplified sounds into their four original categories: instruments, birds, vehicles, or voices. We adapted a recent signal detection model to dissociate the perceptual sensitivity (d' scores) from the bias, for each category (4-AFC task). Overall, severely simplified sounds could still be recognized above chance by the listeners. Auditory distances, based on spectro-temporal excitation patterns (STEPs), were then computed. Participants' performances on the 4-AFC task were well correlated with the auditory distances between the four categories. Altogether, our results suggest that sound recognition is a very robust perceptual process, and that basic spectral and spectro-temporal differences between sounds, captured by the auditory distances, can account for this robust recognition.

2:15

4pPP4. Loudness effect on pairwise comparisons and sorting. Patrick Susini, Olivier Houix (R&D, IRCAM, 1 Pl. Igor Stravinsky, Paris 75004, France, susini@ircam.fr), and Guillaume Saint Pierre (IFSTTAR, Paris, France)

Effect of loudness on the perceptual structure underlying a corpus of sounds is investigated by two experimental methods: pairwise comparisons and sorting. Both methods are applied to a corpus of recordings sounds presented with their ecological, non-normalized loudness, and to the same corpus equalized in loudness. Two types of perceptual structures (multidimensional scaling and hierarchical cluster analysis) are derived. Domination of one auditory attribute—loudness—on less salient ones is discussed according to the two types of perceptual structures. In the non-loudness-normalized corpus, loudness was the main factor explaining participant's judgments for both tasks. In particular, representations derived from sorting data almost solely reflected sound pressure level difference between sounds. On the opposite, loudness normalization allowed the emergence of several predicted auditory attributes that characterize the tested sounds. On that second corpus however, sorting data were found less appropriate than pairwise data to provide interpretable continuous dimensions.

2:30

4pPP5. Neuromagnetic correlates of the vocal characteristics of vowels in auditory cortex. Roy D. Patterson (Physiol., Development and Neurosci., Univ. of Cambridge, Downing Site, Cambridge CB2 3EG, United Kingdom, rdp1@cam.ac.uk), Martin Andermann (Section of Biomagnetism, Dept. of Neurology, Univ. of Heidelberg, Heidelberg, Germany), Stefan Uppenkamp (Medizinische Physik, Univ. of Oldenburg, Oldenburg, Germany), and André Rupp (Section of Biomagnetism, Dept. of Neurology, Univ. of Heidelberg, Heidelberg, Germany)

As a child grows up, the formants of their vowels move down in frequency and the spacing of the harmonics, which determines voice pitch, decreases. Perceptually, these variables determine who we hear speaking (child, woman, or man). In a logarithmic-frequency spectrum, the envelope with its formant peaks moves toward the origin as a unit, without changing shape, as a child grows up. Similarly, the harmonics, which constitute the fine structure of the spectrum, move toward the origin as a unit without changing shape, but at a different rate. This paper describes neuromagnetic studies which show that the generator associated with voice pitch is in Heschl's gyrus just lateral to primary auditory cortex, while the generator associated spectral envelope position is in planum temporale, some distance behind the pitch activity generator. The posterior generator is close to the location of the large N1m that typically accompanies the onset of acoustic energy of any sort. The pitch processing component of the N1m was isolated from the energy-onset component by presenting sequences of vowels with minimal inter-vowel intervals; the pitch component appears isolated in the responses to the second and succeeding vowels.

2:45–3:00 Break

3:00

4pPP6. Incidental absolute pitch learning in an interactive multi-modal environment. Shannon L. Heald, Stephen C. Van Hedger, and Howard C. Nusbaum (Psych., Univ. of Chicago, 5848 S. University Ave., c/o Shannon Heald (B406), Chicago, IL 60637, smheald@gmail.com)

Absolute pitch (AP), the ability to label an isolated note without the aid of a reference note, is to some degree trainable in an adult population. Most training studies ask participants to make overt categorization judgments and provide explicit feedback. Here, we examined the incidental learning of four isolated notes in an interactive multi-modal environment. Participants played a video game, whereby four characters, requiring different actions, were marked by specific note classes. Half of the participants experienced an invariant mapping, such that the same perfectly in-tune note always marked each character. The other half experienced a variant mapping, where a distribution of notes that varied in intonation marked each character (the mode of which was an in-tune note). While participants could anticipate a character via its associated note class, such anticipation was not required or encouraged. In a follow-up task, participants matched the characters to their associated notes. Results showed significant rote learning for the character-note pairings with above-chance generalization to novel timbres and octaves. The variant condition mitigated the performance gap found between those with and without video game experience in the invariant condition. Results will be discussed in terms of the mechanisms involved in implicitly acquiring AP categories.

3:15

4pPP7. Some factors influencing loudness asymmetries between rising and falling-intensity stimuli. Emmanuel Ponsot (IRCAM, Paris, France), Sabine Meunier (LMA-CNRS, Marseille, France), and Patrick Susini (IRCAM, 1 Pl. Igor Stravinsky, Paris 75004, France, susini@ircam.fr)

Previous research demonstrated that the loudness asymmetry between 1-kHz rising and falling-intensity tones is a robust phenomenon, whose origins still remain unclear. In the present study, this phenomenon was further examined as a function of two stimuli characteristics: the spectral content and the intensity-region. In a first experiment, the global loudness of rising and falling-intensity sounds with various spectral contents (pure tones from 250 Hz to 8 kHz and broadband noises) presented in different intensity-regions (from [50–65 dB SPL] to [70–85 dB SPL]) was assessed in an absolute magnitude estimation task. Significant asymmetries were found for tones at all frequencies, but not for broadband noises. In addition, a significant interaction between the stimulus direction and the intensity-region was observed for both tones and noises. This interaction was further examined in a second experiment using an adaptive loudness-matching procedure. Although greater asymmetries were again observed for tones, significant asymmetries were found for noises as well. Furthermore, the size of the asymmetries was significantly decreased with the intensity-region when the pairs were composed of rising followed by falling stimuli. These results are discussed in the light of recent physiological and neuroscience studies conducted with this type of stimuli.

3:30

4pPP8. The perception of vocal characteristics in normal-hearing and cochlear-implanted children. Deniz Baskent (UMCG Otorhinolaryngology, Univ. of Groningen, Univ. Medical Ctr. Groningen, PO Box 30.001, Groningen 9700RB, Netherlands, d.baskent@umcg.nl), Jacqueline Libert (Res. School of Behavioral and Cognit. Neurosci., Univ. of Groningen, Groningen, Netherlands), Deborah Vickers (Ear Inst., Univ. College London, London, United Kingdom), and Etienne Gaudrain (Otorhinolaryngology, Univ. of Groningen, Univ. Medical Ctr. Groningen, Groningen, Netherlands)

Voice characteristics provide important cues for many speech-related tasks, such as gender categorization, vocal emotion perception, and understanding speech in the presence of competing voices; tasks known to be difficult for cochlear implant (CI) users. While most related research has focused on voice fundamental frequency (F0), the work from Smith *et al.* [J. Acoust. Soc. Am., vol. **117**, pp. 305–318, 2005] has shown the importance of another voice characteristic: the vocal tract length (VTL; related to speaker size). Recent work has indicated that while CI users have difficulty in utilizing F0, they are even more limited at VTL perception. This study aimed to further understand factors potentially contributing to this limitation, by exploring F0 and VTL perception in NH children and children with CIs. We hypothesized that children may learn to utilize F0 before VTL cues because of exposure to exaggerated F0 cues in infant-directed-speech, and acquisition of VTL cues may be later, due to the necessity for exposure to multiple talkers; implying a hierarchy in processing the two vocal cues. We further hypothesized that children with CIs may be better at processing voice cues than adult-implanted CI users, suggesting that these perceptual cues could be learned.

3:45

4pPP9. Sound source localization for filtered noises when listeners rotate. William Yost (ASU, PO Box 870102, Tempe, AZ 85287, william.yost@asu.edu)

Yost and Zhong (2015, JASA, 137, 2200) recently showed that sound source localization when listeners rotate depends on both spatial cues and information about the world-centric location of the listener. In these experiments, the sound was a broad-band (0.125—15 kHz), 200-ms noise burst. This stimulus does not allow for an estimation of the role of interaural time (ITD) and/or level (ILD) differences in sound source localization when listeners rotate. The present experiment used the following stimuli: low-pass noise (0.125–0.5 kHz) implicating ITD cues, high-pass noise (2–8 kHz) implicating ILD cues, broad-band noise (0.125–8 kHz) implicating both ITD and ILD cues, and mid-frequency noise (1–4 kHz) for which neither an ITD nor an ILD cue provides good information about sound source location. Listeners rotated at constant velocity (45o/sec) in the azimuth plane, while sounds changed position around a 24-loudspeaker array in the same azimuth plane. Listeners responded in an eyes-open condition in which they would receive information about their world-centric location and in an eyes-closed condition in which they would not. The task was to decide if two successive noise bursts were presented from the same or different loudspeakers as listeners rotated. [Research funded by an AFOSR grant.]

4:00

4pPP10. Verification of an automated headphone-based test of spatial release from masking. Frederick J. Gallun (Otolaryngology/Head&Neck Surgery, Oregon Health and Sci. Univ., 3710 SW US Veterans Hospital Rd., Portland, OR 97239, Frederick.Gallun@va.gov), Sean D. Kampel (National Ctr. for Rehabilitative Auditory Res., VA Portland Health Care System, Portland, OR), Kasey M. Jakien, Nirmal K. Srinivasan (Otolaryngology/Head&Neck Surgery, Oregon Health and Sci. Univ., Portland, OR), Meghan M. Stansell, and Samuel Y. Gordon (National Ctr. for Rehabilitative Auditory Res., VA Portland Health Care System, Portland, OR)

Currently, there are many different laboratory-based tests of spatial release from masking (SRM) that use speech materials; however, there is still disagreement as to the impact of age and hearing loss on SRM. The time is ripe, then, for taking these tests out of the laboratory and testing larger numbers of listeners varying in age and hearing ability in order to provide the statistical power needed to answer the questions currently being asked. Unfortunately, most of the tests that have been developed are either open set, and thus require a tester to administer them, or require complex soundfield speaker arrays. Our laboratory has recently developed and verified an automated headphone-based test that can be presented in only five to ten minutes and that provides results that are predictive of results obtained in an anechoic chamber. The data associated with the verification of this test procedure will be presented.

THURSDAY AFTERNOON, 5 NOVEMBER 2015 — DAYTONA, 2:45 P.M. TO 4:35 P.M.

Session 4pSA

Structural Acoustics and Vibration: Novel Treatments in Vibration Damping

Kenneth Cunefare, Chair
Georgia Tech, Mechanical Engineering, Atlanta, GA 30332-0405

Invited Papers

2:45

4pSA1. Vibration damping and isolation using negative stiffness structures. Michael R. Haberman, Carolyn C. Seepersad, and Preston S. Wilson (Appl. Res. Labs. and Dept. of Mech. Eng., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, haberman@arlut.utexas.edu)

This work presents a novel class of engineered structures with significant promise to improve vibration damping treatments and isolation systems: negative stiffness (NS) elements. A mechanical system displaying negative stiffness is characterized by a loading state that requires a decreasing force level to increase the deformation of the system. Systems displaying NS will possess regions of negative curvature in their strain energy response as a function of deformation; hence, they are unstable when unconstrained. Analytical and experimental results will be presented demonstrating that NS systems comprised of buckled beams in parallel with positive stiffness springs can be used to construct quasi-zero stiffness vibration isolation systems, which provide high static but low dynamic stiffness for compact base isolation design. Transmissibility measurements of these same systems show that the nonlinearity of NS systems constructed from buckled beam structures enable tunable vibration isolation behavior and isolation from impact. Finally, modeling results will be presented demonstrating that sub-wavelength NS elements embedded in a viscoelastic material can be used to design vibration damping treatments with increased loss factor and minimally reduced stiffness to reduce the ringdown time for an impulsively loaded multi-layered beam. [Work supported by DARPA, ARO, and NSF.]

3:05

4pSA2. Periodically distributed piezoelectric patches optimization for waves attenuation and vibrations damping. Manuel Collet, Yu Fan, and Mohamed Ichchou (Dynamic of Complex systems, CNRS LTDS, Ecole Centrale de Lyon, 36 av G. de Collongue, Ecully 69131, France, manuel.collet@ec-lyon.fr)

The deformation of a structure can be understood as a superposition of waves which are induced by the excitation and reflected by the boundaries. According to this regard, attenuations of waves can lead to a strong reduction of the structural response and prevent energy to be propagated. In this work we consider periodically distributed piezoelectric patches onto the host structure so that by designing shunted electric circuits the properties of the waves can be modified. The success of the idea is also directly related to the extent of electromechanical coupling. In terms of structural modes, the coupling factor can be estimated by the open-circuit (OC) and short-circuit (SC) natural frequencies. However, in terms of waves, few criteria are available. In this work, a wave-based criterion is proposed to evaluate the coupling factor of the piezoelectric composite. To do this, enhanced Wave and Finite Element Method (WFEM) is employed to obtain the dispersion relations and the shapes of the waves. Then, the factor can be calculated in three different but equivalent formulas. An example is given thereafter, where a piezoelectric waveguide with semi-active circuits is used to control the energy flow from a source to the far-field. We show that the coupling factor is frequency dependent and it is strongly related to the geometric parameters; therefore, it significantly changes the optimal performance of the piezoelectric waveguide and it capacity to dampen vibrations.

3:25

4pSA3. Managing property distribution errors in arrays of coupled resonators. Joseph F. Vignola (Mech. Eng., The Catholic Univ. of America, 620 Michigan Ave., NE, Washington, DC 20064, vignola@cua.edu), Andrew J. Kurdila (Mech. Eng., Virginia Polytechnic and State Univ., Blackburg, VA), John Sterling (Naval Surface Warfare Ctr., Carderock Div., West Bethesda, MD), John A. Judge (Mech. Eng., The Catholic Univ. of America, Washington, DC), Aldo A. Glean (CertainTeed Corp., Northboro, MA), and Teresa J. Ryan (Eng., East Carolina Univ., Greenville, NC)

Arrays of small attachments can be designed to draw mechanical energy from a primary structure in a manner far in excess of their proportional size. Earlier work has shown that slight variations or errors in the property distributions of arrays of coupled resonators can have a dramatic effect on the response of primary structure to an external force. This work investigates the use of an electro-mechanical approach to making small adjustments in the stiffness of the individual elements of the array to alter the response of the primary. The electro-mechanical coupling is achieved by way of laminated thin piezoactuators mounted on a fraction of the subordinate elements. The piezoactuators are electrically coupled to a switching network that changes the effective stiffness of the subordinate elements. This ability to adjust the stiffness distribution facilitates real time control of the rate at which the energy is transferred into the coupled array. This apparent damping can then be adjusted to draw or reject energy from specific frequency bands. The presentation will describe the underling theory, present numerical results, and some preliminary experimental results.

3:45

4pSA4. Removable damping treatments for use in sheet metal fabrication. Kenneth A. Cunefare (Georgia Tech, Mech. Eng., Atlanta, GA 30332-0405, ken.cunefare@me.gatech.edu)

Many industrial fabrication processes on metallic structures generate a great deal of work place noise. For example, riveting and chiseling on light-weight, stiff aerospace structures such as fusalage components and flight surfaces subject the workpiece to repeated impacts and consequent noise generation. Many such structures may be lightly damped. Many damping treatments are targeted toward permanent installation on a structure, but if the treatments are considered to be removable, one has design freedoms including placement on a structure, as well as mass and damping means, that might not otherwise exist if the treatment had to remain in place. A shot-mass or particle-impact damper configured in an elastic, cellular array is shown to provide high loss factors and mass loading, which, along with ease of placement and removal, may provide significant reduction in component vibration during fabrication and consequent reduction in noise in the work place. The concept applied to an example aircraft panel when subjected to riveting operations yields more than 10 dB of noise reduction.

Contributed Papers

4:05

4pSA5. Solid-liner suppressor design, construction, and development. Ryan Salmon and Kenneth A. Cunefare (Mech. Eng., Georgia Inst. of Technol., 771 Ferst Dr., Atlanta, GA 30332, rsalmon3@gatech.edu)

Noise in a fluid system can be treated with a prototypical liner-style suppressor, an expansion chamber which includes an internal annulus of syntactic foam. A syntactic foam liner consists of host material with hollow microspheres which collapse under pressure to add compliance to the suppressor. The liner effectively increases the transmission loss of the suppressor, or ratio between inlet and outlet acoustic energy. Currently, liner-style suppressors are not commercially available. This study will investigate the integration of solid liner material within suppressor shells while also analyzing the effect of flow-smoothing diffusors on the transmission loss of the suppressor. The diffusors function to center the liner within the device, while reducing the potential for turbulence-induced self-noise. The diffusor may also impact the longevity of the liner, by reducing mechanical erosion. The results of the study should provide additional insight to the commercial viability of the liner-style suppressor.

4:20

4pSA6. Investigation of hysteresis friction in elements under complex stress state. Smirnov Vladimir, Ilya Tsukernikov, Igor Shubin, and Nina Umniakova (NIISF RAABS, Jaroslavskoe shosse, 26, Moscow 129337, Russian Federation, belohvost@list.ru)

Recent studies have indicated that the best type of damping in vibration isolation system is hysteresis (internal) damping, which effectively reduces vibration amplitudes at resonance, and it does not increase as compared with viscous damping vibration amplitudes after the resonance. Modern damping investigation methods are based on the experimental determination of the loss factor for a certain form of vibration isolators with fixed dimensions and loading parameters. At the same time, with the emergence of complex nonlinear vibration isolators, such as discussed by the authors, there is a task of expanding this theory on the case of complex designed vibration isolators. In this paper, we consider the problem of calculating the loss factor in the beam—columns of variable cross-section under complex loading conditions. A method for calculating the amount of losses in the nonlinear vibration isolators is based on Panovko's energy theory, which consists of experimental loss factor determination in the material of the vibration isolator and subsequent calculation of the loss factor in the isolator in view of obtained data.

Session 4pSCa

Speech Communication and Signal Processing in Acoustics: Advancing Methods for Analyzing Dialect Variation

Cynthia G. Clopper, Chair
Ohio State University, 222 Oxley Hall, 1712 Neil Ave, Columbus, OH 43210

Chair's Introduction—2:00

Invited Papers

2:05

4pSCa1. Reconceptualizing the vowel space in analyzing regional variation. Robert A. Fox and Ewa Jacewicz (Speech and Hearing Sci., The Ohio State Univ., 110 Pressey Hall, 1070 Carmack Rd., Columbus, OH 43210-1002, fox.2@osu.edu)

Vowel space area calculated on the basis of the corner vowels has emerged as a metric for the study of regional variation, speech intelligibility, and speech development. We verify the basic assumptions underlying both the concept of the vowel space and the utility of the vowel space area in making speaker, dialect, or language comparisons. Undeniably, the traditional vowel triangle and vowel quadrilateral both fail as a metric in the context of dialect variation because substantial parts of the actual working space are excluded from analysis. Utilizing the formant values at a number of different locations for a wider range of individual vowels has significant implications for the size and shape of the resulting vowel space. Indeed, dialectal variations in vowel production can best be characterized in terms of formant density regions in the formant space and not as locations of individual vowel categories. The formant density approach is based on the assumption that vowel sounds are dynamically changing multidimensional units, which naturally overlap in the acoustic space. The formant density approach is able to minimize the amount of empty space within the overall shape while still respecting the outer boundary of the dataset.

2:25

4pSCa2. Eliciting comparable, natural speech from children and adults. Elizabeth A. McCullough (Psych., Ohio State Univ., 1835 Neil Ave., Columbus, OH 43210, eam@ling.ohio-state.edu), Cynthia G. Clopper (Linguist, Ohio State Univ., Columbus, OH), and Laura Wagner (Psych., Ohio State Univ., Columbus, OH)

In sociophonetic studies, the goal of eliciting natural speech is often at odds with the goal of eliciting speech that is easily comparable across participants. In this examination of regional pronunciation variation among English-speaking children and adults in the United States, these goals were balanced using two speech elicitation methods: color naming and picture-prompted storytelling. In the color naming task, participants saw solid blocks of color on a computer screen and were recorded saying the name of each color. In the storytelling task, participants were recorded narrating the familiar stories of *Little Red Riding Hood* and *Goldilocks and the Three Bears* using self-paced picture prompts. These methods were successfully used with participants of a wide range of ages, from pre-literate children (4 years old) to older adults (75 + years old). The color naming task yielded largely identical target words across participants because only common colors were presented. While the storytelling task allowed participants to be more creative, targets such as character names, words for important objects in the stories, and canonical lines of dialog were repeated frequently across participants. Thus, these elicitation methods avoided naturalness concerns associated with read speech while providing identical word-length and similar sentence-length examples for comparison.

2:45

4pSCa3. United States accents compared: The relation of acoustic distances and perceptual tasks. Clelia R. LaMonica (Dept. of English, Stockholm Univ., Stockholm 106 91, Sweden, clelia.lamonica@english.su.se)

This project investigates correlations between accent production and perception by comparing two sets of data: acoustic distances of American dialectal speech samples, measured using Euclidean distances of each vocalic stimuli's first three formants across a trajectory, and perceptual data in the form of online survey material and EEG tests which judge the differences between the same varieties. The speech samples used consist of six sentences from eight regions of the United States; each sentence containing phonological features that may be marked as perceptually relevant for dialect classification. Here, we examine the preliminary outcomes from such perceptual tests performed by naïve listeners, which include free classification, identification, perceived difference and ranking similarity, as well as attitude judgment tasks. Additionally, EEG tests were carried out to evaluate the relation between acoustic distance of accents and ERPs. Of special interest here is the relation between a non-regionally specified American accent and others. This "standardized" variety is judged in comparison with each of the regional accents in order to investigate listeners' perceptions of non-regional vs. regional accents, and in turn the correlation between measured accent (dis)similarity and perception.

3:05

4pSCa4. Using ultrasound articulatory signals to investigate the phonetic motivations of English /æ/ tensing. Jeff Mielke, Erik R. Thomas (English, North Carolina State Univ., 221 Tompkins Hall, Campus Box 8105, Raleigh, NC 27695-8105, jimielke@ncsu.edu), and Christopher Carignan (The MARCS Inst., Univ. of Western Sydney, Penrith, NSW, Australia)

A common simplifying technique in ultrasound studies of variation is to select a single representative frame for each token, sacrificing dynamic information that is often critical for understanding the phonetic motivations of phonological phenomena. We examine the phonetic motivations for tongue body raising in English /æ/ tensing (e.g., Labov *et al.* 2005) in 23 North American English speakers using phonetically meaningful time-varying articulatory signals extracted directly from ultrasound video. An articulatory measure of /æ/ tenseness is generated using regression to find the linear combination of articulatory principal components (found using EigenTongue Feature Extraction; Hueber *et al.* 2007) that best accounts for the F2-F1 difference in front vowels. We have previously shown (Carignan *et al.* 2015) that tensing before /m n/ involves tongue body raising that is timed to the vowel nucleus, whereas tensing before /ɡ ŋ/ involves anticipating the velar closure to different degrees in different dialects. Here, we examine the phonetic motivations for this /æ/ tensing. An articulatory measure of velar fronting shows increased /ɡ/ fronting in speakers who tense /æ/ before /ɡ/, and a purely lingual analog of F1 shows that the effect of nasalization on F1 is much smaller than the phonological effect of pre-nasal /æ/ tensing.

3:25

4pSCa5. Modeling consonant-context effects in dialectal variation in a large database of spontaneous speech recordings. Michael Kiefte (Human Commun. Disord., Dalhousie Univ., 1256 Barrington St., Halifax, NS B3J 1Y6, Canada, mkiefte@dal.ca) and Terrance M. Nearey (Linguist, Univ. of AB, Edmonton, AB, Canada)

Given recent interest in the analysis of naturally produced spontaneous speech, we collected, processed, and analyzed a large database of speech samples from the Canadian province of Nova Scotia with the primary aim of examining regional variation in vowel-formant patterns. Although the actual collection of audio recordings is relatively straightforward, the analysis of spontaneous speech suffers from several disadvantages relative to that of laboratory, citation speech: Different vowels have different frequencies of occurrence, surrounding consonants have a large influence on formant peak frequencies, and the distribution of consonant contexts across different vowels is highly unbalanced. To overcome these problems, we developed a statistical procedure inspired by that of Broad and Clermont [1987, J. Acoust. Soc. Am., 81, 155] to estimate the specific effects of both onset and coda consonant-context effects on vowel formant frequencies. However, in contrast to their procedure, both vowel formant frequencies and consonant-context effects were allowed to vary freely across the duration of the vocalic portion of a syllable. Thirty-five hours of recorded speech samples from 223 speakers were automatically segmented and formant-frequency values were calculated for all stressed vowels in the database. Consonant effects were factored out to produce context-independent vowel-formant frequencies that varied across time. These data can then be used to examine dialectal variation in vowel production throughout the region.

3:45

4pSCa6. Voices of coastal Georgia. Margaret E. Renwick and Rachel M. Olsen (Linguist Program, Univ. of Georgia, University of Georgia, 240 Gilbert Hall, Athens, GA 30602, mrenwick@uga.edu)

The Linguistic Atlas of the Gulf States (LAGS; http://www.lap.uga.edu/Site/LAGS.html) contains sociolinguistic interview data from 914 speakers collected from 1968 to 1983. Impressionistic transcriptions of single words and phrases contributed to the dialectal description and mapping of the southern United States, but without systematic acoustic analysis. We gather recordings of target lexical items from ten LAGS speakers in Georgia's coastal region, focusing on data collected near St. Marys in four counties (Camden, Glynn, Charlton, and Ware). When interviewed in 1972, speakers ranged from 23 to 80 years (mean 63.7; 5M, 5F); the data thus represent dialect features of the early-mid-20th Century, including the Gullah-speaking community of St. Simons Island. The analysis focuses on features known to characterize the dialect of this region: the monophthongal vowel space, the degree of vowel diphthongization, vowel mergers before /l/ and nasals, and rhoticity of the speaker's dialect. Acoustic measurements are automatically extracted for comparison across speakers. The recordings (1 to 9 hours/speaker) were digitized as .wav within the Linguistic Atlas Project (LAP; Kretzschmar 2011). Recordings collected for the LAP are an under-attended resource that will serve as a valuable comparison to contemporary studies of regional variation in the southern United States.

Session 4pSCb

Speech Communication: Contributions to the Special Sessions (Poster Session)

Cynthia G. Clopper, Cochair
Ohio State University, 222 Oxley Hall, 1712 Neil Ave, Columbus, OH 43210

Mary E. Beckman, Cochair
Linguistics, Ohio State University, 222 Oxley Hall, 1712 Neil Ave, Columbus, OH 43210-1298

Authors will be at their posters from 4:05 p.m. to 5:00 p.m. To allow authors an opportunity to view other posters in their session, all posters will be on display from 2:00 p.m. to 5:00 p.m.

Contributed Papers

4pSCb1. How speaker identity interacts with perceptual judgments in children with residual sound errors. Sarah Hamilton, Suzanne E. Boyce, Noah Silbert, and Kirsten Mosko (Dept. of Comm. Sci. and Disord., Univ. of Cincinnati, Mail Location 379 University of Cincinnati, Cincinnati, OH 45267, Suzanne.Boyce@uc.edu)

Recent research suggests that listeners store complex phonetic representations when learning speech. Encoding fine perceptual details, such as indexical features of the talker's voice, appears to influence performance in a variety of ways (e.g., processing speed and intelligibility of words in noise). Children with residual sound errors (RSE) for /r/ have been shown to have difficulty judging productions of /r/ from other child speakers along a normalized continuum of third formant values. We hypothesized that children with RSE may make more accurate judgments if they are given stimuli with more familiar indexical characteristics (i.e., their own speech) along the same continuum. In this study, we presented 15 children with a range of stimuli recorded from their own productions as well as productions from other children. In a forced-choice task, children indicated if the word contained a "correct" /r/. Responses to stimuli were compared across children. Initial results suggest that for RSE children, hearing one's own speech does not improve accuracy in judging the correctness of sounds in words.

4pSCb2. Using automatic alignment on child speech: Directions for improvement. Thea Knowles (Univ. of Western ON, Elborn College, Rm. 1510, School of Commun. Sci. and Disord., London, ON, Canada, thea.knowles@gmail.com), Meghan Clayards, Morgan Sonderegger, Michael Wagner, Kris Onishi, and Aparna Nadig (McGill Univ., Montreal, QC, Canada)

Phonetic analysis is labor intensive, limiting the amount of data that can be considered. Automated techniques (e.g., forced alignment based on Automatic Speech Recognition, ASR) have recently emerged allowing for larger-scale analysis. While forced alignment can be accurate for adult speech (e.g., Yuan & Liberman, 2009), ASR techniques remain a challenge for child speech (Benzeghiba *et al.*, 2007). We used a trainable forced aligner (Gorman *et al.*, 2011) to examine the effect of four factors on alignment accuracy with child speech: (1) Datasets CHILDES (McWhinney, 2000):—Spontaneous speech (single child)—Picture naming (multiple children, Paidologos data); (2) Phonetic Transcription—Manual—Automatic—CMU dictionary (Weide, 1998); (3) Training data—Adult lab data—one dataset of child data—All child data—Child & adult lab data; (4) Segment—voiceless stops—voiceless sibilants—vowels Automatically generated alignments were compared to hand segmentations. While there were limits on accuracy, in general, better results were obtained with (1) picture naming, (2) manual phonetic transcription, (3) training data including child speech, and (4) voiceless stops. These four factors increase the utility of analyzing children's speech production using forced alignment, potentially allowing researchers to conduct larger-scale studies that would not otherwise be feasible.

4pSCb3. The differential development of vowel context effects on sibilant fricatives. Patrick Reidy (Commun. Sci. & Disord., Univ. Wisconsin–Madison, 24A Oxley Hall, 1712 Neil Ave., Columbus, OH 43210, patrick.francis.reidy@gmail.com)

Previous work has found that vowel-context effects on the static acoustic properties of sibilant fricatives weaken as children age. The current study extended this prior work by analyzing the development of context effects on the spectral dynamics of English sibilants. Native adults and children (2 through 5 years old) produced /s, ʃ/ in a range of pre-vocalic contexts (/i, e, ɑ, o, u/). Effects of vowel rounding and vowel height were investigated through two pyschoacoustic measures computed from auditory spectra: peak ERB number and excitation drop (difference between maximum high-band and minimum low-band excitation). These measures were estimated from 17 20-ms windows spaced evenly across each production. Effects of vowel context on the intercept and shape of the resulting 17-point trajectories were analyzed with polynomial growth-curve models. Context effects were found to differentially weaken or strengthen in children depending on whether it was the intercept or the shape of the trajectory that was affected. For both sibilants, rounding and height effects on the peak-ERB and excitation-drop intercepts generally weakened with age; whereas, height effects on the shape factors of both trajectories tended to strengthen with age. Future work will extend the analysis to Japanese /s, ɕ/.

4pSCb4. Working memory problems of the elderly arise in the central processor, not the phonological loop. Susan Nittrouer (Speech, Lang., and Hearing Sci., Univ. of Florida, 915 Olentangy River Rd., Ste. 4000, Columbus, OH 43212, nittrouer.1@osu.edu) and Joanna H. Lowenstein (Speech, Lang., and Hearing Sci., Univ. of Florida, Gainesville, FL)

Declines in cognitive and communicative capacities can negatively impact quality of life for even healthy aging individuals. One such capacity involves working memory, which is modeled as consisting of a phonological loop that recovers phonological structure for storage and a central executive that processes that structure. In this study, working memory was investigated to evaluate which component is responsible for age-related declines. Two groups of listeners with good hearing participated: 20 young adults (18 to 32 years) and 25 elderly adults (60 to 80 years). Accuracy and speed of recall were measured for forward and backward digits, and for three sets of CVC words: nonrhyming nouns, rhyming nouns, and nonrhyming adjectives. Phonological awareness was also assessed. Results showed no differences in digit span, but significant age-related differences in accuracy of

word recall and in speed of recall for both digits and words. Phonological awareness did not differ across groups. When speed of recall was used as a covariate, effect size of age on word recall diminished. It was concluded that the problems observed in cognitive and communicative functioning of the elderly can be attributed to a slowing of processing, not to a diminishment in sensitivity to phonological structure. [NIH R01-DC000633.]

4pSCb5. Speech production in the later years: Changes in fundamental frequency and speech breathing. Eric J. Hunter, Simone Graetzer (Dept. of Communicative Sci., Michigan State Univ., 1026 Red Cedar Rd., East Lansing, MI 48824, ejhunter@msu.edu), and Ethan J. Hunter (Haslett Middle School, Haslett, MI)

With age, vocal quality can be affected by changes in articulatory, velopharyngeal, laryngeal, and breathing function. For example, past studies have shown that the voice is sensitive to hormonal changes. Other studies have indicated that there is atrophy of the vocal folds and a hardening of the laryngeal cartilages in old age. In the current study, recordings of three female adults (spanning 18 to 30 years) and three male adults (spanning 38 to 48 years, with one individual being 98 years old at the latest recording) were acquired and analyzed for changes in speech production. For a representative segment of each recording, the temporal boundaries of speech breath groups were identified by raters so that the durations of groups could be calculated. Additionally, the effect of age on speech fundamental frequency was considered. Finally, how fundamental frequency changes during a long speech task was examined, to track vocal fatigue. Results suggested a decrease in breath group duration for most subjects as age increased. Importantly, the effect of age on fundamental frequency seemed to change more than is commonly discussed (based on cross-sectional studies). Finally, there was a gender difference in how both breathing and fundamental frequency changed with age.

4pSCb6. Using vowel trajectories for southern U.S. monophthongization. Paul E. Reed (Linguist, Univ. of South Carolina, 909 Welsh Humanities Bldg., Columbia, SC 29208, reedpe@email.sc.edu)

It is widely known that monophthongization of the diphthong /ɑɪ/ is a feature of Southern U.S. English (e.g., Labov, Ash, and Boberg 2006). However, most studies of this phenomenon only use two measurement points, one from the onset and one from the glide, typically around 25% and 75% of the token's duration. While informative about whether or not a particular vocoid is monophthongal, this type of measure does not permit distinguishing among differing types of monophthongal realizations. Research including varieties from the Southern US (cf. Thomas 2000) has found that the trajectory of the vocoid can differentiate social groups. In this investigation, I track the monophthongal realizations from 24 speakers (12 men, 12 women) from Northeast Tennessee, balanced for age and education. I measure each token at 10ms intervals for both F1 and F2 for the entire vocalic duration. Plotting these measures allows for visualization and comparison of the entire trajectory of the articulation. In preliminary work in this population, flatter trajectories inversely correlate with social factors such as age, education, and local attachment. More finely nuanced measures, such as the whole vocalic trajectory, can capture social variation that may be lost when only comparing a few measurement points.

THURSDAY AFTERNOON, 5 NOVEMBER 2015 — CITY TERRACE 7, 1:30 P.M. TO 3:15 P.M.

Session 4pSP

Signal Processing in Acoustics: Algorithm, Analysis, and Beamforming

John R. Buck, Chair
ECE, UMass Dartmouth, 285 Old Westport Rd, North Dartmouth, MA 02747

Contributed Papers

1:30

4pSP1. Frequency-sum beamforming in a random scattering environment. Shima H. Abadi (Lamont–Doherty Earth Observatory, Columbia Univ., 122 Marine Sci. Bldg., University of Washington 1501 NE Boat St., Seattle, Washington 98195, shimah@ldeo.columbia.edu), David C. Leckta (Mech. Eng., Univ. of Michigan, Ann Arbor, MI), Karla Mercado, Kevin J. Haworth (Dept. of Internal Medicine, Univ. of Cincinnati, Cincinnati, OH), and David R. Dowling (Mech. Eng., Univ. of Michigan, Ann Arbor, MI)

In a uniform environment, sound propagation direction(s) or the location of a sound source may be determined from array-recorded signals by beamforming. However, the beamforming results may be degraded when there is random scattering between the source and the receivers. Such sensitivity to mild scattering may be altered through use of an unconventional beamforming technique that manufactures higher frequency information from lower-frequency signal components via a quadratic product of complex signal amplitudes. This presentation will describe frequency-sum beamforming, and then illustrate it with simulation results and near-field acoustic experiments. The simulations suggest that frequency-sum beamforming may be beneficial when there is one loud source and the environment provides one primary propagation path. The experiments were conducted in either a 1.0-m-deep 1.07-m-diameter cylindrical water tank using 50 kHz and 100 kHz signals broadcast from a single source to an array of 16 hydrophones when discrete scatterers are present and absent from the tank or in a tissue-mimicking phantom with a dominant scatterer embedded and insonified at 2 MHz and the scatter received by a 128-element array. The results from frequency-sum beamforming are compared to the output of conventional delay-and-sum beamforming and minimum variance beamforming. [Work supported by NAVSEA through the NEEC.]

1:45

4pSP2. Robust plane-wave decomposition of spherical microphone array recordings for binaural sound reproduction. David L. Alon, Jonathan Sheaffer, and Boaz Rafaely (Elec. and Comput. Eng., Ben-Gurion Univ. of the Negev, P.O.B. 653, Beer-Sheva 84105, Israel, davidalo@post.bgu.ac.il)

Rendering binaural signals from spherical microphone recordings is becoming an increasingly popular approach, with applications in telecommunications, virtual acoustics, hearing science, and entertainment. Such binaural signals can be generated from a plane-wave decomposition of a sound field measured by a spherical microphone array. This process may exhibit ill-conditioned transformations when performed at low frequencies and using high spherical-harmonics orders, thus resulting in a poor robustness to measurement inaccuracies and noise. Previous studies have addressed this issue by employing standard regularization techniques, such as diagonal loading and radial filter limiting. In this paper, we propose an optimal frequency-dependent regularization method that balances system robustness to measurement noise against accuracy of plane-wave decomposition. Unlike previously suggested approaches, the proposed method analytically relates the measured signal-to-noise ratio to the corresponding regularization parameters, hence facilitating a means for controlling the regularization process using a closed-form expression. The method is compared to previously suggested regularization techniques in terms of spatial, temporal, and spectral effects on the resulting binaural signals. Objective results, which illustrate the improved performance of the proposed method, are complemented with a subjective validation of the regularized signals.

2:00

4pSP3. Double zero minimum variance distortionless response beamformer. Saurav R. Tuladhar and John R. Buck (ECE Dept., UMass Dartmouth, 285 Old Westport Rd., North Dartmouth, MA 02747, johnbuck@ieee.org)

Adaptive beamformers (ABFs) place deep beampattern notches near interferers to suppress the interferers' power in the ABF output. The common sample matrix inversion (SMI) Minimum Variance Distortionless Response (MVDR) ABF computes the beamformer weights by substituting the sample covariance matrix (SCM) for the unknown ensemble covariance matrix in the MVDR expression. Errors in the SCM estimate of interferer direction due to limited sample support or interferer motion degrade the ABFs ability to suppress the interferer. This presentation exploits array polynomial properties to design a robust ABF. The array polynomial for a uniform linear array beamformer is the z-transform of the array weights. The array polynomial zeros on the unit circle correspond to the beampattern nulls. The proposed double zero (DZ) MVDR ABF solves for the MVDR weights using the SCM for a half-aperture subarray, then convolves the half-aperture weights with themselves to obtain the full aperture ABF weights. The resulting array polynomial for the DZ MVDR ABF has second-order zeros, producing broader and deeper notches in the interferer direction. The DZ MVDR ABF outperforms the SMI MVDR and covariance matrix tapered ABFs in simulations with stationary and moving interferers. [Work supported by ONR 321US.]

2:15

4pSP4. Algorithms for measuring periodicity in F0 tracking. Xiao Chen, Hao Zhang, and Stephen A. Zahorian (Dept. of Elec. and Comput. Eng., State Univ. of New York at Binghamton, 4400 Vestal Parkway East, Binghamton, NY 13902, xchen49@binghamton.edu)

Measuring periodicity is an important measurement in speech processing. It can be used in many areas, especially for tracking fundamental frequency (F0), typically referred to as pitch. This seemingly easy measurement is made difficult since even voiced sections of speech are only semi-periodic or periodic over short intervals. In this paper, four functions for measuring periodicity are compared for both time domain and frequency domain processing. They are autocorrelation, normalized cross correlation function, spectral harmonics correlation, and normalized spectral harmonics correlation. In some cases, these four functions behave very similarly to each other; however, there are advantages and disadvantages, depending on conditions. The functions were experimentally evaluated in an F0 tracking experiment based on the Keele database, which has "ground truth" for pitch.

2:30

4pSP5. Automated detection of honeybee begging signals from long term vibration monitoring of honeybee hives. Michael-Thomas Ramsey, Martin Bencsik, and Michael Newton (Sci. and Technol., Nottingham Trent Univ., Phys. and Mathematical Sciences' College of Arts & Sci., Nottingham Trent Univ., Clifton Ln., Nottingham NG11 8NS, United Kingdom, n0530709@ntu.ac.uk)

We are using high performance accelerometers embedded in the honeycomb to create long term vibrational data sets comprising a range of individual bee pulsed vibrational messages. The statistics of the "vibrational language" of the honeybee can thus be explored. The aim of this study is to create and optimize software that detects and analyses the occurrences of honeybee "begging signals." Recording from a beehive were monitored for 117 days (July to November 2014) from within its central frame using two ultra-high performance accelerometers (Brüel and Kjær, 1000 mV/g), one in the center and the other one 7 cm lower down. Home-build Linux 'bash code' was written to continuously record the accelerometer output into separate one-hour long files, without any data loss. Matlab® code has been developed, that compares a template and instantaneous spectrograms to detect a begging signal. Six independent parameters of the template signal waveform were further optimized, in order to yield the maximum correct detections. Finally, the detection threshold was tuned ensuring that only begging signals were detected. The results show long term trends in the begging signal statistics, a breakthrough in exploring seasonal variations in the honeybee vibrational language.

2:45

4pSP6. Time varying broadband acoustic response compensation at very low signal to noise ratio. Praswish Maharjan and Paul J. Gendron (Dept. of Elec. and Comput. Eng., Univ. of Massachusetts Dartmouth, 285 Old Westport Rd., Dartmouth, MA 02747, pmaharjan@umassd.edu)

Shallow water ocean acoustic environments are challenging due to their multi-path arrival structure and the various Doppler processes that are associated with platform motion and surface interactions. For moving platforms, each acoustic arrival exhibits a Doppler offset that can be quite significant, which results in an overall response that is doubly spread, one which disperses signals in both time and frequency. A virtual ocean acoustic laboratory based on the Bellhop ray tracing model is used to study the distortion of broadband waveforms through dynamic shallow water ocean acoustic environments. Acoustic response functions are presented in order to illuminate the challenges posed by shallow water environments and mobile platforms. The shared Doppler process of the arrivals is illustrated in order to illuminate the value of bulk Doppler compensation schemes. This virtual acoustic laboratory is used to test spread spectrum communication performance at very low SNR. Bulk Doppler compensation schemes derived from joint estimation of symbols and the acoustic response are compared to the actual path dilation processes. Simulation results are presented for fixed source with moving receiver shallow water environments at diverse center frequencies and bandwidths.

3:00

4pSP7. Wind farm infrasound—Are we measuring what is actually there or something else? Steven E. Cooper (The Acoust. Group, 22 fred St., Lilyfield, NSW 2040, Namibia, drnoise@acoustics.com.au)

In the olden days of acoustics (pre digital), low frequency analysis used analog narrow band filters and cathode ray oscilloscopes for special problems leading to the general use of peak values. Analog filters have time constants that can affect the derived rms values requiring caution where high crest factors are involved. Modern narrowband digital analysis is based on a FFT of the time signal to extract the periodic function that occurs in the time domain that are then displayed as discrete peaks in the frequency domain. FFT analysis of turbines show discrete infrasound peaks at multiples of the blade pass frequency in addition to sidebands in the low frequency range spaced at multiples of the blade pass frequency. Are these signals actually there or are they a product of modern day analysis. Is the infrasound signature a clue to a different area of investigation? The paper will show the results of testing to compare old fashioned and modern day analysis.

Session 4pUW

Underwater Acoustics and Signal Processing in Acoustics: Environmental Variability Impact on Shallow Water Acoustics II

Kevin D. Heaney, Cochair
OASIS Inc., 11006 Clara Barton Dr., Fairfax Station, VA 22039

Sergio Jesus, Cochair
University of Algarve, Campus de Gambelas, Faro 8005-139, Portugal

Contributed Papers

1:00

4pUW1. Remote sensing for bottom inversion. Sergio M. Jesus (Univ. of Algarve, Campus de Gambelas, Faro 8005-139, Portugal, sjesus@ualg.pt)

The scenarios of interest for estimating bottom and sub-bottom physical properties now encompass both deep and shallow, or very shallow, coastal waters, for the deployment of renewable energy platforms (e.g., wind farms and wave/tidal energy plants). This new paradigm, together with the continuous requirement for reducing survey time (and cost), spinned out the concept of a distributed and reconfigurable seismic survey system composed of a fleet of autonomous underwater vehicles (AUV) carrying acoustic sensing arrays. Such system poses a number of technical as well as scientific challenges, among which that of sensor positioning for optimal bottom inversion performance in a given scenario. The present work addresses this issue through the eye of the sensor configuration that maximizes diversity and proposes sampling incoherence bounds for 1, 2, and 3D array systems. Random sampling is a concept that favors diversity and allows for the usage of low-complexity inversion schemes such as those based on compressed sensing. Simulations on realistic environments illustrate the proposed concept. [This work is part of project WiMUST—Widely Scalable Mobile Underwater Sonar Technology funded under program H2020 of the European Union.]

1:15

4pUW2. Source depth discrimination: An evaluation and comparison of several classifiers. Ewen Conan, Julien Bonnel (Lab-STICC, ENSTA Bretagne, 2 rue François Vermy, Brest 29200, France, ewen.conan@ensta-bretagne.org), Thierry Chonavel (Lab-STICC, Telecom Bretagne, Brest, France), and Barbara Nicolas (Créatis, Université de Lyon, Lyon, France)

Source depth estimation with a vertical linear array generally involves mode filtering, followed by matched-mode processing. However, this method has two main limitations: the problem of mode filtering is ill-posed in the case of partially spanning arrays; matched-mode processing is sensitive to environmental mismatch. Therefore, concerns for robustness motivate a simpler approach. The problem of depth estimation is reduced to a binary classification problem: source depth discrimination. Its aim is to evaluate whether the source is near the surface or submerged. These two hypotheses are formulated in terms of normal modes, using the concept of trapped and free modes. Several classification rules, based on modal filtering or on subspace projections, are studied. Monte-Carlo methods are used to evaluate their performance and compute receiver operating characteristics. This allows the choice of a discrimination threshold according to some expected performance. The benefits of considering a source depth discrimination problem rather than a source localization one are highlighted. The influence of noise and environmental mismatch are investigated, as well as the choice of the discrimination depth and the choice of the limit between trapped and free modes.

1:30

4pUW3. Interpretations of the frequency difference autoproduct in multipath environments. Brian M. Worthmann (Appl. Phys., Univ. of Michigan, 2010 Lay Automotive Lab., 1231 Beal Ave., Ann Arbor, MI 48109, bworthma@umich.edu) and David R. Dowling (Mech. Eng., Univ. of Michigan, Ann Arbor, MI)

When locating remote acoustic sources in a shallow ocean sound channel, the established array signal processing technique known as matched field processing (MFP) has shown much success. However, MFP is sensitive to mismatch between the modeled and actual environments, and may fail to localize acoustic sources in the presence of such mismatch, particularly at high frequencies. A recent nonlinear array signal processing technique, frequency difference MFP (*Abadi et. al.* 2012, *Worthmann, et. al.,* under review), has shown some success in localizing high frequency sources by moving the replica calculations to a lower, out-of-band, difference frequency where the detrimental effects of environmental mismatch are less severe. To extract the requisite out-of-band difference frequency information from the measured signals, a quadratic product, termed the autoproduct, is formed from complex signal amplitudes separated by the difference frequency but still lying within the signal bandwidth. Through the use of simple multipath propagation environments, the nature of this autoproduct is explored, and the reasons that it provides out-of-band field information are presented. More complex propagation environments are simulated as well to demonstrate some of the expected and unexpected behaviors of the autoproduct. [Sponsored by the Office of Naval Research and the National Science Foundation.]

1:45

4pUW4. The relation between the waveguide invariant and array invariant. Hee-Chun Song and Chomgun Cho (Scripps Inst. of Oceanogr., 9500 Gilman Dr., La Jolla, CA 92093-0238, hcsong@mpl.ucsd.edu)

The waveguide invariant β is based on the dependence of group speed on phase speed and summarizes the robust interference phenomenon in the range-frequency plane. Over the last decade the elegant approach has been utilized for various applications including passive source ranging. Separately, the array invariant approach [Lee and Makris, J. Acoust. Soc. Am., vol. 119, pp. 336–351 (2006)] has been proposed for a robust source-range estimator from beam-time intensity data using either a horizontal or vertical array. In this letter, it is shown that the array invariant can be derived from the waveguide invariant theory assuming $\beta = 1$.

2:00

4pUW5. Robust source-range estimation using a vertical array and the array invariant. Chomgun CHO, Hee-Chun Song, and William S. Hodgkiss (Scripps Inst. of Oceanogr., UCSD, 9500 Gilman Dr., La Jolla, San Diego, CA 92093-0238, chomgun@gmail.com)

The array invariant based on beam-time migration has been proposed for instantaneous source-range estimation in acoustic waveguides using a horizontal or vertical array. With minimal knowledge of the environment, the approach has been demonstrated successfully with experimental data in shallow water using a horizontal array. Moreover, it was determined recently that the array invariant can be derived from waveguide invariant theory based on the dependence of group speed on phase speed. In this paper, the array/waveguide invariant approach to source-range estimation is applied to a vertical array in a fluctuating ocean environment over a one-day period. Specifically, the range estimates using a 12-m long vertical array in ~100 m deep water are within 9–16% relative error for a 2–3 kHz source at 6-km range, demonstrating the robustness of this approach.

2:15

4pUW6. Single and multiple snapshot compressive matched field processing. Kay L. Gemba, William S. Hodgkiss, and Peter Gerstoft (Marine Physical Lab., Scripps Inst. of Oceanogr., Univ. of California, San Diego, 8820 Shellback Way, Spiess Hall, Rm. 446, La Jolla, CA 92037, gemba@ucsd.edu)

Matched field processing is a generalized beamforming method which matches received array data to a dictionary of replicas to locate and track a source. The solution set generally is sparse since there are considerably fewer sources than replicas. This underdetermined problem can be solved with sparse processing (SP) which potentially is attractive for several reasons. The traditional spatial matched-filter problem is reformulated as a convex optimization problem subject to a sparsity constraint. For example, an elastic net seems to be an appropriate penalty in order to find the best match among a correlated group of replicas. Another advantage is that SP does not require inversion of the sample covariance matrix and therefore can outperform conventional high-resolution processors in snapshot deficient scenarios (i.e., fast moving sources). A third potential advantage is that SP can achieve super-resolution at high SNR and discriminate between closely spaced sources. Here, we demonstrate the performance of single and multi-snapshot SP to track a towed source using the SWellEx-96 data set. Results are benchmarked using the Bartlett and the white noise constraint processors. We further discuss the processing of multiple frequencies in order to improve the source tracking.

2:30

4pUW7. The exponential decay of underwater acoustic intensity with increasing altitude of low-frequency sound from a Robinson R44 helicopter. Dieter A. Bevans and Michael J. Buckingham (Marine Physical Lab., Scripps Inst. of Oceanogr., Univ. of California, San Diego, 9500 Gilman Dr., La Jolla, CA 92093-0238, dbevans@ucsd.edu)

A series of underwater acoustic experiments utilizing a Robinson R44 helicopter and an underwater receiver station has been conducted in shallow (16 m) and deep (>100 m) water. The receiver station consisted of an 11-element nested hydrophone array with a 12 m aperture. In the shallow water experiments the array was configured as a horizontal line (HLA) 0.5 m above the seabed; whereas in deep water the array was suspended from the surface in a vertical line (VLA). An in-air microphone was located immediately above the surface. In this paper the power spectral density as a function of helicopter altitude will be reported from measurements on a single hydrophone. The main rotor blades of the helicopter produce low-frequency harmonics, the lowest frequency being ~13 Hz. The tail rotor produces a sequence of harmonics approximately six times higher in frequency. Between heights of 30 m to 600 m above the sea surface the underwater intensity was found to decay exponentially with increasing helicopter altitude. Interpretation of the observed low-frequency sound signatures is being facilitated with a numerical three-layer (atmosphere-ocean-sediment) acoustic propagation model in which the source may be moving or stationary. [Research supported by ONR, NAVAIR, and SIO.]

2:45

4pUW8. The effect of signal bandwidth on partially saturated broadband ocean acoustic transmission scintillation in a shallow water waveguide. Delin Wang, Wei Huang, and Purnima Ratilal (Elec. and Comput. Eng., Northeastern Univ., 302 Stearns Ctr., Rm. 312, 360 Huntington Ave., Boston, MA 02115, wang.del@husky.neu.edu)

The acoustic field in a shallow water waveguide at small source-receiver separations <3 km are partially saturated. Here, we examine the scintillation statistics of partially saturated broadband ocean acoustic transmissions in the low to mid frequency range collected using a high-resolution towed horizontal coherent hydrophone array system. The standard deviation of intensity fluctuations and scintillation index are quantified as a function of signal bandwidth and source-receiver separation. The amplitude and intensity distributions are compared with corresponding distribution models for the partially saturated field to determine the number of independent statistical fluctuations present in the broadband data.

3:00–3:15 Break

3:15

4pUW9. Three-dimensional inversion technique in ocean and seabed acoustics using the parabolic equation method. Camilo C. Roa (Ocean and Mech. Eng., Florida Atlantic Univ., 101 North Beach Rd., Dania Beach, FL 33004, croa1@fau.edu) and George V. Frisk (Ocean and Mech. Eng., Florida Atlantic Univ., Boca Raton, FL)

A three-dimensional parabolic equation (PE) and perturbation approach is used to invert for the range-dependent geoacoustic characteristics of the seabed. The model assumes that the sound speed profile is the superposition of a known range-independent profile and an unknown depth and range-dependent perturbation. Using a Green's function approach, the total measured pressure field in the water column is decomposed into a background field, which is due to the range-independent profile, and a scattered field, which is due to the range-dependent perturbation. When the Born approximation is applied to the resulting integral equation, it can be solved for the range-dependent profile using linear inverse theory. For simplicity, the sound speed profile in the water column is assumed to be known, and the range-dependent perturbation is added to the index of refraction $n(x,y,z)$, rather than the sound speed profile $c(x,y,z)$. The method is implemented in both Cartesian (x,y,z) and cylindrical (r,θ,z) coordinates with the forward propagation of the field in x and r, respectively. Synthetic and experimental data are used to demonstrate the validity of the method. Keywords: Three-dimensional parabolic equation method, geoacoustic inversion, range-dependent sound speed profile, linear inversion, Born approximation, and Green's functions.

3:30

4pUW10. Broadband impulse response modeling using the single frequency parabolic equation solution. Kevin D. Heaney (OASIS Inc., 11006 Clara Barton Dr., Fairfax Station, VA 22039, oceansound04@yahoo.com)

The parabolic equation has been considered the workhorse of low to mid frequency propagation modeling in range dependent environments due to its accurate handling of both refraction and diffraction. The frequency domain solution approach of the PE has severely limited its use in broadband pulse synthesis die to the computational cost of computing each Fourier component. In this paper a hybrid approach that blends the PE forward model for field computation and an analytic model for travel time computation is presented. The PE is computed at the center frequency in a range dependent environment. The beam arrival angle for a virtual line array at the receiver depth (as well as sea surface and sea floor) is stored at each range. The average group slowness as a function of phase speed at the source (wave number invariant) is computed using analytic acoustics methods (Snell's law). The impulse response at each range is computed by incoherently adding the pier of each beam at the arrival time of the associated phase speed energy (the range averaged group slowness). This approach reproduces the behavior of the pulse spread, without the cost of the broadband PE.

3:45

4pUW11. Three-dimensional iterative parabolic approximations. Pavel S. Petrov (School of Natural Sci., Far Eastern Federal Univ., 8 Suhanova St., Vladivostok 690950, Russian Federation, petrov@poi.dvo.ru)

A hierarchy of the 3D coupled parabolic equations is derived by the method of multiple scales. The solutions of the derived equations represent the successive terms in an asymptotic expansion of the solution of the 3D Helmholtz equation. The equations are complemented with the consistent interface and boundary conditions. The Cauchy initial conditions for the parabolic equations are set up in such a way that the solution in the far field approximates the solution of the Helmholtz equation in the unbounded 3D space. The derived parabolic equations are used to solve a problem of the propagation in a perfect 3D wedge. The comparison to the image source solutions is used for the validation of the proposed parabolic approximation.

4:00

4pUW12. Stochastic-basis matched-field processing. Steven I. Finette, Peter C. Mignerey, and Roger M. Oba (U.S. Naval Res. Lab., Peter Mignerey code 7160, Washington, DC 20375-5350, peter.mignerey@nrl.navy.mil)

Matched-field processing (MFP) suffers serious degradation due to environmental mismatch between received acoustic-field vectors and modeled replica vectors. Physical reasons for degradation include uncertainty caused by incomplete descriptions of the parameters and fields necessary for correct specification of the acoustic waveguide (i.e., environmental uncertainty), and system uncertainty associated with incomplete knowledge of the array configuration, source depth, etc. Recent research in the theory of stochastic-basis expansions (polynomial chaos) provides a mathematically consistent way of incorporating both types of uncertainty into MFP. Such expansions are used efficiently to construct replica matrices that steer high-rank subspaces capable of capturing signals with uncertain wavefronts. When combined with cross-spectral density matrices, stochastic-basis steering matrices enable the design of new processors with properties not previously evaluated in a MFP context. A maximum likelihood processor is developed which incorporates environmental uncertainty through polynomial chaos expansions. The processor can be written as a Frobenius product between an estimated cross-spectral density matrix and the inverse of a stochastic-basis replica matrix. This talk will outline the theoretical foundation of stochastic-basis MFP, and illustrate the method with some simulations. [Work supported by the Office of Naval Research.]

4:15

4pUW13. Acoustic propagation in the Strait of Georgia. Nicos Pelavas, Sean Pecknold (DRDC Atlantic Res. Ctr., 9 Grove St., Dartmouth, NS B2Y 3Z7, Canada, nicos.pelavas@drdc-rddc.gc.ca), Matthew Coffin, Kevin Dunphy (GeoSpectrum Technologies, Inc., Dartmouth, NS, Canada), and Dugald Thomson (Dept. of National Defence, Trinity - Acoust. Data Anal. Ctr., , Halifax, NS, Canada)

In recent years, there has been a proliferation of Ocean Observing Systems (OOS) along with a wide distribution of their associated data products. The collected data support scientific research, industry, and government organizations by providing long term measurements of biological, chemical, and physical properties of the ocean environment. However, the collection and distribution of underwater acoustic data poses a potential security risk for naval vessels operating in the vicinity of OOS. The Canadian Forces Maritime Experimental and Test Ranges (CFMETR) provide an underwater tracking facility for naval tests, and are approximately 50 km from hydrophones of the Victoria Experimental Network Under the Sea (VENUS) observatory. Under an existing CFMETR-VENUS agreement, data are diverted during certain naval tests. In order to minimize the frequency of these data diversions, a study is being conducted to investigate acoustic propagation in the Strait of Georgia. The results of acoustic modeling and measurement of transmission loss from CFMETR to VENUS will be presented. A software application called CAVEAT is also presented. The application was developed to integrate the transmission loss results along with other sonar parameters to enable operators at CFMETR to determine the risk of acoustic exposure.

4:30

4pUW14. A comparison of the reflection coefficient predictions of two competing sediment acoustic models. Anthony L. Bonomo, Nicholas Chotiros, and Marcia Isakson (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78713, anthony.bonomo@gmail.com)

Currently, there are several competing models that have been used to describe the acoustic properties of sandy sediments. These models include those that assume the sediment to behave as an acoustic fluid, a viscoelastic solid, and as a porous media following Biot theory. Perhaps the two most sophisticated acoustic models that have been applied to sand are the Viscous Grain Shearing (VGS) model of Buckingham and the Extended Biot (EB) model of Chotiros. While both of these models have been found to agree with measured sound speed dispersion and attenuation data, previous work has shown that the reflection coefficients predicted using these models disagree. In this work, the reflection coefficient predictions of the VGS and EB models will be compared for both the case of a homogeneous sand half-space and the case of a sand layer overlying a rock substrate. [Work supported by ONR, Ocean Acoustics.]

4:45

4pUW15. Hydroacoustic survey of geotechnical activities for the Virginia offshore wind technical advancement program. Erik J. Kalapinski (Energy Programs, Tetra Tech, Inc., 160 Federal St., Fl. 3, Boston, MA 02210, erik.kalapinski@tetratech.com) and Kristjan A. Varnik (Eng., Tetra Tech, Inc., Boston, MA)

As offshore wind energy development increases across the eastern seaboard, there is a growing need to determine the short and long-term effects of activities associated with this development on marine ecosystems. One area of particular importance is the potential effects of underwater noise on marine life. To better characterize underwater sound levels associated with geotechnical activities, a hydroacoustic measurement program of geotechnical survey activities was completed in support of Dominion's Virginia Offshore Wind Technical Advancement Program. An important component of this project was the advancement of new technologies and the use of best available science to collect data for more accurate impact determinations. The overall goal of the hydroacoustic measurement program was to field-verify the projected acoustic impacts during geotechnical activities. This new insight will support both decision making in the execution of offshore wind site characterization surveys, and reduce the potential for future impacts. Measurements were made using a combination of equipment including a cabled real-time hydroacoustic analysis systems and fixed autonomous static recorders. Upon conclusion of the hydroacoustic survey, data were downloaded and directly correlated with daily activity logs from the vessels used in performing the offshore geotechnical work thereby providing the means to coordinate acoustic events.

Poster Papers

Posters will be on display from 1:00 p.m. to 5:00 p.m. Authors will be at their posters from 3:00 p.m. to 3:15 p.m.

4pUW16. Computationally efficient algorithm for Kirchhoff approximation. Donghyeon Kim (Ocean Eng., Korea Maritime and Ocean Univ., 727 Taejong-ro, Yeongdo-Gu, 253, Ocean Sci. and Technol., Busan ASIKRKS012BUSAN, South Korea, donghyeonkim@kmou.ac.kr), Hubum Jin (Mathematics, POSTECH, Pohang, South Korea), Yoon Hee Ji (LIG Nex1, Seongnam, South Korea), Ho Seuk Bae, Woo-Shik Kim (Agency for Defense Development, Changwon-si, South Korea), and Jea Soo Kim (Ocean Eng., Korea Maritime and Ocean Univ., Busan, South Korea)

In order to simulate the target echoes scattered from submerged underwater objects, Kirchhoff approximation is widely used for high frequency region. Since Kirchhoff integration is based on integrating the contributions from discretized boundary elements, the computation can be time-consuming especially for broadband pulses. In this study, a numerically efficient method for generating the scattered signal in time domain based on convolution is proposed and tested. It is shown that the computational time can be reduced by an order of 10–100 in typical cases.

4pUW17. Temporal variation of transmission loss by internal tide in the southern sea of Jeju island in summer. Juho Kim, Hansoo Kim, Dong-Guk Paeng (Jeju National Univ., Ara 1 dong, Jeju 064-756, South Korea, lizard@jejunu.ac.kr), and Ig-Chan Pang (Jeju National Univ., Jeju, Jeju Special Self-Governing Province, South Korea)

Temporal variations of acoustic transmission loss (TL) affected by internal tide are simulated using oceanic data measured at two sites in the southern sea of Jeju island. Temperature and salinity were measured every hour for 25 hours during July 27th and 28th 2009. The periodic fluctuation of temperature due to the internal tide was observed and its vertical displacement was more than 10m. In order to investigate the variation of TL, acoustic propagation between two measurement sites (3.8 km distance) was computed at a source depth of 10 m. Acoustic propagation model RAM was used for the simulation. Standard deviations of TL variation were 4.2 dB and 3.7 dB for center frequencies of 100 Hz and 1 kHz with 1/3 octave band, respectively. The lower frequency was more correlated with the tidal period than the higher one. Detection range was also varied by the internal tidal, up to 1 km for the 60 dB detection level. These results imply that tidal variation of TL should be considered for acoustic researches at the southern sea of Jeju island. [This work was supported by UD130007DD and IITP-2015-H8601-15-1004.]

4pUW18. Modeling of sound in coastal oceans with a finite volume method. Wen Long (PNNL, 1100 Dexter Ave North, Seattle, WA 98109, wen.long@pnnl.gov), Ki Won Jung (PNNL, Richland, WA), Zhaoqing Yang (PNNL, Seattle, WA), Zhiqun Deng (PNNL, Richland, WA), and Andrea Copping (PNNL, Seattle, WA)

In this research, the finite volume method is employed to solve the 3D Helmholtz equation of sound propagation in the coastal environment. The grid system consists of triangular grids in horizontal plane and sigma-layers in vertical dimension. A 3D sparse matrix solver with complex coefficients is formed for solving the resulting acoustic pressure field. The recent CSLP and ADEF1 preconditioning methods are applied to efficiently solve the matrix system iteratively with MPI parallelization using a high performance cluster. This model is then coupled with the Finite Volume Community Ocean Model (FVCOM) for simulating sound generated by offshore wind energy platform constructions in a range-dependent setting. Details of the development and initial validation will be presented. Keywords: Sound, Coastal Ocean, Finite Volume, Offshore Wind, Coupling, CSLP, ADEF1, Helmholtz equation.

OPEN MEETINGS OF TECHNICAL COMMITTEES

The Technical Committees of the Acoustical Society of America will hold open meetings on Tuesday, Wednesday, and Thursday. See the list below for the exact schedule.

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

Committees meeting on Tuesday, 3 November

Committee	Start Time	Room
Engineering Acoustics	4:30 p.m.	Orlando
Acoustical Oceanography	7:30 p.m.	River Terrace 2
Animal Bioacoustics	7:30 p.m.	City Terrace 9
Architectural Acoustics	7:30 p.m.	Grand Ballroom 3
Physical Acoustics	7:30 p.m.	St. Johns
Psychological and Physiological Acoustics	7:30 p.m.	Grand Ballroom 7
Structural Acoustics and Vibration	7:30 p.m.	Daytona

Committees meeting on Wednesday, 4 November

Committee	Start Time	Room
Biomedical Acoustics	7:30 p.m.	Clearwater
Signal Processing in Acoustics	8:00 p.m.	City Terrace 7

Committees meeting on Thursday, 5 November

Committee	Start Time	Room
Musical Acoustics	7:30 p.m.	Grand Ballroom 1
Noise	7:30 p.m.	Grand Ballroom 2
Underwater Acoustics	7:30 p.m.	River Terrace 2