Session 3aAA

Architectural Acoustics and Noise: Advancements and Best Practices in Instrumentation for Architectural Acoustics and Noise

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

Eric L. Reuter, Cochair
Reuter Associates LLC, 10 Vaughan Mall, Portsmouth, NH 03801

Chair's Introduction—8:00

Invited Papers

8:05

3aAA1. Special tools and procedures for measuring ground vibration. James E. Phillips (Wilson, Ihrig & Associates, Inc., 6001 Shellmound St., Suite 400, Emeryville, CA 94608, jphillips@wiai.com)

This talk will present some of the tools and procedures used to analyze ground vibration and the propagation of vibration through soils. These tools and procedures are typically used to evaluate and control groundborne noise and vibration associated ground based transportation such as trains.

8:25

3aAA2. Using real sources to measure the acoustical behavior in rooms. Bruce C. Olson (Ahnert Feistel Media Group, 8717 Humboldt Avenue North, Brooklyn Park, MN 55444, bcolson@afmg.eu)

It is common practice to use dodecahedron loudspeakers and balloons as the source for acoustical measurements. The drawback to this of course, is that it means an interruption to the normal performance in order to collect data in the presence of an audience. New techniques are now available for measurements using real acoustic sources without degrading the performance.

8:45

3aAA3. Advancements in instrumentation for source identification. Matthew V. Golden (Scantek, 6430c Dobbin Rd, Columbia, MD 21045, goldenm@scantekinc.com)

In the beginning, acousticians only had their ears to identify sources. Today we have many more advanced instrumentation, even more than the simple intensity probe that we had just a few decades ago. This paper will review several of these new devices. The first will be an Intensity Tracking System that uses machine vision to produce sound intensity maps. The second is an Acoustic Camera that uses 256 microphones in a two dimension array to create real time videos of the sound field overlaid with standard video. The final instrument will be an environmental noise monitor that uses the arrival times at multiple microphones to detect the location of sources in three dimensional space. Real life measurement results will be shown from each instrument. These new tools give acousticians far more tools than they had just a few years ago.

9:05

3aAA4. Applications of mobile computing devices in acoustics. Benjamin Faber (Faber Acoustical, LLC, 931 Valley View Dr, Santaquin, UT 84655, ben@faberacoustical.com)

In the emerging post-PC era, more and more day-to-day computing tasks will be accomplished with mobile devices, such as the iPhone and iPad. Efforts to bring acoustical measurement and analysis tools to mobile devices have already begun. Mobile devices are not only smaller and lighter even than notebook computers, but they typically employ capacitive touchscreen technology, which enables an unprecedented level of interactivity between user and device. The media-centric nature of the current crop of mobile devices also makes them well-suited for acoustics-related applications. Several examples of hardware and software solutions for acoustical measurements with mobile devices will be presented and discussed.

9:25

3aAA5. Airflow resistance—A comparison between international and american test methods. Marek Kovacik (Scantek Inc., 6430c Dobbin Rd, Columbia, MD 21045, kovacikm@scantekinc.com)

This paper presents the results of a study to determine differences between two test methods that purport to measure airflow resistance. Airflow resistance can be used to determine the acoustical absorption characteristics of materials. In ASTM C522-03(R2009) method, the air is supplied at a steady rate and the pressure difference across the test specimen is measured. In the ISO 9053/EN 29053 B method, a low frequency acoustic wave is produced by a moving piston. The pressure change measured inside the testing apparatus is directly related to airflow resistivity. In a recent round robin, sponsored by ASTM, samples of various densities were tested using the ASTM method. Airflow resistance was also measured with a Norsonic 1517 Airflow Resistance Measurement System following the ISO method. Differences between methods, effects of sample orientation, and other causes of uncertainty will be presented.

9:45

3aAA6. Variations in standing-wave impedance tube design and the effect on the resulting data. Bonnie Schnitta and Greg Enenstein (SoundSense, LLC, 46 Newtown Lane, Suite One, East Hampton, NY 11937, bonnie@soundsense.com)

Standing-wave impedance tubes are a practical and common method for estimating the acoustic absorption characteristics of a material. However, the standing-wave impedance tube test contains inherent variability with its design. Impedance tube standards allow for flexibility in tube material and tube dimensions. These variables in the design criteria of the impedance tube can produce disparities between measured absorption coefficients across different impedance tubes as well as when compared to well-established reverberation room data. Consequently, when designing a tube, in order to obtain accurate absorption values, it becomes necessary to optimize a tube, beyond merely preventing cross modes and allowing for a long enough tube to develop a pressure high and low. This study examines the effects of surface interactions in impedance tubes and how varying tube dimensions, most notably the width, will affect wave development.

WEDNESDAY MORNING, 24 OCTOBER 2012 — JULIA LEE A/B, 8:55 A.M. TO 11:15 A.M.

Session 3aAB

Animal Bioacoustics: Vocalization, Hearing, and Response in Non-Human Vertebrates I

Michael A. Stocker, Chair
Ocean Conservation Research, P.O. Box 559, Lagunitas, CA 94938

Chair's Introduction—8:55

Contributed Papers

9:00

3aAB1. Temporal coherence in Budgerigars (*Melopsittacus undulatus*). Erikson G. Neilans and Micheal L. Dent (Psychology, University at Buffalo, SUNY, Buffalo, NY 14260, mdent@buffalo.edu)

Auditory scene analysis has been suggested as a universal process that exists across all animals. Relative to humans, however, little work has been devoted to how animals isolate sound sources to create auditory objects. Frequency separation of sounds is arguably the most common parameter studied in auditory streaming, yet it is not the only factor. Elhilali et al. (2009) found that in humans, synchronous tones are heard as a single auditory stream, even at large frequency separations, compared to asynchronous tones with the same frequency separations, which are perceived as two sounds. These findings demonstrate how both timing and frequency separation of sounds are important for auditory scene analysis. It was unclear how animals, such as budgerigars (Melopsittacus undulatus), would perceive synchronous and asynchronous sounds. Budgerigars were tested on the perception of synchronous, asynchronous, and partially overlapping pure tones and budgerigar contact calls. Budgerigars segregate partially overlapping sounds in a manner predicted by computational models of streaming. However, overlapping budgerigar contact calls are more likely to be segregated than pure tone stimuli with the same temporal overlap. These results emphasize the necessity of using complex communication signals when examining complex sound perception processes such as auditory scene analysis.

9:15

3aAB2. Spatial release from electronic clutter masking in FM bat echolocation. Michaela Warnecke, Mary E. Bates, and James A. Simmons (Neuroscience, Brown University, 185 Meeting St, Providence, RI 02912, michaela_warnecke@brown.edu)

For big brown bats, angular separation of target and clutter echoes causes spatial release from clutter masking. Experiments using echoes that are electronically-generated by loudspeakers show that lowpass-filtering of normally masking echoes also causes clutter masking to disappear. Such lowpass-filtering induces amplitude-latency trading, which retards neural response times from clutter echoes at higher frequencies relative to lower frequencies. Countervailing changes in presentation-times of higher frequencies in electronically-generated clutter echoes restores masking. We present new results showing that moving the clutter-delivering loudspeakers to a different azimuth and elevation causes clutter masking to disappear. But, similar to the earlier experiments, the countervailing changes in presentation of higher frequencies reintroduce masking. In the bat's inferior colliculus, FM sounds that mimic broadcasts and echoes evoke ~1 spike per sound at each neuron's best frequency. However, amplitude-tuning is very broad, so bats work in the latency domain instead, to exploit their high acuity for detecting coherence or non-coherence of echo responses. Overall, the results indicate that big brown bats use neuronal response timing for virtually all auditory computations of echo delay, including those involved in clutter rejection derived from echo spectra. [Work supported by ONR and NSF.]

9:30

3aAB3. Dhole (asiatic wild dog) and tapir vocalizations: Whistling in the jungle. David Browning (Physics Department, URI, 139 Old North Road, Kingston, RI 02881, decibeldb@aol.com) and Peter M. Scheifele (Communication Sciences Dept., University of Cincinnati, Cincinnati, OH)

Few land mammals whistle, and then, as with the marmots, usually just a quick alarm signal to visually alert. Yet in the jungle are found two distinctly different examples, both unique in their animal groups, which employ whistling as a means of communication in an acoustically noisy and visually very limited environment. Dholes, commonly referred to as whistling dogs, belong to a pack which typically breaks into smaller groups to hunt, primarily in the daytime. They whistle to keep in contact while trying to surround game hidden in the bush. In contrast, tapirs are solitary herbivores, grazing primarily at night. Both have found that a frequency sweep whistle is an effective means of communication either in the "screeching noise" daytime environment or the "rasping noise" insect dominated darkness.

9:45

3aAB4. Using bond graphs to model vocal production in túngara frogs.
Nicole M. Kime (Biological Sciences, Edgewood College, 1000 Edgewood College Drive, Madison, WI 53711, nkime@edgewood.edu), Michael J. Ryan (Section of Integrative Biology, University of Texas, Austin, TX), and Preston S. Wilson (Mechanical Engineering, University of Texas, Austin, TX)

Male anurans (frogs and toads) produce species-specific advertisement calls. Conceptual models illustrate how anurans produce calls, but quantitative models of vocal production are rare. Consequently, how frogs produce complex or nonlinear signals, and how differences in morphology result in call differences among species, is in many cases unclear. Bond graphs are representations of dynamic physical systems that allow one to model both the hydraulic elements and mechanical oscillators of air-driven vocal production systems. They can be used to model either single components or interactions among components within vocal systems. This paper uses túngara frogs to show how bond graphs can be used to model vocal production. Túngara frogs produce complex calls that contain a species-specific "whine" and a facultative "chuck". Their calls are frequency- and amplitude-modulated, and exhibit nonlinearities. A bond graph model of the túngara vocal folds produces sustained oscillations at a frequency typical of this frog. The complexity of the túngara frog call is explained by modulating the behavior of integrated components within its vocal production system. The bond graph modeling approach can be used to understand how the biomechanics of vocal production underlies the diversity of animal signals.

10:00–10:30 Break

10:30

3aAB5. Inventory size and complexity in the song of the American Robin.
Kayleigh E. Reyes (Linguistics, UNC Chapel Hill, Chapel Hill, NC 27599, kreyes@live.unc.edu)

This paper discusses American Robin morning song and offers a component system to explain its complex structure and discuss the relationship between inventory size and syllable complexity. Previous research has noted that Robin song is combinatorial, meaning Robins are capable of creating different songs by concatenating elements called syllables. For each robin, all unique syllables produced during a 40 minute recording were grouped into a syllable inventory. Inspired by observations linguists have made about human phoneme inventories, a component system was devised that broke each syllable into its smallest parts. These parts were found by using minima in intensity contours to correspond to natural breaks in the robin's utterances. This component system included all unique syllable parts, 7 total, from 11 syllable inventories with between 8 and 24 different syllables. This system showed certain components occurred more often than others and robins with larger syllable inventories had more rare components on average. It also showed each token of the same syllable type had the same component structure. This component system will also be used to test for patterns in sequencing of components in syllable structure to determine where components can occur and constraints on syllable sequencing in overall song structure.

10:45

3aAB6. Use of social sounds by humpback whales (*Megaptera novaeangliae*) in the Western Antarctic Peninsula feeding grounds.
Michelle Klein (College of the Atlantic, 105 Eden Street, Bar Harbor, ME 04609, mklein@coa.edu) and Douglas Nowacek (Marine Science & Conservation, Duke University Marine Lab, Beaufort, NC)

Humpback whales (*Megaptera novaeangliae*) are renowned for the complex structured songs produced by males. A second, relatively understudied area of humpback acoustic communication concerns un-patterned sounds known as "social sounds," produced by both males and females. This paper explores the use of non-song sounds by humpback whales on the Western Antarctic Peninsula feeding ground. To obtain high quality, close range recordings of humpback whale social sounds, digital acoustic tags (DTAGs) were placed on humpback whales exhibiting foraging behavior during the austral autumn of 2009 and 2010. Overall vocalization rates of two types of social sounds, "wops" and "grunts," showed that there was not a significant diurnal pattern in call production, suggesting that perhaps these calls are not used exclusively for foraging on the feeding ground. These results enhance our understanding of this acoustically advanced species, and will also be useful in conservation and management efforts. The acoustic parameters of the calls that were identified can be used in the development of automatic detection algorithms, and the behavioral contexts during call production can assist in interpretation of passive acoustic monitoring research on humpback whales, and potentially other baleen whale species as well.

11:00

3aAB7. Is the ocean really getting louder?
Michael A. Stocker (Ocean Conservation Research, P.O. Box 559, Lagunitas, CA 94938, mstocker@OCR.org) and John T. Reuterdahl (Ocean Conservation Research, Mill Valley, CA)

In 1975 Donald Ross indicated a long term trend of low frequency anthropogenic noise increase of 0.55dB/year between 1958 and 1975. This trend in ocean ambient noise levels due to expansion in global shipping has yielded an increase in the ambient noise floor of the ocean that is anywhere from 6dB to 12dB higher than what it was in 1958 (depending on location). What became known as the "Ross Prediction" did not incorporate other anthropogenic sources of noise such as navigation and communication signals, noise from offshore fossil fuel exploration and extraction, and the noises from other marine industrial enterprises. There is a concern that the increase in ambient noise is masking biologically significant sounds, although the evidence for this is still scarce and somewhat speculative. Meanwhile perhaps 90 percent of the biomass of complex vertebrates has been removed from the ocean since 1850 due to industrialized whaling and fishing operations. This paper examines whether the ocean ambient noise floor may have been significantly higher in 1800 than in the 1958 baseline year of the "Ross Prediction," and speculates that ambient noise levels may be less of a biological aggravator than the particular characteristics of a noise source.

Session 3aAO

Acoustical Oceanography: Acoustical Oceanography Prize Lecture

Martin Siderius, Chair
ECE Dept., Portland State Univ., Portland, OR 97201

Chair's Introduction—11:00

Invited Paper

11:05

3aAO1. The problem of sound propagation through the fluctuating ocean and the interplay between ocean acoustics and physical oceanography. John A. Colosi (Department of Oceanography, Naval Postgraduate School, 833 Dyer Road, Monterey, CA 93943, jacolosi@nps.edu)

While the ocean provides a particularly amenable environment for acoustic remote sensing, navigation, and communication, the confounding effects of fluctuations have bedeviled ocean acousticians for decades. To attack this problem the communities of ocean acoustics and physical oceanography became new bedfellows, with the development of ocean acoustic tomography and path-integral/moment-transport theories driven by internal-wave models being major accomplishments for describing low and high-frequency fluctuations. With the advent of relatively inexpensive oceanographic mooring equipment, experiments in the last decades have deployed nearly as many oceanographic instruments as acoustic instruments, thus leading to fundamental discoveries concerning internal tides, solitons, random internal waves and spicy thermohaline structure. These measurements are in turn providing strong constraints on acoustic fluctuation theories and Monte Carlo simulations used to interpret acoustic observations. This presentation will give a review of what has been learned about ocean sound-speed structure and how this information can be better integrated into acoustic fluctuations calculations. It will be shown that acoustic fluctuation theory has developed to the point in which reasonable inversions for internal-wave parameters are now possible.

WEDNESDAY MORNING, 24 OCTOBER 2012 TRIANON A/B, 7:55 A.M. TO 12:00 NOON

Session 3aBA

Biomedical Acoustics and Signal Processing in Acoustics: Measurement of Material Properties Using Wave Propagation Methods

Matthew W. Urban, Chair
Department of Physiology and Biomedical Engineering, Mayo Clinic College of Medicine, Rochester, MN 55905

Chair's Introduction—7:55

Invited Papers

8:00

3aBA1. Acoustic waves in characterizing biological materials: From molecules to soft tissue and bone. A review. Armen Sarvazyan (Artann Laboratories, 1459 Lower Ferry Rd., Trenton, NJ 08618, armen@artannlabs.com)

Various types of acoustic waves: bulk compressional, shear, surface, and numerous modes of guided waves, are used for characterizing structure and composition of biological media. Bulk waves can be used to assess the composition of biological fluids, such as blood, plasma, milk, urine, stomach juice, and biopolymers in solutions and both bulk and shear waves can be used when characterizing soft tissues. While the assessment of skin is primarily performed using surface waves; various modes of guided waves are used to characterize bone and blood vessels. Historically, the problem of acoustic characterization of tissue properties became a hot topic in the early 1970s in conjunction with the emergence and great success of ultrasonography. The idea of remotely assessing the properties of tissue in a region of interest, as seen on an ultrasonic image, attracted the attention of numerous researchers, but until recently and despite intensive studies and great expectations, not many examples of acoustic characterization of tissue properties made a significant impact. Only in

the last decade, this idea started to bring first fruits and is being implemented in commercial devices which are using shear elasticity modulus of tissue as a characterization parameter.

8:20

3aBA2. Abdominal magnetic resonance elastography. Meng Yin (Radiology, Mayo Clinic, 200 First Street SW, Rochester, MN 55905, yin.meng@mayo.edu)

Many disease processes, both focal and diffuse, cause marked changes in cell and tissue mechanical properties. Magnetic resonance elastography (MRE) is a magnetic resonance imaging-based technique for quantitatively assessing the mechanical properties of tissues based on the propagation of shear waves. Multiple studies have demonstrated MRE can be successfully implemented to assess abdominal organs with many potential applications, from detecting diffuse disease processes to characterizing tumors. The first clinical application of MRE to be well documented is the detection and characterization of hepatic fibrosis, which systematically increases the stiffness of liver tissue. In this diagnostic role, it offers a safer, less expensive, and potentially more accurate alternative to invasive liver biopsy. Emerging results also suggest that measurements of liver and spleen stiffness may provide an indirect way to assess portal hypertension. Preliminary studies have demonstrated that it is possible to use MRE to evaluate the mechanical properties of other abdominal structures, such as the pancreas and kidneys. Steady technical progress in developing practical protocols for applying MRE in the abdomen and the pelvis provides opportunities to explore many other potential applications of this emerging technology.

8:40

3aBA3. A comparison of mechanical wave measurement techniques to quantify soft tissue viscoelasticity up to 8 kHz: A phantom study of shear, Rayleigh and Lamb waves. Temel K. Yasar (Mechanical & Industrial Engineering, University of Illinois at Chicago, Chicago, IL), Thomas J. Royston, and Richard L. Magin (Bioengineering, University of Illinois at Chicago, 851 South Morgan St., Chicago, IL 60607, troyston@uic.edu)

Over the past few decades different techniques based on measurement of mechanical wave motion have been developed for noninvasive quantitative measurement and mapping of soft biological tissue shear viscoelastic properties. In this talk we compare two different measurement approaches, three wave types, and several models for quantifying material viscoelasticity up to 8 kHz for a soft tissue phantom material known as ecoflex. Surface waves and Lamb waves are measured using scanning laser Doppler vibrometry (SLDV). Lamb waves and shear waves are measured using magnetic resonance elastography (MRE). Different linear models of viscoelasticity, including Voigt, Maxwell, more generalized and fractional order types, are optimized and compared based on the different experiments. Challenges and limitations of the different techniques and model types, and their adaptation to more complex biological tissue and anatomical structures are discussed. [Work supported by NIH Grants EB012142 and EB007537.]

3a WED. AM

9:00

3aBA4. Quantifying viscoelasticity of boundary sensitive tissues using mechanical wave dispersion ultrasound vibrometry. Ivan Nenadic, Matthew Urban, Cristina Pislaru (Mayo Clinic, 200 1st Street SW, Rochester, MN 55905, ivandulan@gmail.com), Miguel Bernal (Institut Langevin, Paris, Paris, France), and James Greenleaf (Mayo Clinic, Rochester, MN)

The cardiovascular diseases such as atherosclerosis, coronary artery disease, hypertension and diastolic dysfunction have been associated with arterial stiffening and decreased ventricular compliance. Noninvasive techniques capable of quantifying elasticity and viscosity of cardiovascular tissues could facilitate early diagnosis and improve treatment. Here, we present a technique that uses focused ultrasound radiation force to excite mechanical waves in the tissue of interest and pulse echo to track the tissue deformation. Analysis of tissue deformation as a function of time using Fourier transforms allows calculation of the phase wave velocity dispersion (change of velocity as a function of frequency) of various modes of deformation. Continuum mechanics equations governing the motion of a viscoelastic plate and a tube are used to model the myocardial wall and arteries, respectively. Dispersion equations are derived for the two geometries and fit to the measured velocity dispersion to estimate tissue elasticity and viscosity. ECG-gated in vivo measurements of porcine myocardial and arterial elasticity and viscosity through a heart cycle are reported. The results show that both elasticity and viscosity increase during systole and decrease during diastole in the myocardium and arteries, consistent with underlying physiology.

9:20

3aBA5. An energy functional approach for inverse characterization of material properties in elastodynamics. Wilkins Aquino (Civil and Environmental Engineeing, Duke University, Hudson Hall, Durham, NC 27708, wa20@duke.edu), Manuel Diaz (Civil and Environmental Engineering, Cornell University, Ithaca, NY), and Matthew Urban (Department of Physiology and Biomedical Engineering, Mayo Clinic College of Medicine, Rochester, MN)

We present an inverse problem methodology based on the Error in Constitutive Equations (ECE) approach for the identification of material properties in the context of frequency-domain elastodynamics. In the ECE approach, we define a cost functional based on an energy norm that connects a set of kinematically admissible displacements and a set of dynamically admissible stresses. The set of kinematically admissible displacements is composed of fields that satisfy essential boundary conditions and possess sufficient regularity (i.e. smoothness). The set of dynamically admissible stresses is composed of fields that satisfy conservation of linear momentum and natural (i.e. traction) boundary conditions. The inverse problem is solved by finding material properties along with admissible displacement and stress fields such that the ECE functional is minimized. Experimental data is introduced in the formulation as a quadratic penalty term added to the ECE functional. The talk will focus on the reconstruction of elastic and viscoelastic properties in heterogeneous materials in the context of frequency-domain dynamics. Our findings indicate that ECE methods provide faster and more accurate results than conventional least-squares minimization. We will show numerical and experimental results that demonstrate the salient features of the method.

9:40

3aBA6. Quantitative ultrasound imaging for assessing and monitoring therapy. Goutam Ghoshal, Jeremy P. Kemmerer, Chandra Karunakaran, and Michale L. Oelze (Electrical and Computer Engineering, University of Illinois at Urbana-Champaign, 405 N. Mathews, Urbana, IL 61801, gghoshal@illinois.edu)

Conventional ultrasound, which is routinely used for diagnostic imaging applications, is mainly qualitative. However, novel quantitative ultrasound (QUS) imaging modes are being adapted to quantify tissue properties for diagnosing disease, classifying tissues and monitoring and assessing therapy. Ultrasound is a propagating wave that interacts with a medium as a function of the spatially-dependent mechanical properties of the medium. By analyzing the backscattered wave, various properties of the propagating media can be quantified. QUS techniques based on parameterizing spectral features and envelope statistics of the backscattered signal were used to monitor and assess therapy from high intensity focused ultrasound (HIFU) treatment. QUS parameters were obtained by fitting theoretical models to backscatter coefficients (BSCs) that are estimated from backscattered radiofrequency signals. Additional parameters were estimated by fitting the homodyned K distribution to the statistics of the envelope of the backscattered signal. These parameters can be related to tissue properties and microstructure, such as the shape, size and organization of microstructure. Experimental results will be presented to demonstrate the applicability of QUS imaging to monitor and assess HIFU treatments on mouse mammary tumors.

10:00–10:15 Break

Contributed Papers

10:15

3aBA7. Focused, radially polarized shear wave beams in tissue-like media. Kyle S. Spratt, Yurii A. Ilinskii, Evgenia A. Zabolotskaya, and Mark F. Hamilton (Applied Research Laboratories, University of Texas, Austin, TX 78712, sprattkyle@gmail.com)

In the past decade there has been a surge in the optics literature regarding the unique characteristics of focused, radially-polarized light beams. Of particular interest is the existence of a longitudinal component to the electric field in the focal region of the beam, of comparable amplitude to the radial component and yet with a smaller beamwidth [cf. Q. Zhan, Adv. Opt. Photon. 1, 1-57 (2009)]. In the linear approximation there exists a direct analogy between these light beams and radially-polarized shear wave beams in incompressible elastic media, and hence we may interpret the results found in the optics literature as applying to low-frequency shear waves propagating through tissue-like media. When considering nonlinear effects, however, the fact that the gradient of the field experiences a larger gain due to focusing than the field itself implies that the shear wave case is more susceptible to nonlinear behavior than its optical analog. Second-harmonic generation in the focal region of a focused, radially-polarized shear wave beam in a soft solid is investigated. [Work supported by the ARL:UT McKinney Fellowship in Acoustics and by NIH DK070618.]

10:30

3aBA8. Shear wave generation using hybrid beamforming methods. Alireza Nabavizadeh, James F. Greenleaf, Mostafa Fatemi, and Matthew W. Urban (Physiology and Biomedical Engineering, Mayo Clinic College of Medicine, 200 First St SW, Rochester, MN 55905, nabavizadehrafsanjani.alireza@mayo.edu)

Elasticity imaging is a medical imaging modality that measures tissue elasticity to aid in diagnosis of certain diseases. Shear wave-based methods have been developed to perform elasticity measurements in soft tissue. Hybrid beamforming applies both the conventional spherical and axicon focusing to produce a beam for generating a shear wave with increased depth-of-field so that measurements can be made with a plane-like shear wave. We present our aperture design and beam optimization performed using Field II simulations. We varied the number of elements devoted to spherical and axicon focusing as well as the opening angle used for axicon focusing. We tested hybrid beamforming in three elastic phantoms and an excised kidney. We evaluated the shear wave speed measurements accuracy in the phantoms as well as the depth-of-field for each hybrid beam. We compared our results with those from using beams generated using spherical and axicon focusing. Our results show that hybrid beamforming is capable of producing a long narrow beam that performs well when among the 128 elements of transducer, 48 elements are allocated to each axicon portion and 32 elements for spherical aperture while the angle of the axicon aperture is set less than 20.

10:45

3aBA9. On the compressional-shear coefficient of nonlinear elasticity in soft isotropic solids. Bojan Guzina, Egor Dontsov (Civil Engineering, University of Minnesota, 500 Pillsbury Drive SE, Minneapolis, MN 55455, guzina@wave.ce.umn.edu), Matthew Urban, Randall Kinnick, and Mostafa Fatemi (Department of Physiology and Biomedical Engineering, Mayo Clinic College of Medicine, Rochester, MN)

Acoustoelasticity is a technique that allows quantification of the elastic nonlinearity coefficients (the so-called third-order moduli) of a material by measuring the variation of the sound speed in different directions (and/or polarizations) versus the applied uniaxial static stress. When dealing with nominally isotropic solids, the variation of the shear wave speed in two orthogonal directions with the applied stress permits the computation of two out of three independent nonlinearity coefficients. To generate the shear wave in an experimental setting, the acoustic radiation force pulse was applied to uniaxially deformed phantoms with strains of up to 30%. The results demonstrate that the compressional-shear coefficient C, which governs the variation of the (linear) shear modulus with hydrostatic pressure was found to vary substantially from one phantom to another, with the highest value observed in an agarose-based phantom. The importance of this nonlinearity parameter resides in the fact that the magnitude of the acoustic radiation force (ARF) in soft solids is proportional to C-1. For consistency, the values of C obtained from the acoustoelasticity experiment are compared to those deduced from the shear wave amplitude via its relationship to the magnitude of the ARF used to generate the shear wave.

11:00

3aBA10. Use of the radon transform for estimation of shear wave speed. Matthew W. Urban and James F. Greenleaf (Department of Physiology and Biomedical Engineering, Mayo Clinic College of Medicine, 200 First Street SW, Rochester, MN 55905, urban.matthew@mayo.edu)

Many methods in the field of elasticity imaging use shear waves to investigate the material properties of various soft tissues. The accurate and robust measurement of the shear wave speed is necessary for reliable clinical measurements. We propose using the Radon transformation on the spatiotemporal shear wave data to estimate the shear wave speed. A similar algorithm called the Radon sum transformation was proposed by Rouze, *et al* (IEEE Trans. Ultrasonics Ferroelectr. Freq. Control. 2010. pp. 2662-70), but this algorithm requires different input parameters that can affect the results. The use of the full Radon transform allows for the differentiation of waves that are traveling in different directions, thus this method can be used as a directional filter. We will also demonstrate the connection between estimating shear wave speeds in the Radon transform domain and estimating shear wave speeds from Fourier transform k-space. Results from shear wave measurements using shear waves induced by ultrasound radiation force will be shown. We will examine the accuracy of measurements made in calibrated elasticity phantoms, and show examples of shear wave speed

estimation in arteries and kidneys. [This research was supported in part by NIH grants EB002640 and DK082408.]

11:15

3aBA11. Radiation-force-based estimation of acoustic attenuation using harmonic motion imaging. Jiangang Chen, Gary Y. Hou, Fabrice Marquet, and Elisa Konofagou (Department of Biomedical Engineering, Columbia University, Columbia University Medical Campus, 622 W 168th, New York City, NY 10032, ek2191@columbia.edu)

Tissue characterization such as attenuation estimation remains challenging but important. Attenuation represents the energy loss during wave propagation through biological tissues, thus affects both therapeutic and diagnostic ultrasound applications. In this study, a novel attenuation estimation approach was developed using radiation-force-based method of Harmonic Motion Imaging (HMI). The HMI set-up comprised of a forcing transducer (fcenter = 4.7 MHz, AM frequency = 25 Hz) in conjunction with a confocal pulse-echo transducer (fcenter = 7.5 MHz), with the former inducing tissue displacement and the latter simultaneously acquiring RF signals. Tissue displacements were estimated from the RF signals using a 1-D cross-correlation method (window size: 1 mm; overlap: 90%). 2D displacement images were obtained through raster-scan (10×10 mm^2). A linear regression model was applied to the displacements at different depths for calculating attenuation. Gel phantoms with known attenuation (n = 5) (CIRS Inc.) and in vitro canine livers (n = 3) were tested. Results demonstrated that attenuations obtained from the phantoms showed good correlation ($R^2 \approx 99\%$) with the independently obtained values (0.28–1.45 dB/cm/MHz) ($13 \pm 8\%$ underestimated), while those from the canine liver were 0.32 ± 0.03 dB/cm/MHz, within the normal range reported in the literature (0.28–1.01 dB/cm/MHz) (Duck, Academic Press 1990). Future studies will entail attenuation measurements in pathological tissues and HIFU monitoring.

11:30

3aBA12. Elastography and tactile imaging for mechanical characterization of superficial muscles. Diego Turo (Department of Bioengineering, George Mason University, Fairfax, VA), Paul Otto (Dept. of Electrical and Computer Eng., George Mason Univ., Fairfax, VA), Vladimir Egorov, Armen Sarvazyan (Artann Laboratories, Trenton, NJ), Lynn H. Gerber (Department of Rehabilitation Science, George Mason University, Fairfax, VA), and Siddhartha Sikdar (Dept. of Electrica and Computer Eng. and Bioengineering, George Mason Univ., 4400 University Drive, MS 1G5, Fairfax, VA 22030, ssikdar@gmu.edu)

Quantification of the mechanical properties of muscle is of significant clinical interest. Local changes in the mechanical properties of muscle are often associated with clinical symptoms. In particular, myofascial trigger points (MTrPs) are a very common, yet poorly understood and overlooked, cause of nonarticular musculoskeletal pain. MTrPs are localized, stiff, hyperirritable tender nodules, palpated in taut bands of skeletal muscle. Objective validated measures of the mechanical properties of MTrPs could potentially be a clinical outcome measure. We are investigating ultrasound shear wave elastography and tactile imaging as complementary objective methods to assess the mechanical properties of MTrPs. In an ongoing clinical study, we recruited 50 subjects (27 healthy controls and 23 with symptomatic chronic neck pain and active MTrPs). The upper trapezius muscles in these subjects were imaged using shear wave elastography using an external vibration source with varying frequency in the range [50–200] Hz to measure shear wave speed and dispersion in tissue, and tactile imaging using an array of pressure sensors allowing 3D reconstruction of mechanical structure of tissue. Preliminary analysis demonstrates that symptomatic muscle tissue in subjects with neck pain is mechanically more heterogeneous and stiffer compared to normal muscle in control subjects ($p < 0.05$).

11:45

3aBA13. Rayleigh wave propagation method for the characterization of viscoelastic properties of biomaterials. Siavash Kazemirad and Luc Mongeau (Mechanical Engineering Department, McGill University, 817 Sherbrooke Street West, Montreal, QC H3A 0C3, Canada, siavash.kazemirad@mail.mcgill.ca)

The frequency-dependent viscoelastic properties of injectable biomaterials used for vocal fold augmentation and repair must be characterized to ensure the integrity with the vibrating tissue throughout the frequency range of vocalization. Experimental methods for quantifying the frequency-dependent viscoelastic properties of biomaterials over a broad frequency range (i.e., up to 4 kHz) using Rayleigh wave propagations were investigated. Appropriate models for Rayleigh wave propagations in single and layered media were developed. Different silicone rubber samples were made and tested to evaluate the proposed methods. Rayleigh waves at different frequencies were launched on the surface of different samples; i.e., single layer samples and samples composed of a substrate with known material properties coated with a thin layer of the soft material that is to be characterized. The input vibrations of the actuator and the motion of the sample surface were measured using an accelerometer and a laser Doppler vibrometer, respectively. A transfer function method was used to obtain the complex Rayleigh wavenumbers. Finally, the complex shear and elastic moduli and the loss factor of samples were calculated through the proper modelling using the measured wavenumbers. The results were compared and shown to be in good agreement with those obtained from other measurement methods.

Session 3aEA

Engineering Acoustics and ASA Committee on Standards: Sound Intensity Measurements

Allan J. Zuckerwar, Cochair
Analytical Services & Materials, 1052 Research Dr., Hampton, VA 23666-1340

Robert J. Hickling, Cochair
Sonometrics Inc., 8306 Huntington Rd., Huntington Woods, MI 48070-1643

Invited Papers

8:00

3aEA1. Sound-power flow. Robert J. Hickling (Sonometrics Inc., 8306 Huntington Road, Huntington Woods, MI 48070-1643, hickling.robert@gmail.com)

To understand sound-intensity measurement, it should be realized that sound intensity is sound-power flow per unit area in the direction of sound propagation. Sound intensity can be measured in fluids such as air or water. Generally measurements cannot be made inside a solid. However there may be ways of measuring sound intensity in a solid, if the solid is transparent. Also sound intensity can be computed in a solid and then measured in a surrounding fluid to collate with the calculated results. In general the measurement of sound-intensity is relatively new. It can be measured either as a vector in three or two dimensions or as a single vector component. These features are discussed, together with some examples of sound-intensity measurement. An important use of a single component of sound intensity is the measurement of the sound power of a noise source. Another use is locating the primary sources of noise in operating machinery. The full sound-intensity vector can determine the direction of a noise source. Two vectors can determine the location of the source in space.

8:25

3aEA2. Complex technique of sound intensity measurements and properties of the basic sound fields. Jiri Tichy (Acoustics, Penn State Univ., State College, PA 16804, tichy@engr.psu.edu) and Gary W. Elko (President, mhacoustics LLC, Summit, NJ)

An overview of sensors and signal processing for the measurement of sound intensity and other energy related quantities such as reactive intensity, potential and kinetic energy is summarized. Many examples of energy propagation, vortices formation, radiationn from sources, sound interference and other acoustic phenomena are presented and analyzed.

8:50

3aEA3. Sound intensity measurements in vehicle interiors. Svend Gade (Brüel & Kjær University, Skodsborgvej 307, Nærum DK-2850, Denmark, sgade@bksv.com), Jørgen Hald, and Jakob Mørkholt (Brüel & Kjær University, Brüel & Kjær Sound & Vibration Measurement, Nærum, Sjælland, Denmark)

In some cases it is important to be able to measure not only the total sound intensity on a panel surface in a vehicle cabin, but also the components of that intensity due to sound radiation and due to absorption from the incident field. For example, these intensity components may be needed for calibration of energy flow models of the cabin noise. A robust method based on surface absorption coefficient measurement is presented in this paper.

9:15

3aEA4. Wide band pressure and velocity (p-v) tympanometry with calibrated sound intensity micro-probes. Domenico Stanzial and Giorgio Sacchi (Research Section of Ferrara, CNR - Institute of Acoustics and Sensors "Corbino", v. Saragat, 1, Ferrara, Ferrara 44122, Italy, domenico.stanzial@cnr.it)

Wide band p-v tympanometry can be defined as the measurement of the acoustic immittance of the ear, possibly in normal air pressure condition of the ear canal, and in the full audio frequency range. The most important innovation pioneered by the p-v tympanometry regards the introduction of a different principle of measurement based on the direct acquisition of, both, pressure and velocity (p-v) signals at the ear canal entrance. The measurement can be done by using a pre-calibrated couple of dedicated micro-sensors: an ordinary microphone and an acoustic velocimetric micro-device. This invited speech will report about wide band measurements of ear immittance functions carried out by means of a modified tympanometric probe hosting a pre-calibrated sound intensity micro-probe, and their comparison, with data obtained by standard 226 Hz tympanometry.

9:40

3aEA5. A comprehensive examination of the acoustic vector fields scattered by cylindrical bodies. Robert J. Barton, Geoffrey R. Moss (Naval Undersea Warfare Center Division Newport, 1176 Howell St, Newport, RI 02841, robert.barton@navy.mil), and Kevin B. Smith (Naval Postgraduate School, Monterey, CA)

In this study, the properties of the scattered acoustic vector fields generated by infinite-length and finite-length rigid and elastic cylinders are investigated. Analytical solutions are derived from general acoustic pressure scattering models, and analyzed for wave numbers in the resonance region. The separable active and reactive components of the acoustic intensity are used to investigate the structural features of the scattered field components. Numerical results are presented for the near field and transition regions. A finite element model is developed for both rigid and elastic cylindrical bodies. The finite cylinder model and analysis is then extended to include interactions with an elastic half space. The vector properties of the time-independent complex intensity components and their relations to field energy density quantities are summarized.

10:05–10:20 Break

10:20

3aEA6. Investigating measurement of acoustic intensity for rocket sound field characterization. Kent L. Gee (Dept. of Physics and Astronomy, Brigham Young University, N243 ESC, Provo, UT 84602, kentgee@byu.edu), Jonathan D. Blotter (Dept. of Mechanical Engineering, Brigham Young University, Provo, UT), Scott D. Sommerfeldt, Derek C. Thomas (Dept. of Physics and Astronomy, Brigham Young University, Provo, UT), Kenneth S. Bostwick (Dept. of Mechanical Engineering, Brigham Young University, Provo, UT), and Benjamin Y. Christensen (Dept. of Physics and Astronomy, Brigham Young University, Provo, UT)

An effort to characterize the aeroacoustic source regions and noise environment around launch vehicles has resulted in study of the hardware and processing methods used to calculate acoustic intensity. Because of the extremely harsh measurement environment and other source region characteristics, these investigations have included selection, calibration, and arrangement of microphones and examination of the required pressure and particle velocity estimates. The results of analytical, laboratory, and field experiments are described as a summary of lessons learned during the on-going effort.

10:45

3aEA7. A comparison of directional robustness for endfire versus Blumlein microphone arrays used in hearing aids. Thomas Burns (Starkey Hearing Technologies, 6600 Washington Ave S, Eden Prairie, MN 55416, tom_burns@starkey.com)

An endfire microphone array uses two omnidirectional microphones in a delay-and-sum configuration. A Blumlein array mixes one omnidirectional and one (bi)directional mic. Each can be engineered to provide any 1st order directional pattern. The three critical factors for providing good directionality include the relative sensitivity and phase between the microphones in addition to the placement of the hearing instrument on the user's head. In this context, a directional system is robust if its factors can operate over a wide range of levels without degrading the directional performance. In this study, each array was engineered to have the same aperture spacing and tuned to the same freefield polar pattern; this tuning provided the nominal operating levels. Both arrays were placed in-situ on a measurement manikin and 614 impulse responses were acquired in ten degree resolution on all four microphones for different in-situ positions. The data for each array were combined as described above, and the aforementioned factors were varied around the nominal range of levels in a simple central composite design of experiments. The results of the in-situ directional response show improved robustness for a Blumlein mix.

11:10

3aEA8. Doubly steered array of modal transducers. John L. Butler, Alexander L. Butler, and Michael J. Ciufo (Image Acoustics, Inc, 97 Elm Street, Cohasset, MA 02025, jbutler@imageacoustics.com)

Line or planar arrays steered to end fire generally require quarter wavelength spaced transducer elements with 90 degree sequential phase shifting to attain a unidirectional beam in an end fire direction. Half wave length element separation with 180 degree sequential phase shifts yield end fire but in both directions and at a reduced level. Part of this lower level bipolar directionality is due to the fixed broadside directionality of the transducer elements. However, use of reduced-size, leveraged-circular, acoustical-modal transducer elements, which allow incremental steering, provide a means for substantial end fire steering without the need for quarter wavelength spaced elements. In this case the elements, as well as the array, can be steered in the same direction attaining full-strength, unidirectional end fire steering at half wavelength array spacing. We present the physics of the leveraged-circular, acoustic-modal, transducers and their operation in the monopole, dipole and quadrupole modes along with their implementation and application in doubly steered arrays.

11:35

3aEA9. Underwater vector intensity measurements in the ocean and laboratory. David R. Dall'Osto and Peter H. Dahl (Mechancial Engineering and the Applied Physics Laboratory, University of Washington, Seattle, WA 98103, dallosto@u.washington.edu)

Underwater measurements of the acoustic intensity vector field can be provided by either spatially separated hydrophones or by a sensor measuring a property of particle motion, such as particle acceleration. These measurements are used to formulate the vector intensity as the product of pressure and particle velocity. The magnitude of the vector intensity is not necessarily equal to the plane-wave intensity (the mean square pressure divided by the density and sound-speed of the medium) which is often used to define pressure measurements in terms of intensity. In regions of strong destructive interference, the magnitude of the vector intensity may be greater than the plane-wave intensity. Measurements of an impulsive source on a vertical line array of pressure sensors spanning a shallow sea (60 m) off the coast of South Korea are presented to demonstrate properties of the complex intensity vector field in an ocean waveguide. Here, the vertical complex intensity is formulated by finite-difference methods. These vertical intensity observations in the ocean waveguide have implications on properties of the complete vector field. A laboratory experiment using a tri-axial particle acceleration sensor is presented to provide a connection between measurement of elliptical particle motion and complex intensity.

Session 3aEDa

Education in Acoustics: Hands-On Acoustic Demonstrations for Middle-School Students

Andrew Morrison, Chair
Joliet Junior College, Natural Science Dept., Joliet, IL 60431

Approximately 20 acoustics demonstrations will be set up for local students to interact with at the meeting. These students will be assisted by Acoustical Society of America (ASA) Education in Acoustics members and the Student Council members. Conference participants are encouraged to attend this session to help guide student exploration in acoustics phenomena.

WEDNESDAY MORNING, 24 OCTOBER 2012 BASIE FOYER, 10:00 A.M. TO 12:00 NOON

Session 3aEDb

Education in Acoustics: Undergraduate Research Exposition Poster Session

Mardi C. Hastings, Cochair
George W. Woodruff School of Mechanical Engineering, Georgia Institute of Technology, Atlanta, GA 30332-0405

Preston S. Wilson, Cochair
Applied Research Lab., Univ. of Texas at Austin, Austin, TX 78712-0292

Contributed Papers

All posters will be on display and all authors will be at their posters from 10.00 a.m. to 12.00 noon.

3aEDb1. Effect of boundary diffusers in a reverberation chamber: A preliminary investigation. Jacob R. Adelgren, David T. Bradley (Physics + Astronomy, Vassar College, 124 Raymond Avenue, #745, Poughkeepsie, NY 12604, dabradley@vassar.edu), Markus Mueller-Trapet, and Michael Vorlaender (Inst. of Tech. Acoust., RWTH Aachen Univ., Aachen, Nordrhein-Westfalen, Germany)

In this project, the sound field behavior in a 1:5 scale reverberation chamber has been measured and analyzed. Both hanging diffusers and boundary diffusers have been applied in an effort to increase the chamber's sound field diffusivity, which has been characterized based on the guidelines set forth in several American and international standards, including ASTM C423, ASTM E90, and ISO 354. Objective data from measured impulse responses for several configurations of the diffusers will be presented. These data will be compared to those from the empty chamber and to the criteria from the standards. The relative effectiveness of hanging diffusers vs. boundary diffusers will be discussed.

3aEDb2. Source tracking and scatter localization in a reverberant environment. Laura M. Williamson, Justin D. Risetter, Michael A. Pierfelice, and David R. Dowling (Dept. of Mech. Engineering, University of Michigan, Ann Arbor, MI 48109, drd@umich.edu)

Matched field processing (MFP) has been shown to be effective for remote sound source localization when the receiving array clearly records direct-path sound from a stationary source. Unfortunately, in imperfectly characterized confined environments, source motion, echoes, and reverberation commonly degrade localization performance. This poster presentation describes three acoustic technology development efforts focused on using matched-field processing, with and without the first reflections, (i) to track a moving source, (ii) to improve localization results by adjusting the receiving array geometry, and (iii) to determine the conditions under which a discrete scatterer may be localized. Experiments were conducted in a 1.0-meter-deep and 1.07-meter-diameter cylindrical water tank using a single sound projector, a receiving array of 16 hydrophones, and a linear actuator capable of moving the source at a speed of 0.5 m/s. Measured localization performance is reported for impulsive (100 micro-second) and longer duration signals having center frequencies from 30 kHz to near 100 kHz. As expected, source and scatterer localization accuracy is found to be limited by reverberation. The eventual application of this research is localizing sub-visual cavitation bubbles and other hydroacoustic sound sources in hydrodynamic test facilities. [Work supported by NAVSEA through the Naval Engineering Education Center.]

3aEDb3. Doppler measurement of the motion of a physical pendulum. Jean Paul Ngabonziza and Carl Frederickson (Physics and Astronomy, University of Central Arkansas, Conway, AR 72035, carlf@uca.edu)

The Doppler shift of a reflected acoustic signal has been used to characterize the motion of a physical pendulum. The pendulum is covered with a rough surface to provide specular reflection at any angle. Comparison between theoretical and measured spectrograms will be presented. The measurement dependence on the frequency of the source signal will be explored. Source frequencies will be in the audible range. The system is being evaluated for use with a double physical pendulum modeling the motion of a human leg.

3aEDb4. Auditory change detection with common and uncommon sounds. Amanda L. Heberle and Caroline M. DeLong (Psychology, Rochester Inst. of Tech., 6750 Lakeside Rd., Ontario, NY 14519, amanda.heberle@gmail.com)

Change deafness, the inability to detect a change in an auditory scene, is similar to change blindness, the inability to detect a change in a visual scene. In this experiment, participants were asked to detect changes in auditory scenes (one, three, or five sounds). The sounds were either common sounds (e.g. alarm clock) or uncommon sounds (e.g. science fiction laser). Only one sound was modified in pitch or loudness for half of the trials. Participants were not always able to detect changes in a sound sequence (M = 67.1%) even though they could almost perfectly discriminate between the ten sounds (M = 99.2%). Participants performed best with a scene size of one (M = 82.6%) and worse with a scene size of five (M = 63.8%). Participants performed significantly better with common sounds (M = 74.7%) vs. uncommon sounds (M = 69.1%). Participants were significantly better at detecting a pitch change (M = 80.8%) than a loudness change (M = 53.5%). These results are consistent with the idea of change deafness. We remember the gist of an auditory scene but we don't detect changes in every sound.

3aEDb5. Techniques for measuring ultrasonic tissue properties of cells. Aislinn R. Daniels, Aditya D. Mahajan, Yim J. Rodriguez, and Maria-Teresa Herd (Physics, Earlham College, 801 National Road West, Richmond, IN 47374, herdma@earlham.edu)

Measuring ultrasonic characteristics of cells outside the cellular matrix is of interest. Analyzing these properties at the cellular level may identify qualities specific to a cell type, possibly leading towards cell identification and disease diagnosis without invasive techniques such as biopsy. The purpose of this research is to develop a reliable method of examining cellular characteristics using quantitative ultrasound. Measurements were made using single element transducers at frequencies of 5-50 MHz in a controlled water-tank environment. Speed of sound and attenuation were measured using through transmissions with unfocused transducers, and backscatter was measured using pulse/echo transmissions with focused transducers. To test our experimental techniques we measured high-frequency properties of a tissue mimicking phantom and compared the results to the current standards. As our experiment required testing at a smaller scale than previous tests of these methods, we also created small holding tubes with smaller phantoms of the same material to compare the larger sample measurements. These miniature phantoms show a remarkable consistency in the data obtained when compared to a large phantom, which verifies the applicability of the methods on a small scale.

3aEDb6. Ultrasound characterization of Chinese hamster ovary cells. Aditya D. Mahajan, Aislinn R. Daniels, Yim J. Rodriguez, and Maria-Teresa Herd (Physics, Earlham College, 801 National Road West, Richmond, IN 47374, herdma@earlham.edu)

Ultrasonic characteristics of various tissues are currently not known in enough detail to be used reliably for tissue identification or diagnosis. Analysis at a cellular level as opposed to a tissue level can examine these qualities specific to a cell type. The purpose of this research is to find the ultrasonic tissue characterization of Chinese hamster ovary (CHO) cells to develop a general test for modeling cells. To analyze the characteristics, CHO cells are cultured and prepared into a pellet-sized sample, which are then scanned with single element transducers at high frequencies (5–50 MHz). The speed of sound and attenuation of the pellets are measured using through transmissions with unfocused transducers, and backscatter coefficients are measured using pulse/echo transmissions with focused transducers. This study may establish a possible model and experimental method, in addition to providing a control for the characteristics of other cell types, specifically comparing normal and cancerous cells.

3aEDb7. Quantitative ultrasound characterization and comparison of prostate cancer cells and normal prostate cells. Yim J. Rodriguez, Aislinn R. Daniels, Aditya D. Mahajan, and Maria-Teresa Herd (Physics, Earlham College, 801 National Road West, Richmond, IN 47374, herdma@earlham.edu)

Ultrasound plays an important role in helping diagnose prostate cancer as part of Ultrasound Guided Biopsies; however by better characterizing normal and cancerous prostate cells - and not the actual tumor- this study enhances ultrasound as a first-hand diagnostic tool. Using quantitative ultrasound, normal and cancerous prostate cells were analyzed and compared. Experiments to determine tissue characteristics were performed using single element transducers ranging from 5-50 MHz. Measurements of speed of sound, attenuation, and backscatter coefficients were made. The current results present a valuable insight on the differences between benign and malignant formations by analyzing them at the cellular level. Analysis of cellular behavior at smaller scales provides significant information for better understanding the properties of tumors at a larger scale. These findings contribute to enhance tissue characterization. Moreover, the results obtained present relevant data regarding speed of sound, attenuation, and backscatter coefficients useful for comparative studies and further analysis.

3aEDb8. Designing and building transducers for use in a molecular acoustics experiment. Ashley J. Hicks and William V. Slaton (Physics & Astronomy Department at the University of Central Arkansas, 201 Donaghey Ave, Conway, AR 72035, a.jean.hicks@gmail.com)

This work describes the design, construction, and testing of two capacitance transducers for use in a larger project investigating the molecular absorption of sound in certain gases. The transducers are based on designs presented in the literature, modified to work optimally in our system which consists of 4-inch diameter steel pipe. The experiments will be conducted at atmospheric pressure, eliminating design constraints involved when using high pressure gas. However, work done by Bass & Shields shows that to work in these experiments at atmospheric pressure, transducers must have a frequency range of 1 kHz - 100 kHz. [J. Acoust. Soc. Am. Vol 62, p. 346-353, 1977] The basic concept of our transducer depends upon creating a parallel plate capacitor from metal that is flexible enough to move when a sound wave hits it. Our design utilizes 0.051 mm thickness alumninized Mylar film tensioned with a brass retaining ring over a brass backing plate with both secured to a Delrin plastic base for its electrically insulating properties. We will report on the transducer's frequency response characteristics and initial testing in a send/receive configuration in a sound absorption experiment.

3aEDb9. The role of vowel inherent spectral change in the intelligibility of English vowels spoken by English-, Chinese-, and Korean-native speakers. Stephanie Tchen, Su-Hyun Jin, and Chang Liu (Communication Sciences and Disorders, University of Texas, 1 University Station, Austin, TX 78712, shjin@mail.utexas.edu)

This study is designed to investigate the relationship between Vowel Inherent Spectral Change (VISC) and vowel intelligibility of native and non-native speakers of English. VISC refers to the relatively slow varying changes in formant frequencies associated with each vowel category (Neary and Assman, 1986). Such spectral change has been known to be a major factor in the perception of both the phonetic and phonemic English vowels. In the previous research projects conducted in our lab, we recorded 12 English vowels in /hVd/ format spoken by English native (EN), Chinese native (CN) and Korean native (KN) speakers and examined vowel intelligibility. Overall, vowel intelligibility was significantly higher for native talkers than for non-native talkers. Within non-native speaker groups, CN had slightly higher intelligibility than KN speakers. In this study, we are going to analyze an acoustic feature of the vowels, VISC, spoken by these native and non-native speakers. It is hypothesized that the acoustic differences of vowels spoken by different groups of speakers can account for the variances in vowel intelligibility.

3aEDb10. Floor vibration response to Irish dancing. Valerie K. Smith and Murray S. Korman (Physics Department, U.S. Naval Academy, 572 C Holloway Road, Chauvenet Hall Room 295, Annapolis, MD 21402, korman@usna.edu)

Using Irish step dance impulses of actual techniques, one could use various vibration sensors (B&K microphone and I/O SM11 geophone) to perform a time frequency analysis of the transient response of a supported portable wooden dance floor resulting from forced transient vibration. The steps included (1) a "tap" (the wooden tap on the toe of the shoe hitting the floor), (2) a "stamp" (a combination of the wooden toe and plastic heel hitting the floor simultaneously) and (3) a "shuffle" (a brushing of the wooden tap on the toe once forwards and once backwards against the dance floor). Experiments were performed using laminated veneer lumber (plywood)

supported by four small rubber mounts near the edges. Floors were (a) 1 m square (d = 3/4 inch thick), (b) 0.5 m square (d = 1 inch), (c) 1m by 0.5m (d = 1 inch) and (d) 0.5 m diam (d = 1 inch). FFT analysis of a transient is compared with the geophone/microphone frequency response (same location) using a swept sine loudspeaker excitation. In (b) the lowest frequencies were 110 and 470 Hz for a "tap" at the center. Performance is enhanced. Green's function analysis is presented. [Ji, and Ellis, The Structural Engineer 3rd ser. 72 (1994), p 37- 44.]

3aEDb11. Nonlinear scattering of crossed ultrasonic beams in a constricted flow for the detection of a simulated deep vein thrombosis: Part II. Markus S. Rebersak and Murray S. Korman (Physics Department, U.S. Naval Academy, 572 C Holloway Road, Chauvenet Hall Room 295, Annapolis, MD 21402, korman@usna.edu)

Experiments show that the turbulence generated by a thrombosis (ping pong ball) lodged into a polyethylene cylindrical tube (leg vein) can be detected by the nonlinear scattering at the combination (3.8 MHz sum frequency) from two mutually perpendicular primary wave components (f1=1.8 MHz and f2 = 2.0 MHz). In study (1) the nonlinear scattering at the sum frequency is measured vs. angle from turbulence generated by a submerged water jet. In (2) the thrombosis model vein is submerged in the turbulent flow field, while in (3) the vein remains in place but the thrombosis is removed. Nonlinear scattering at the combination frequency in (1) shows significant Doppler shift and frequency broadening vs. angle. In (2) nonlinear scattering exists but is diminished in amplitude, Doppler shift and spectral broadening, as was expected. In case (3), there is virtually no scattering at the sum frequency, since the vein mitigates the turbulent flow. Results are presented at fine angular increments, and improved alignments to measure mean Doppler shift, standard deviation, skewness and kurtosis vs. scattering angle and thus characterize certain aspects of the turbulence behind the clot. Results extend the original work of Sean M. Mock [J. Acoust. Soc. Am., 129, 2410 (2011)].

3aEDb12. Electro-dynamic soil-plate-oscillator transducer for monitoring the buried vibration response to airborne sound excitation. Amie E. Nardini and Murray S. Korman (Physics Department, U.S. Naval Academy, 572 C Holloway Road, Chauvenet Hall Room 295, Annapolis, MD 21402, korman@usna.edu)

A plastic drum-like anti-personal mine simulant (2 inch diam, by 1 inch tall, by 1/4 inch thick aluminum tube, capped by a 1/4 Al bottom circular plate and an elastic acrylic 1/16 inch thick circular top plate) was constructed. It was then modified to generate an electrical response to top-plate vibration. The mine simulant was converted to a transducer by fastened a sewing machine size bobbin wound with fine enamel copper wire onto the inside face on the bottom plate of the simulant. Next, a rare earth magnet was fastened to the inside surface of the elastic top plate (using a tacky wax). Leads were connected using insulated feed through connectors to complete the transducer design. In testing out the electro-dynamic transducer landmine simulant, results showed that the design was adequate to detect the vibration resonant response using a 2 inch burial depth in a concrete soil box filled with dry masonry sand. Two 12 inch diameter subwoofers were located a meter above the soil box and radiated sound levels on the order of 80-90 dB re 20 micro Pa, were measured near the soil surface. The swept sinusoidal transducer/microphone response exhibited a fundamental resonance near 190 Hz.

3aEDb13. Monitoring the pressure impact of a runner's footstep on the inner sole of the shoe. Jacqueline A. Blackburn and Murray S. Korman (Physics Department, U.S. Naval Academy, 572 C Holloway Road, Chauvenet Hall Room 295, Annapolis, MD 21402, korman@usna.edu)

The research goals are to learn about the biomechanics of human footsteps and apply the knowledge to the understanding of the subsequent wave energy that is transferred into the ground or track surface. Phase (1) research is focused on learning about the time development of the footstep pressure that a runner's foot is transferring to the innersole of an athletic shoe, along with accelerometer information. A Tekscan FlexiForce Model A401 force sensor (25.4 mm sensing diameter and 0.208 mm thick) coated with a polyester substrate was chosen due to its versatile force range if biased and placed in an operational amplifier feedback loop. The response time is <5 microseconds. Two force sensors will be placed on the upper side of a removable innersole to measure force near the ball and heel of the foot. Phase (2) is to use a Digi Intl XBee 802.15.4 Development Kit to communicate (using a wireless link) the transducer voltage responses. The transmitting module is strapped to the runner's ankle. A receiving module receives data streams and connects to a USB computer interface. Phase (3) converts the data streams to measurements of pressure vs. time. Preliminary experiments and a data analysis will be presented.

3aEDb14. Investigation of helmet-to-helmet collisions in football: Experiments using a mechanical lumped element coupled harmonic oscillator model structure for the skull, fluid, and brain mass. Duncan M. Miller and Murray S. Korman (Physics Department, U.S. Naval Academy, 572 C Holloway Road, Chauvenet Hall Room 295, Annapolis, MD 21402, korman@usna.edu)

The study of extreme helmet to helmet football collisions may lead to future precautions to prevent serious head injuries. First, (a) oscillations of the helmet alone (much like a tuning fork) are investigated. Then (b) a thin walled 1/16 inch thick 2 inch tall by 6 inch diameter polycarbonate hoop is used to model the skull (in the n = 2 mode). Next, (c) the hoop is filled with a light weight polyurethane foam to model fluid in the structure. Then (d) a solid brass cylindrical 1 Kg weight is inserted in a carved out slot in the foam. The hoop-foam-brass weight structure is studied in transient vibration. Finally (e) the "skull", "fluid", "brain mass" structure is set in the helmet and cushioned with extraneous foam. A second identical helmet on a pendulum is released with some angular momentum and collides with the helmet (fitted with the structure (e)) that is suspended vertically by its own pendulum cord - initially at rest. In laboratory collision studies three single axis accelerometers are placed on (1) the helmet at rest, (2) the hoop and (3) the end of the cylindrical mass, in an effort to rudimentary model the vibration of the model brain mass.

3aEDb15. Acoustic properties of flight approved materials. Justin Mann, Matthew Sisson, and William Slaton (Physics and Astronomy, University of Central Arkansas, 201 Donaghey Ave, Conway, AR 72035, wvslaton@uca.edu)

The purpose of this project is to analyze the acoustic impedance and absorption properties of various flight approved materials currently and potentially used by NASA in its work with the International Space Station. These materials, consisting of Bisco, Acoustifoam, and other metallic foams, in addition to Durette, Kevlar, and other manufactured felts, will be used in an experimental procedure utilizing an impedance tube. This procedure uses two microphones at fixed positions from a material under test. An audio source, at the opposite end of the testing material, drives sound through the impedance tube and sweeps through a range of distinct frequencies. As the sound propagates down the tube, the two microphones measure the superposition of the driven, incident sound wave and the sound wave reflected off the test material. When used in conjunction with processing software packages, these microphone responses can be recorded and evaluated to produce complex impedance quantities as functions of frequency. By using these results as a means to measure sound absorption coefficients of specific materials, these tested, flight approved materials may be specifically arranged and utilized to both maximize efficiency and minimize excess noise. These possible applications will not only provide scientific data but also potentially affect astronauts on current and future missions for NASA.

3aEDb16. Design characterization and testing of a custom air horn. Frederick J. Ward and William Slaton (Physics & Astronomy Department, University of Central Arkansas, 201 Donaghey, Conway, AR 72034, wvslaton@uca.edu)

Construction and testing of an air horn can provide educational insight into how certain design decisions can influence resulting acoustic properties. Using readily available materials such as pvc pipe and tin sheeting, one can construct an air horn capable of producing sound waves in the 100+ decibel range and frequencies between 150 and 400 Hz. Upon completion of a prototype, many experimental opportunities are available. The degradation of sound intensity over a distance can be tested by use of a sound level meter. Due to the unidirectional behavior of the sound waves from the horn, samples from different distances and angles from the source can provide more understanding of how sound propagates as a wave in an open environment, as opposed to it being a simple directional wave. Upon completion of the

testing, changes to the initial construction design can be implemented to investigate the relationship between the new model's performance and the prototype's. The air horn provides many opportunities for experimentation and testing. For example, frequencies and sound intensity can be altered by making design adjustments such as: diaphragm size, diaphragm material, housing material, bell size, nozzle length, etc. With a better understanding of the inner workings of these sound sources, one could use this design as a blueprint to expand the concept to either much larger or much lower frequency ranges which have applications in many different fields of study.

3aEDb17. Objective and subjective differences between adjacent seats in a multipurpose hall. Michael J. Dick, Jenna M. Daly, and Michelle C. Vigeant (Acoustics Prog. and Lab., Dept. of Mech. Eng., University of Hartford, 200 Bloomfield Avenue, West Hartford, CT 06117, michelle.vigeant@gmail.com)

The purpose of this study was to evaluate differences between adjacent seats in terms of objective measures and subjective perception based on measurements and recordings taken in a mid-sized multipurpose hall. The work is based on the hypothesis that minimal differences should be found between nearby seats. Measurements were taken in three groups of nine adjacent seats. The differences between seats were analyzed in terms of the number of just noticeable differences (JNDs), using 1 dB for strength (G), 5% for early decay time (EDT) and 1 dB for clarity index (C80). The average differences between adjacent seats within a given group were approximately 2.5 JNDs for G, 4 JNDs for EDT and 2.5 JNDs for C80, which implies that these differences might be audible. However, these differences may be within measurement error. Differences in late lateral energy level (GLL), a measure of listener envelopment (LEV), were all less than 1 dB. A total of 35 test subjects were presented binaural recordings from two seat groups and were asked to evaluate LEV and acoustical quality. In general, subjects did not perceive differences between the recordings as hypothesized, with the exception of two cases where differences in LEV were statistically significant.

WEDNESDAY MORNING, 24 OCTOBER 2012 BASIE A1, 7:55 A.M. TO 10:00 A.M.

Session 3aMU

Musical Acoustics: Physics of the Blues

Andrew C. H. Morrison, Chair
Natural Science Dept., Joliet Junior College, Joliet, IL 60431

Chair's Introduction—7:55

Invited Papers

8:00

3aMU1. Physics of the blues—Scales, harmony, and the origin of blues piano styles. J. Murray Gibson (Northeastern University, 360 Huntington Ave, 115 Richards Hall, Boston, MA 02115, m.gibson@neu.edu)

The development of the equal-temperament scale was driven, not by compromise, but by the need to extend the composer's "palette" and increase the harmonic sophistication of western music. Many interesting musical idioms emerged from harmonic artifacts associated with equal temperament. In particular the blues "crushed-note" piano style resulted during the birth to the blues from the melding of this scale with the pure melodies and harmonies. In the presentation I will relate the history from a scientific perspective and illustrate with short keyboard performances. In homage to Kansas - the home of Scott Joplin -related ragtime piano idioms will also be covered. I'll conclude by emphasizing how music is an excellent medium that displays the intimate relationship between science and art, and is a great vehicle to teach science.

8:25

3aMU2. Sound radiated from vibrating trumpet bells and its share in the overall radiated sound of that instrument. Wilfried Kausel, Vasileios Chatziioannou (Inst. of Music Acoustics, Univ. of Music and Performing Arts, Anton-von-Webern-Platz 1, Vienna 1030, Austria, kausel@mdw.ac.at), and Thomas R. Moore (Department of Physics, Rollins College, Orlando, FL)

Dallas Blues, published in 1912 by Hart A. Wand, is often considered to be the first published blues song. In this work the trumpet is the lead voice, and this is still often the case in blues music. Indeed, it is difficult to overstate the importance of this instrument to the development of the genre. Recent research indicates that bell vibrations with circular symmetry, such as breathing modes and axial length oscillations (i.e., piston-like vibrations of the bell or rim), have the potential to affect the input impedance and pressure transfer function of the trumpet. This in turn can significantly affect the sound. Structural and acoustical finite element simulations of an axisymmetric trumpet bell show that the sound level radiated from the vibrating walls is of the same order of magnitude as the sound level radiated by the air column in a wide frequency band around the structural resonances. Since these axial structural resonances have a much wider bandwidth than elliptic modes, it is not necessary that air resonances exactly match structural resonances in order to produce a significant effect. The contribution of axial bell vibrations to the sound of the instrument is shown to be consistent with these simulations.

8:45

3aMU3. Guitar pickups—Where next? Mark French (MET, Purdue University, 138 Knoy Hall, 401 N. Grant St, West Lafayette, IN 47907, rmfrench@purdue.edu) and Davin Huston (Electrical Engineering Technology, Purdue University, West Lafayette, IN)

Electromagnetic guitar pickups have been in wide use for more than 50 years. Not only are the basic operational principles unmodified, some of the most popular pickups are exact copies of 50 year old designs. This situation begs the obvious question of where pickup design is headed and where there are opportunities for improvements. There are only a few underlying physical phenomena that can be harnessed to sense string motion. However, it would seem that there is plenty of room for innovative applications of familiar physical principles. This paper discusses current trends in pickup designs, suggests direction for further development and describes an effort to update the design of inductive pickups.

9:05

3aMU4. Pitch bending in the diatonic harmonica. James P. Cottingham (Physics, Coe College, 1220 First Avenue, Cedar Rapids, IA 52402, jcotting@coe.edu)

Pitch bending by wind instrument players is standard practice in many genres of music, but it is considered essential in blues harmonica playing. Some simple pitch bends involve the coupling of a single reed to a resonator such as the vocal tract of the player, but a full description of pitch bending in the harmonica involves consideration of the coupling between the two reeds, one for each direction of airflow, that share a single reed chamber. The most common pitch bends are those for which the primary reed in the reed chamber sounds a note higher in frequency than that sounded by the secondary reed. In these cases notes can be bent downward to pitches between those sounded by the two reeds. In addition, some players use more advanced techniques to bend notes beyond the frequencies of the chamber reeds. This paper reviews experimental work and theoretical modeling done on pitch bending in the harmonica during the last thirty years. This includes measurements made while pitch bends are produced by players as well as experiments on harmonicas in more conventional laboratory settings.

9:25

3aMU5. The harmonica as a blues instrument: Part I. Gordon Ramsey (Physics, Loyola University Chicago, 6460 N Kenmore, Chicago, IL 60626, gprspinphys@yahoo.com)

This is part one of two presentations on the same project. The modern harmonica, or harp, has been around since the early 19th century. It is typically used in blues, country, rock and roll and folk music. These musical genres are somewhat similar in structure and form, and often borrow ideas from each other. The harmonica is appropriate as a backup to the main vocal melody and instruments due to its rich harmonic structure and subdued intensity. The ability to apply vibrato and gradual slurs make it a perfect instrument to get the "bluesy" idea across. Our harp research group has investigated the physical properties of harmonica structure to illustrate how different structures lead to varied sounds, each of which is appropriate to a particular style of music.

Contributed Paper

9:45

3aMU6. The harmonica as a blues instrument: Part II. Joseph Wiseman, Chris Banaszak, and Gordon Ramsey (Physics, Loyola University of Chicago, Chicago, IL 60626, wsgywrest@gmail.com)

This is part two of two presentations on the same project. The modern harmonica, or harp, has been around since the early 19th century. It is typically used in blues, country, rock and roll and folk music. These musical genres are somewhat similar in structure and form, and often borrow ideas from each other. The harmonica is appropriate as a backup to the main vocal melody and instruments due to its rich harmonic structure and subdued intensity. The ability to apply vibrato and gradual slurs make it a perfect instrument to get the "bluesy" idea across. Our harp research group has investigated the physical properties of harmonica structure to illustrate how different structures lead to varied sounds, each of which is appropriate to a particular style of music.

Session 3aNS

Noise, Physical Acoustics, and Structural Acoustics and Vibration: Launch Vehicle Acoustics

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

R. Jeremy Kenny, Cochair
NASA, Huntsville, AL 35812

Invited Papers

8:00

3aNS1. Extension of a launch pad noise prediction model to multiple engines and directional receivers. Kenneth J. Plotkin (Wyle, 200 12th Street South, Arlington, VA 22202, kenneth.plotkin@wyle.com) and Bruce T. Vu (NASA, Kennedy Space Center, FL)

A model, PAD, has been developed for prediction of noise in the vicinity of launch vehicles, with original application to the mobile launcher and tower for the Ares I launch vehicle. It follows the basic principles of a traditional NASA model (NASA SP-8072, 1971), with updated source components, including impingement, water suppression and acoustic shielding by three dimensional launcher configurations. For application to Space Launch System, the model has been extended to multi-engine vehicles, using the plume merging model developed by Kandula, Vu and Lindsay (AIAA Paper 2005-3091) and accommodating multiple flame holes in the deck. The capability has also been added to account for receiver directivity. This can be significant when predicting the load on the surfaces of enclosures on the launch tower. It is also an issue for model scale tests (such as ASMAT) where microphones and their mounts are not small enough to be omnidirectional, and thus do not measure free field levels. (Work supported by the National Aeronautics and Space Administration.)

8:20

3aNS2. Full-scale rocket motor acoustic tests and comparisons with models: Revisiting the empirical curves. Michael M. James, Alexandria R. Salton (Blue Ridge Research and Consulting, LLC, 15 W. Walnut St., Suite C, Asheville, NC 28801, michael.james@blueridgeresearch.com), and Kent L. Gee (Department of Physics and Astronomy, Brigham Young University, Provo, UT)

Development of the next-generation space flight vehicles has prompted a renewed focus on rocket sound source characterization and near-field propagation modeling. Improved measurements of the noise near the rocket plume are critical for direct determination of the noise environment. They are also crucial in providing inputs to empirical models and in validating computational aeroacoustics models. NASA's SP 8072 acoustic load prediction model is a widely used method for predicting liftoff acoustics. SP-8072 implements two Distributed Source Methods (DSM-1 and DSM-2), which predict the loading as the sum of the radiated field from each source distributed along the plume. The prediction model depends largely on empirical curve fits computed from historical data to determine the source power and frequency content at distances along the plume. Preliminary results from measurements of a static horizontal firing of Alliant Techsystems Orion 50S XLG performed in Promontory, UT are analyzed with respect to the historical data that drives the SP-8072 prediction model.

8:40

3aNS3. Full-scale rocket motor acoustic tests and comparisons with models: Updates and comparisons with SP-8072. Michael M. James, Alexandria R. Salton (Blue Ridge Research and Consulting, LLC, 15 W. Walnut St., Suite C, Asheville, NC 28801, michael.james@blueridgeresearch.com), and Kent L. Gee (Department of Physics and Astronomy, Brigham Young University, Provo, UT)

Development of the next-generation space flight vehicles has prompted a renewed focus on rocket sound source characterization and near-field propagation modeling. Measurements taken during a static horizontal firing of Alliant Techsystems Orion 50S XLG are compared to the predicted levels produced from NASA's SP-8072 Distributed Source Methods (DSM-1 and DSM-2). Two modifications to the SP 8072 prediction model are considered in regards to the source directivity and the source power spectral distribution. All models considered provide a good first-order approximation given an appropriate total acoustic sound power. However, a more physical model is needed to adequately map out the critical near-field region as well as the far-field propagation. A new intensity-based measurement system and corresponding procedures are currently being developed for determining this near field energy flow and for achieving source characterization capabilities beyond traditional pressure measurements. These advances are believed to be a step toward improved measurements and modeling of the rocket plume.

9:00

3aNS4. Scale model acoustic test overview. Douglas Counter (NASA George C. Marshall Space Flight Center, Huntsville, AL) and Janice Houston (Jacobs ESTS Group, 1500 Perimeter Pkwy, Suite 400, Huntsville, AL 35806, janice.d.houston@nasa.gov)

Launch environments, such as lift-off acoustic (LOA) and ignition overpressure (IOP), are important design factors for any vehicle and are dependent upon the design of both the vehicle and the ground systems. LOA environments are used directly in the development of vehicle vibro-acoustic environments and IOP is used in the loads assessment. The Scale Model Acoustic Test (SMAT) program was implemented to verify the Space Launch Systems LOA and IOP environments for the vehicle and ground systems including the Mobile Launcher (ML) and tower. The SMAT is currently in the design and fabrication phase. The SMAT program is described in this presentation.

9:20

3aNS5. Frequency-based spatial correlation assessments of the Ares I subscale acoustic model test firings. Robert J. Kenny (NASA, Mail Stop ER42, Bldg 4203, Marshall Space Flight Center, Huntsville, AL 35812, robert.j.kenny@nasa.gov) and Janice Houston (Jacobs Engineering, Huntsville, AL)

The Marshall Space Flight Center has performed a series of test firings to simulate and understand the acoustic enviroments generated for the Ares I liftoff profiles. Part of the instrumentation package had special sensor groups to assess the acoustic field spatial correlation features for the various test configurations. The spatial correlation characteristics were evaluated for all of the test firings, inclusive of understanding the diffuse to propagating wave amplitude ratios, the acoustic wave decays, and the incident angle of propagating waves across the sensor groups. These parameters were evaluated across the measured frequency spectra and the associated uncertainties for each parameter were estimated.

9:40

3aNS6. Prediction of nonlinear propagation of noise from a solid rocket motor. Michael B. Muhlestein, Kent L. Gee, Derek C. Thomas, and Tracianne B. Neilsen (Department of Physics and Astronomy, Brigham Young University, Provo, UT 84602, mimuhle@gmail.com)

The extreme sound pressure levels radiated from rocket motors are such that nonlinear propagation effects can be significant. Here, free-field nonlinear propagation has been modeled for noise produced by a GEM-60 solid rocket motor. Measured waveforms were used as inputs into a numerical model based on the generalized Burgers equation. In both temporal and frequency domains the nonlinear predictions are significantly closer to the measured signals than free-field, linear predictions. In the temporal domain, shock coalescence and a transition from the weak-shock regime of propagation to the beginning of the old-age regime are clearly observed in both the nonlinear prediction and the measured data. These phenomena are completely missing in the linear prediction. In the frequency domain, continual transfer of energy upward in the spectrum reduces attenuation of high-frequency components when compared to predictions from the linear model. Various comparisons are made as a function of input distance for two different radiating angles from the rocket plume; these comparisons illustrate the importance of including nonlinear effects in rocket noise propagation modeling.

10:00–10:20 Break

10:20

3aNS7. Scale model tests for acoustic prediction and reduction of epsilon launch vehicle at lift-off. Seiji Tsutsumi (JEDI, JAXA, 3-1-1 Yoshinodai, Chuuou, Sagamihara, Kanagawa 252-5210, Japan, tsutsumi.seiji@jaxa.jp), Tatsuya Ishii (APG, JAXA, Chofu, Tokyo, Japan), Kyoichi Ui, Sinichiro Tokudome (Epsilon Rocket Project Team, JAXA, Tsukuba, Ibaraki, Japan), and Kei Wada (Science Service, Inc., Chuuou-ku, Tokyo, Japan)

Test campaign using 1/42-scale model is conducted to predict acoustic level of the Epsilon launch vehicle at lift-off. Analogy between sub-scale and full-scale tests is investigated to obtain the same feature of the acoustics. Methodology to correct the measured data obtained in the sub-scale test for predicting the full-scale environment is also clarified in this study. The acoustic results around the practical shape of the launch-pad are successfully obtained. The parametric studies are conducted to reduce noise level in the test campaign with the help of the numerical simulation, and the effect for noise reduction is observed up to 5dB in 1/1-octaveband SPL.

10:40

3aNS8. Acoustic measurement of 1:42 scale booster and launch pad. Tatsuya Ishii, Seiji Tsutsumi, Kyoichi Ui, Shinichiro Tokudome (Japan Aerospace Exploration Agency, 7-44-1 Jindaijihigashi-machi Chofu-shi, Tokyo 182-8522, Japan, ishii.tatsuya@jaxa.jp), Yutaka Ishii (Bruel & Kjaer Japan, Tokyo, Japan), Kei Wada (Science Service, Inc., Tokyo, Japan), and Satoru Nakamura (Tokyo University of Science, Tokyo, Japan)

This paper describes the acoustic measurement of the subscale booster and launch pad. The 1:42 scale solid propellant booster was settled over the launch pad model. The launch pad model was designed to deflect the hot and high speed plume, aimed at mitigating the feedback Mach waves toward the vehicle. The launch pad plays a role in attenuating the sound due to the impingement of the plume and the deflector. To investigate the acoustic field with a different booster height, acoustic measurement was carried out. The measurement involved the conventional acoustic measurement and the sound source localization. The conventional measurement employed the near-field microphones around the booster model and the far-field microphones on an arc centered on the booster nozzle or the impingement point of the plume and the launch pad. In the sound source localization, a phased array microphone system was settled to focus the deflector exit. The obtained acoustic data helped revise the design of the launch pad model.

11:00

3aNS9. Analysis of noise from reusable solid rocket motor firings. Kent L. Gee (Dept. of Physics and Astronomy, Brigham Young University, N243 ESC, Provo, UT 84602, kentgee@byu.edu), R. Jeremy Kenny (NASA Marshall Space Flight Center, Huntsville, AL), Trevor W. Jerome, Tracianne B. Neilsen (Dept. of Physics and Astronomy, Brigham Young University, Provo, UT), Christopher M. Hobbs (Wyle Laboratories, Arlington, VA), and Michael M. James (Blue Ridge Research and Consulting, Asheville, NC)

As part of investigations into the design of next-generation launch vehicles, near and far-field data were collected during horizontal static firings of reusable solid rocket motors. In addition to spectral analysis at individual microphone locations, the spatial variation of overall and one-third octave band pressure levels at sideline and polar arc arrays is considered. Analysis of the probability density functions reveals positively skewed pressure waveforms, but extreme skewness in the first-order estimate of the time derivative because of the presence of significant acoustic shocks.

Contributed Papers

11:20

3aNS10. Decomposition of military jet aircraft mixing noise into fine and large-scale turbulent components. Tracianne B. Neilsen, Kent L. Gee, Alan T. Wall (Dept. of Physics and Astronomy, Brigham Young University, N311 ESC, Provo, UT 84602, tbn@byu.edu), and Michael M. James (Blue Ridge Research and Consulting, Provo, Utah)

Many far-field measurements of laboratory-scale jet noise have shown good agreement with the two similarity spectra developed to represent the contributions of fine-scale and large-scale turbulent structures [Tam et al., AIAA paper 96-1716, 1996]. Measurements near an F-22A Raptor provide a means to study how accurately the similarity spectra describe the noise from a full-scale, high-performance, jet engine. Comparisons have been made using ground-based microphones at 60° to 150° for three engine conditions: intermediate, military and afterburner, with more detailed analyses than described previously [Neilsen et al., J. Acoust. Soc. Am. 129, 2242 (2011)]. The good agreement with Tam's predictions - the fine-scale spectrum at upstream and sideline angles and the large-scale spectrum in the maximum radiation direction - permits a quantitative analysis of the contributions of the two spectra at other angles. The radiated spectra and overall levels for all three engine conditions have been decomposed into contributions from the two spectra as functions of angle. Of particular interest is the appreciable contribution of fine-scale turbulence aft of the maximum radiation directions at military and afterburner conditions. [Work sponsored by the Office of Naval Research.]

11:35

3aNS11. Near-field correlation and coherence of military jet noise. Blaine M. Harker, Kent L. Gee, Tracianne B. Neilsen, Alan T. Wall (Dept. of Phys. and Astronomy, Brigham Young Univ., 562 N 200 E # 17, Provo, UT 84606, blaineharker@byu.net), Michael M. James (Blue Ridge Res. and Consulting, Asheville, Utah), and Sally A. McInerny (Dept. of Mech. Eng., Univ. of Louisiana, Lafayette, LA)

Correlation and coherence analyses in the near field of military jet noise provide insight into source and radiation characteristics. Data were measured parallel to the exhaust centerline of an F-22A Raptor and spatial correlation and coherence values were calculated. Low spatial correlation at the sideline indicates radiation dominated by multiple incoherent sources. In contrast, the downstream region is characterized by high spatial correlation, suggesting radiation primarily from large-scale turbulent structures. Variations in spatial correlation in the axial direction can be related to the spectral dependence on measurement location, which supports the idea of a two-source jet-noise model. Coherence calculations, which decompose the correlation information into narrow frequency bands, further support this idea. [Work supported by the Office of Naval Research.]

WEDNESDAY MORNING, 24 OCTOBER 2012 LESTER YOUNG A, 8:00 A.M. TO 11:45 A.M.

Session 3aPA

Physical Acoustics: Thermoacoustics, Physics, and More

Josh R. Gladden, Chair
Physics & NCPA, University of Mississippi, University, MS 38677

Contributed Papers

8:00

3aPA1. Dynamic stabilization of the Rayleigh-Bénard instability in a cubic cavity. Randy M. Carbo (Graduate Program in Acoustics, Penn State University, State College, PA), Robert W. Smith, Matthew E. Poese (Applied Research Laboratory, Penn State University, P.O. Box 30, State College, PA 16804, rws100@psu.edu), and Anand Swaminathan (Graduate Program in Acoustics, Penn State University, State College, PA)

The dynamic stability of the Rayleigh-Bénard instability with vertical vibration in a cubic container is computationally modeled. Two periodic parametric drives are considered (sinusoidal and rectangular), as well as two thermal boundary conditions on the sidewalls (insulating and conducting). The linearized equations are solved using a spectral Galerkin method and Floquet analysis. Floquet analysis recovers both the synchronous and the subharmonic regions of instability. The conditions necessary for dynamic stability are reported for Rayleigh numbers from critical to 10^7 and for Prandtl numbers in the range of 0.1-7.0, and the approach produces maps over a wide range of Rayleigh number and vibration parameters for stability. The linear model is compared to data set available in the literature [G. W. Swift and S. Backhaus J. Acoust. Soc. Am. **126**, 2273 (2009)] where the performance of system simulating an inverted pulse tube cryocooler is measured. The relevant instability for this case is the synchronous instability. Over this limited data set, the model appears to bound the empirically observed conditions for stability, but in some cases the model would seem to predict significantly higher required periodic acceleration amplitudes that appear to have been observed by Swift. Comparison with another data set is on-going. [Research supported by the Office of Naval Research and ARL Exploratory and Foundational Research Program.]

8:15

3aPA2. Thermoacoustic device for nuclear fuel monitoring and heat transfer enhancement. Randall A. Ali, Steven L. Garrett (Grad. Prog. in Acoustics, Penn State University, Grad. Prog. in Acoustics, P.O. Box 30, State College, PA 16804, randallali@gmail.com), James A. Smith, and Dale K. Kotter (Fundamental Fuel Properties Group, Idaho National Laboratory, Idaho Falls, ID)

The Fukushima Dai'ichi nuclear disaster of 2011 exposed the need for self-powered sensors that could transmit the status of the fuel rods within the reactor and in spent fuel ponds that was not dependent upon availability of external electrical power for either sensing or telemetry. One possible solution is the use of a thermoacoustic standing wave engine, incorporated within a fuel rod, which is heated by the nuclear fuel. The engine's resonance frequency is correlated to the fuel rod temperature and will be transmitted by sound radiation through the reactor's or storage pond's surrounding water. In addition to acting as a passive temperature sensor, the

thermoacoustic device will serve to enhance heat transfer from the fuel to the surrounding heat transfer fluid. When activated, the acoustically-driven streaming flow of the gas within the fuel rod will circulate gas away from the nuclear fuel and convectively enhance heat transfer to the surrounding coolant. We will present results for a thermoacoustic resonator built into a Nitonic® 60 (stainless steel) fuel rod that can be substituted for conventional fuel rods in the Idaho National Laboratory's Advanced Test Reactor. This laboratory version is heated electrically. [Work supported by the U.S. Department of Energy.]

8:30

3aPA3. Addition of a series spring to a lumped element model of a novel thermoacoustic refrigerator. Eric C. Mitchell, Steven L. Garrett, Robert W. M. Smith, Matthew E. Poese, and Robert M. Keolian (Applied Research Laboratory, Penn State University, State College, PA 16804, emitchell756@gmail.com)

A lumped element model is introduced for an "in-line" traveling-wave thermoacoustic refrigerator, based on a recent patent by Backhaus and Keolian [US Pat. No. 7,908,856 (22 Mar 2011)]. This model couples three electro-mechanical motors to the acoustical domain using a flexure sealed piston. Results of this lumped-element model were found to be in good agreement with a DELTAEC model. Since large displacements result in flexure fatigue, the addition of a series spring between the motor and piston was evaluated to reduce flexure seal displacement while matching the optimum load impedance. The assignment of the correct dynamic mass of the series spring when both the motor and flexure ends of the spring are moving has not been addressed previously in the literature. To determine the two effective masses required at each end of the spring, the spring is discretized into a large number of elements and is converted into a simplified circuit model to assign values to lumped-element components. These values depend upon an initial knowledge of both end velocity magnitudes and phases. This enables an iterative solution for the effective spring masses since velocities need to be known for effective masses to be determined and vice versa. [Work supported by the Applied Research Laboratory and the U.S. Department of Energy.]

8:45

3aPA4. Nusselt numbers of laminar, oscillating flows in stacks and regenerators with pores of arbitrary cross-sectional geometry. John Brady (Los Alamos National Laboratory, MSD429, Los Alamos, NM 87544, jbrady@lanl.gov)

General expressions for the Nusselt numbers of laminar oscillating flows within the pores of stacks and regenerators are derived from thermoacoustic theory developed by Rott and Swift. These expressions are based on bulk (velocity-weighted, cross-sectionally averaged) temperature, rather than the cross-sectionally averaged temperature. Two cases are considered: flow with oscillating pressure and no external temperature gradient, and oscillating velocity within an external temperature gradient and negligible pressure oscillations. These expressions are then applied to parallel plates, circular pores, rectangular pores, and within the boundary layer limit. Steady-flow Nusselt numbers are recovered when the thermal penetration depth is at least as great as the hydraulic radius of the pore. In addition, temperature and flow profiles within this regime are like those of steady flows.

9:00

3aPA5. Complex intensity in circular ducts containing an obstruction. Ray Kirby (Mechanical Engineering, Brunel University, Uxbridge, Middlesex UB8 3PH, United Kingdom, ray.kirby@brunel.ac.uk), Jevgenija Prisutova (School of Engineering, University of Bradford, Bradford, West Yorkshire, United Kingdom), Wenbo Duan (Mechanical Engineering, Brunel University, Uxbridge, Middlesex, United Kingdom), and Kirill Horoshenkov (School of Engineering, University of Bradford, Bradford, West Yorkshire, United Kingdom)

Sound intensity may be defined as a complex quantity in which the real part of the intensity is related to the magnitude of the local mean energy flow, and the imaginary part to the local oscillatory transport of energy. By treating intensity as a complex quantity it is possible to visualise energy flow in a different way and this has the potential to aid in the interpretation of, say, sound fields scattered by objects. Accordingly, the sound field scattered by an object placed in a semi-infinite circular duct is examined here. Experimental measurements of complex intensity are obtained in three (orthogonal) directions using a Microflown intensity probe, and measurements are compared to predictions obtained using a finite element based theoretical model. Comparisons between prediction and measurement are undertaken for both plane wave and multi-modal sound fields and here it is noted that when at least one higher order mode propagates it becomes more difficult to obtain good agreement between prediction and experiment for the complex intensity.

9:15

3aPA6. Modeling interferometric sensor response to the photoacoustic effect in layered systems. Logan Marcus, Richard Raspet, and Vyacheslav Aranchuk (NCPA, University of Mississippi, 1 Coliseum Drive, NCPA Room 1101, University, MS 38677, lsmarcus@olemiss.edu)

Chemical elements and molecules have characteristic absorption spectra that can be used for identification and detection. The photoacoustic effect has previously been used to perform spectroscopic measurements. We describe a modified photoacoustic spectroscopy method to accomplish standoff detection of thin layers of materials using an interferometric sensor. The interferometric sensor measures changes to the optical path length of an interrogation beam incident on a surface. We have developed a detailed model of the physical processes that result when a system comprised of a thin layer on a larger substrate is excited by the absorption of a modulated Gaussian laser beam. The modulated excitation beam generates heating in the sample which leads to surface motion, modulation of the temperature profile in the adjacent air, and an acoustic wave which all contribute to the signal. The model allows for the calculation of the measured signal of the interferometric sensor using the physical properties of the sample and the excitation beam. The presented model and experimental work are all applied to an idealized system comprised of a thin layer of gold on a low absorptivity borosilicate substrate to validate the computation. Future work will extend to a variety of layers and substrates.

9:30

3aPA7. Acoustic radiation force and radiation torque on Rayleigh particles. Tiago P. Lobo and Glauber T. Silva (Physical Acoustics Group - IF, UFAL, Instituto de Física, Campus A. C. Simões - Av. Lourival Melo Mota, s/n, Cidade Universitária, Maceió, Alagoas 57072-900, Brazil, tomaz.glauber@gmail.com)

In this work, the acoustic radiation force and radiation torque exerted by an arbitrary shaped wave on a spherical particle in the Rayleigh approximation (i.e. the incident wavelength is much smaller than the particle dimensions) are discussed. The host fluid in which the particle is suspended is assumed to be inviscid. Expressions for the force and the torque are obtained in terms of the incident acoustic fields, namely pressure and particle velocity. As it turns out, the obtained radiation force expression represents a generalization of Gor'kov's formula [Sov. Phys. Dokl. 6, 773-775 (1962)]. Moreover, the radiation torque can be expressed in terms of the incident Reynolds' stress tensor. The method is applied to calculate both radiation force and radiation torque produced by Bessel beams. It is demonstrated that only the first-order Bessel vortex beam generates radiation torque on a particle placed in the beam's axis. In addition, results involving off-axial particles and Bessel beams of different order are illustrated.

9:45

3aPA8. Modeling of photoacoustic Raman spectroscopy with dissipation. David Chambers and Chance Carter (Lawrence Livermore National Laboratory, PO Box 808, Livermore, CA 94551, chambers2@llnl.gov)

Photoacoustic Raman spectroscopy (PARS) is a technique used to identify chemical species mixed in a gas or liquid based on their pattern of vibrational energy levels. Raman spectroscopy differs from the more familiar absorption spectroscopy by using a nonlinear two-photon process that can be more sensitive to small differences in vibrational energy levels. Thus it can detect defect sites in solid-state optical materials, or low concentrations of chemical species in gases. The Raman scattering process generates acoustic pulses that can be detected with a microphone. In this talk we present an

overview of PARS and present an updated model that includes dissipation for the production of acoustic pulses from the energy deposited in the medium during the Raman scattering process. We also show some preliminary measurements of the process for Raman conversion in hydrogen. This work performed under the auspices of the U.S. Department of Energy by Lawrence Livermore National Laboratory under Contract DE-AC52-07NA27344.

10:00–10:15 Break

10:15

3aPA9. Impact of rigid sphere scattering on measurement of acoustic shocks. Michael B. Muhlestein, Derek C. Thomas, and Kent L. Gee (Department of Physics and Astronomy, Brigham Young University, Provo, UT 84602, mimuhle@gmail.com)

Multi-microphone arrays embedded in a rigid spherical housing have been used to estimate field quantities such as vector intensity. However, the measured pressure waveforms are modified by the scattering of the incident pressure wave on the sphere. The frequency response function and the corresponding impulse response function for microphones in a sphere can be created using Mie scattering theory. This permits analysis of the scattering effect on pressure measurements for shock-containing broadband waveforms, such as those produced by solid rocket motors. The abrupt pressure rises associated with shocks are seen to be either overestimated or underestimated depending on the angle of the measuring microphone. These shocks are the most affected portion of the waveform due to their high frequency content. In addition, deconvolution of measured rocket signals using the predicted impulse responses of the microphones in the array provides an apparently consistent estimate of the free-field signal at the probe center.

10:30

3aPA10. Actuating micro devices with acoustic white noise. Raul Esquivel-Sirvent (Instituto de Fisica, UNAM, Apdo Postal 20-364, Mexico DF 01000, Mexico, raul@fisica.unam.mx)

We present a theoretical calculation of the actuation of a model micro system, such as a microelectromechanical system (MEMS), by the acoustic pressure of white noise. This is a classical analog of the Casimir effect, thus the name Casimir acoustic pressure. Unlike the quantum case, the acoustic Casimir pressure can be attractive or repulsive depending on the frequency bandwidth of the acoustic noise. As a case study, a one-degree-of-freedom simple-lumped system in an acoustic resonant cavity is considered. By properly selecting the frequency bandwidth of the acoustic field, the acoustic pressure can be tuned to increase the stability in existing microswitch systems by selectively changing the sign of the force. The acoustic intensity and frequency bandwidth are introduced as two additional control parameters in capacitive micorswitches. Applications of this concept in microfluidics will be also discussed.

10:45

3aPA11. Infrasound scattering by the Lamb dipole vortex. Konstantin Naugolnykh (NOAA/Zeltech, 325 Broadway, Boulder, CO 80305, konstantin.naugolnykh@noaa.gov)

The infrasound scattering by the Lamb dipole is considered in the present paper in Born approximation and using the asymptotic presentation for the Green function of the scattered field. The Lamb dipole consists of two vortexes rotating in the opposite direction what induces the specific features of process of scattering infrasound by this object. They probably accompanied the infrasound generation by high-intensity atmospheric events such as cyclones.

11:00

3aPA12. The nonlinearity parameter, B/A, in FC-43 Fluorinert up to 373 K and 13.8 MPa. Blake T. Sturtevant, Cristian Pantea, and Dipen N. Sinha (Materials Physics and Applications, Los Alamos National Laboratory, PO Box 1660, Los Alamos, NM 87545, bsturtev@lanl.gov)

Acoustic imaging systems based on the parametric array concept utilize a nonlinear medium for mixing high frequency sound into a beam of low frequency collimated sound. Fluorocarbon fluids are very appealing as nonlinear mixing media with values of the nonlinearity parameter, B/A, typically greater than 10 at ambient temperature and pressure. To design acoustic imaging systems for high temperature and high pressure environments, such as found in geothermal and petroleum wells, it is important to know how B/A varies with temperature and pressure. This work reports the determination of B/A in FC-43 at temperatures up to 373 K and pressures up to 13.8 MPa using the thermodynamic method. Sound velocities were measured using Swept Frequency Acoustic Interferometry at 11 pressures between ambient and 13.8 MPa along 6 isotherms between ambient and 373 K. A 3rd order least-squares fit of measured sound speeds was used to determine temperature and pressure dependence. The B/A of FC-43 was found to increase with both temperature and pressure near ambient conditions and to go through a maximum around 340 K and 6 MPa.

11:15

3aPA13. Acoustic streaming in channel bifurcated by an elastic partition. Megha Sunny, Taoufik Nabat, and Charles Thompson (Electrical and Computer Engineering, University of Massachusetts Lowell, 1 University Ave, Lowell, MA 01854, taoufik_nabat@student.uml.edu)

Fluid motion in a narrow channel partitioned along its long axis by a flexible membrane is examined. The enclosed fluid is excited by the time harmonic displacement in the channel cross-section. The acoustic streaming that ensues and its impact of microfluidic transport is of particular interest in this work. A three-dimensional regularized Stokeslet based analysis is presented for fluid motion in the low streaming Reynolds number regime. The effect of frequency modulation on fluid motion is examined.

11:30

3aPA14. Acoustic scattering from dual frequency incident fields. Chrisna Nguon, Max Denis, Kavitha Chandra, and Charles Thompson (Electrical and Computer, University of Massachusetts Lowell, Lowell, MA 01854, chrisna_Nguon@student.uml.edu)

The pressure field produced by the spatial interaction of two high frequency incident beams in a three-dimensional scattering object is investigated. Of particular interest is the radiated pressure produced in response to the difference-frequency component generated from the non-linear interaction between the beams and the scattering medium. The influence of high acoustic contrast and resonant scattering is considered in the analysis. This work presents a computational study of the scattered pressure that results from the Reynolds stress in a fluid scatterer. Using Padé approximants, it is shown that the stress tensor can be computed using a uniform expansion in the contrast gauge for the scattered pressure. This allows one to investigate scattering volumes characterized by high compressibility contrast.

Session 3aPP

Psychological and Physiological Acoustics: Perception and Models

G. Christopher Stecker, Chair
Speech and Hearing Sciences, University of Washington, Seattle, WA 98105

Contributed Papers

8:30

3aPP1. Infants' ability to separate superimposed vowels. Lynne Werner, Bonnie Lau, and Ashley Flad (Speech & Hearing Sciences, University of Washington, 1417 North East 42nd Street, Seattle, WA 98105-6246, lawerner@u.washington.edu)

Three- and seven-month-old infants were tested using an observer-based procedure in three tasks to assess sound source segregation and selective attention. The stimuli were tokens of the vowels /a/ and /i/, spoken by two male talkers, 519 ms in duration, presented at 70 dB SPL. Success was defined as achieving 80% correct in fewer than 40 test trials. In the first task, infants heard one vowel spoken by one talker repeated at 1319 ms intervals. They learned to respond when the talker changed on one repetition of the vowel. In the second task, the tokens of the two talkers producing the same vowel were superimposed. Infants heard combined tokens repeatedly and learned to respond when the vowel produced by one talker changed. In the third task, either talker could produce the changed vowel. Infants learned to respond when one talker, but not the other, produced the changed vowel. Nearly all infants succeeded in the first two tasks. Nearly all 7-month-olds, but few 3-month-olds succeeded at the third task. These results suggest that the ability to selectively attend to one of two easily discriminable voices matures after the ability to segregate those voices. [Work supported by R01DC00396 and P30DC04661.]

8:45

3aPP2. Off-frequency masking effects on intensity discrimination. Harisadhan Patra (Audiology & Speech Pathology, Bloomsburg University, 226 CEH, 400 E 2nd Street, Bloomsburg, PA 17815, hpatra@bloomu.edu), Scott Seeman (Department of Communication Sciences & Disorders, Illinois State University, Normal, IL), Adam Burkland, Joseph Motzko, and Erin Lolley (Audiology & Speech Pathology, Bloomsburg University, Bloomsburg, PA)

Intensity discrimination, where a listener detects an intensity increment in an equal duration sinusoid or pedestal, is often used as a measure of intensity resolution. Intensity discrimination may be considered as tone-in-tone masking, where the pedestal is the masker and the increment is the signal. Despite the similarity between intensity discrimination and tone-in-noise masking, research suggests that a high-pass noise outside the critical band centered on the signal frequency adversely affects listeners' intensity-discrimination thresholds. The present study examines the limits of off-frequency masking effects on intensity discrimination in five normal-hearing young adults. Detection thresholds for a 50-ms increment, added to a 50-ms-long 1000-Hz pedestal in phase, were obtained in quiet and notched-noise (NN) conditions. The pedestal and noise levels were 60 dB SPL. NN stimuli were generated by filtering telegraph noise. The low-frequency cutoffs of the NN-notches were 188, 250, 375, 500, and 750 Hz while the high-frequency cutoffs were 1500, 2000, 3000, 4000, and 6000 Hz. The detection thresholds were poorer in NN conditions than in quiet, even when cutoff frequencies were more than one octave away from the signal frequency. Effects of off-frequency maskers on the psychometric functions are discussed. [Supported by BU research and scholarship grant.]

9:00

3aPP3. Perceptual weights for loudness reflect central spectral processing. Suyash N. Joshi and Walt Jesteadt (Psychoacoustics Laboratory, Boys Town National Research Hospital, 555 N 30th St., Omaha, NE 68131, Suyash.Joshi@boystown.org)

Weighting patterns for loudness obtained using the reverse correlation method are thought to reveal the relative contributions of different frequency regions to total loudness, the equivalent of specific loudness. Current models of loudness assume that specific loudness is determined by peripheral processes such as compression and masking. Here we test this hypothesis using 20-tone harmonic complexes (200Hz f0, 200 to 4000Hz, 250 ms, 65 dB/Component) added in opposite phase relationships (Schroeder positive and negative). Due to the varying degree of envelope modulations, these time-reversed harmonic complexes have been shown to produce different outputs at the basilar membrane and different amounts of forward and simultaneous masking. The perceptual weights for loudness did not differ for these two complexes. To determine whether the level rove introduced to obtain weights had changed the fundamental differences in the stimuli, a similar level rove ($\pm$8 dB) was introduced on each component of Schroeder positive and negative forward maskers. The Schroder negative maskers continued to be more effective. These results suggest that perceptual weights for loudness are not completely determined by peripheral processes and reflect a central frequency weighting template. [Work supported by NIH R01 DC011806 and P30 DC004662.]

9:15

3aPP4. Temporal weighting of interaural time and level differences carried by broadband noises. G. C. Stecker (Speech and Hearing Sciences, University of Washington, 1417 NE 42nd St, Seattle, WA 98105, cstecker@uw.edu)

Localization of real sounds involves integrating acoustic spatial cues as they evolve over time. This study measured binaural sensitivity over time, in the form of temporal weighting functions (TWFs) for trains of of noise bursts. Each stimulus comprised sixteen 1-ms bursts of white noise, presented at an interval (ICI) of 2 or 5 ms. In separate conditions, noise samples were either repeated ("frozen") or newly generated ("fresh") across bursts. On each of many trials, listeners indicated the apparent lateral position of a stimulus along a horizontal scale displayed on a touch-sensitive device. Lateral positions varied across trials as interaural time (ITD) and level (ILD) differences ranged +/-500 μs ITD or +/-5 dB ILD. Interaural differences of individual bursts in each train received additional random variation (ranging +/-100 μs and +/-2 dB) to allow calculation of TWFs by multiple linear regression of normalized responses onto per-burst ITD and ILD values. Consistent with past studies, TWFs for "frozen" noise-burst trains demonstrated large ICI-dependent weights on the initial burst ("onset dominance"), elevated weights near offset, and lower weights for interior bursts. Flatter TWFs, smaller onset/offset weights, and greater interior weights were measured for "fresh" vs "frozen" noise burst trains. [Supported by R01 DC011548.]

9:30

3aPP5. The relative contribution of dynamic and spectral cues in virtual sound source localization. Chengyun Zhang and Bosun Xie (Physics Department, School of Science, South China University of Technology, Wushan Rd. 381., Tianhe District, Guangzhou 510640, China, phbsxie@scut.edu.cn)

It is well known that the dynamic cues caused by head turning and pinna-based spectral cue are vital to sound source localization, especially for front-back and vertical localization. A series of localization experiment is carried out via virtual auditory display to explore the relative contribution of dynamic and spectral cues to localization. Virtual sources at different intended spatial locations are recreated using various combinations with noise stimuli at full audio or 4 kHz-lowpass frequency ranges, individualized HRTFs or non-individual HRTFs derived from KEMAR and spherical-head model, and static or dynamic binaural synthesis. Furthermore, statistical analyses are performed on localization performance in terms of the percentage of front-back and up-down confusion as well as the mean angle error in virtual source localization. Primary results indicate that both dynamic and spectral cues contribute to front-back and vertical localization; whereas dynamic cue contributes more to front-back localization, while individualized spectral cue at high frequency above 4 kHz contributes more to vertical localization. Dynamic and high-frequency individualized spectral cues are also helpful to reduce the angle error in localization. [Work supported by the National Natural Science Foundation of China,11174087, 50938003, and State Key Lab of Subtropical Building Science, South China University of Technology.]

9:45

3aPP6. Approximately calculate individual near-field head-related transfer function using an ellipsoidal head and pinnae model. Yuanqing Rui, Guangzheng Yu, and Bosun Xie (Physics Department, School of Science, South China University of Technology, Wushan Rd. 381#, Tianhe District, Guangzhou, Guangdong 510640, China, scgzyu@scut.edu.cn)

Head-related transfer functions (HRTFs) describe the acoustic transmission from a point source to ears and are individual-dependent. Measurement is an effective way to obtain individual HRTFs, but the workload is very heavy. This is particularly difficult in the near-field HRTFs measurement due to their source distance-dependence within 1.0 m. Numerical calculation is another way to obtain HRTFs. To reduce the calculation load, an ellipsoid head and pinnae (HAP) model is proposed in the present work for approximately calculating the individual near-field HRTFs via the fast multipole boundary method (FMBEM). The dimension of ellipsoid head is defined by head width, head height, head depth, which is selected to approximately fit the individual interaural time difference (ITD) and interaural level difference (ILD) below about 5 kHz. The individual pinna geometry obtained via a 3D laser scanner provides individual HRTF spectral cues above 5 kHz. To validate the proposed method, the HAP of KEMAR and human is constructed, and calculating results are compared with the measurements. Psychoacoustic experiment is also carried out to evaluate the model and results. [Work supported by the National Natural Science Foundation of China for Young Scholars (No 11104082) and the Natural Science Foundation of Guangdong Province.

10:00–10:15 Break

10:15

3aPP7. Prior probabilities tune attentional bandwidth. Michael Wolmetz and Mounya Elhilali (Center for Language and Speech Processing, Johns Hopkins University, 3400 N Charles St, Barton Hall, Baltimore, MD 21218, mikew@jhu.edu)

Top-down schemas or prior probabilities associated with different auditory objects and their acoustic features are thought to improve auditory scene analysis and auditory object recognition processes. This study focused on whether listeners implicitly track the prior probabilities associated with different tone frequencies, and whether tracking those probabilities modulates attentional bandwidth to improve detection sensitivity. Using the target-probe paradigm, attentional bandwidth for varying levels of probe probability was estimated for 40 listeners. To estimate the attentional bandwidth, probes were presented in a continuous band-pass noise masker at distances of .75, 1.25, 1.75, and 3.75 Equivalent Rectangular Bandwidths from a target frequency of 250 Hz, at varying probabilities of occurrence. All tones were presented at approximately 90% detection when presented alone as measured during a separate thresholding procedure. Results indicate that prior probability plays a role in attentional bandwidth: the attentional band is more broadly tuned when tones at probe frequencies are more likely, consistent with optimal observer predictions. Sensitivity of attentional bandwidth to endogenous (listener-directed) and exogenous (stimulus-driven) attention will be discussed.

10:30

3aPP8. What is a good musical pattern? On acoustic structure and goodness of pattern. Ronaldo Vigo (Psychology, Ohio University, 211 Porter Hall, Athens, OH 45701, vigo@ohio.edu), Yu Zhang (Communication Sciences and Disorders, Ohio University, Athens, OH), and Mikayla Barcus (Psychology, Ohio University, Athens, OH)

An open problem in acoustical psychophysics involves the extent to which the structuro-acoustical properties of a set of sounds determine goodness of pattern judgments (i.e., how good a pattern is perceived to be). We investigated this question experimentally and theoretically from the standpoint of the dimensional structure of sets of sounds defined over the dimensions of timbre, tone, and duration. We observed a distinctive goodness of pattern ordering for the structures tested which involved sets of four tones defined over the three dimensions. The results were consistent with predictions from categorical invariance theory (CIT; Vigo, 2009, 2011a, 2012) which posits that humans detect the dimensional invariants of a set of discrete sounds — a process described in the theory as structural kernel extraction. Using CIT to interpret our results, we also found that sets of stimuli in "structural equilibrium" (the condition when all of the dimensions of a stimulus set play the same structural role in the set) were perceived as conveying better sound patterns. Based on these results, we propose that the tradeoff between complexity and invariance described in CIT can account for differences in music preferences and style.

10:45

3aPP9. Efficient coding of multiple nonorthogonal redundancies between acoustic dimensions in novel complex sounds. Christian Stilp (Department of Psychological and Brain Sciences, University of Louisville, Life Sciences Building, Louisville, KY 40292, christian.stilp@gmail.com) and Keith Kluender (Speech, Language, and Hearing Sciences, Purdue University, West Lafayette, IN)

Stilp and colleagues (*Proc. Natl. Acad. Sci.* [2010]; *JASA* [2011]; *PLoS One* [2012]) provided perceptual evidence for efficient coding of robust covariance between acoustic dimensions in novel complex sounds. Discrimination of sounds that directly violated this redundancy (Orthogonal condition) was initially inferior to that of sounds obeying overall redundancy (Consistent condition). Performance was consistently predicted by principal components analysis (PCA), as experimental conditions aligned with statistical dimensions in PCA. Stilp and colleagues suggested efficient coding may contribute to perceptual organization for speech, but two aspects of their experimental designs qualify this extension: robust evidence for only one statistical regularity between acoustic dimensions was tested while speech possesses many; and, all statistical structure was mutually orthogonal which is often not true of speech sounds. Here, listeners discriminated sounds supporting two concurrent, nonorthogonal regularities (patterns of covariance between acoustic dimensions: attack/decay and spectral shape.) Despite nonorthogonality, these concurrent statistical regularities were efficiently coded, as discrimination of Consistent sound pairs was initially superior to that of Orthogonal sound pairs. Performance did not adhere to predictions made by PCA. Implications for speech perception and auditory 'category' acquisition will be discussed. [Supported by NIDCD.]

11:00

3aPP10. Modeling normal and hearing-impaired monaural and binaural signal detection in the presence of noise. Miriam Furst (Tel Aviv University, Ramat Aviv, Tel Aviv 69978, Israel, mira@eng.tau.ac.il)

Psychoacoustical investigations have demonstrated that monaural and binaural signal detection is affected by existence of distracting noise. The most prominent phenomena are co-modulation masking release (CMR) in monotic listening, and binaural masking level difference (BMLD) in diotic listening. Both CMR and BMLD are significantly deteriorates in hearing-impaired listeners. Both CMR and BMLD phenomena are tested by a complete model of the auditory system for normal and hearing-impaired systems. Prediction of the amplitude discrimination results are obtained by deriving the Cramer Rao lower bound (CRLB) of the neural activity. The auditory system model includes a complete cochlear model with integrated outer hair cells and tectorial membrane; an inner hair cell-synapse model that transduce the cilia motion to auditory nerve instantaneous rate; Inferior Colliculus that receives inputs from both ears and process them by excitatory-inhibitory (EI) cells. The AN activity is considered as a non-homogeneous Poisson process (NHHP). We have recently showed that EI cells are NHHP as well, if their inputs behave as NHHP. Therefore, CRLB can be derived analytically from both AN and IC outputs. We have successfully predicted major CMR and BMLD properties as a function of frequency and noise properties for normal and impaired auditory system.

11:15

3aPP11. Response to a pure tone in a nonlinear frequency-domain model of the cochlea. Julien Meaud and Karl Grosh (Mechanical Engineering, University of Michigan, Ann Arbor, MI 48104, jmeaud@Umich.edu)

The nonlinear response of the cochlea to a pure tone is simulated using a novel computational model. In this physiologically-based finite element model, the three-dimensional intracochlear fluid dynamics are coupled to a micromechanical model of the organ of Corti and to electrical potentials in the cochlear ducts and outer hair cells (OHC). Active feedback due to OHC somatic electromotility is represented by linearized piezoelectric relations and is coupled to the nonlinear hair-bundle mechanoelectrical transduction current. Using an alternating frequency/time method and a single set of parameters, we simulate the compressive nonlinearity, harmonic distortion and DC shifts in the response of the cochlea to a single tone. Model predictions agree well with available experimental data.

11:30

3aPP12. Pole-zero characterization of middle ear acoustic reflectance data. Sarah Robinson, Cac Nguyen, and Jont Allen (Electrical Engineering, University of Illinois at Urbana-Champaign, Urbana, IL 61801, srrobin2@illinois.edu)

Middle ear acoustic reflectance (AR) measurements have valuable clinical applications. AR is measured using a foam-tipped probe sealed in the ear canal, containing a microphone and receiver (i.e. MEPA3 system, Mimosa Acoustics). From the complex pressure response to a broadband stimulus, the acoustic impedance and reflectance of the middle ear can be calculated as functions of frequency. A sizeable pool of normal and pathological AR data, collected by various researchers, indicates that many pathological ears have an AR that systematically differs from normative data. Assessment of such data typically relies on consideration of the magnitude AR, or separate consideration of AR magnitude and phase. By fitting poles and zeros to AR data, we have achieved a compact and accurate representation of the complex data ($<$5% RMS relative error). It was found that removing an approximated ear canal phase delay from AR data before fitting allowed for better comparison across ears, and better comparison with existing network models of the middle ear. Pole-zero fits indicated isolated regions of individual variation for normal ears, and showed differences between normal and pathological ears. Pole-zero fitting shows promise for more quantitative, robust diagnosis of middle ear pathology than AR magnitude alone.

WEDNESDAY MORNING, 24 OCTOBER 2012 — LIDO, 10:15 A.M. TO 12:00 NOON

Session 3aSA

Structural Acoustics and Vibration: Structural Acoustics Optimization

Micah R. Shepherd, Chair
Applied Research Lab, Penn State University, State College, PA 16801

Invited Papers

10:15

3aSA1. Structural acoustic optimization of ribbed panels excited by complex forcing functions. Micah R. Shepherd and Stephen A. Hambric (Applied Research Lab, Penn State University, PO Box 30, State College, PA 16801, mrs30@psu.edu)

Structural acoustic optimization is performed on a ribbed panel exited by diffuse acoustic field and turbulent boundary layer (TBL) flow. In order to vary the rib location during the optimization, component mode synthesis (CMS) was used with the rib and plate treated as separate substructures. The CMS approach couples the individual modes of the rib and plate using impedances at the connection points thus only requiring full eigenanalysis and radiation resistance calculations once and allowing for more rapid function evaluation during the optimization. The optimization was then performed using an evolutionary strategy with covariance matrix adaption. The rib location and properties were varied by the optimizer to find the best low-noise panel. An exhaustive search was performed to verify the optimum solution and compare results for several objective functions. Alternative design variables and constraints will also be discussed.

10:35

3aSA2. Sound reduction from vibrating thin plates using dimpling and beading design. Kyle R. Myers (Mechanical and Aeronautical Engineering, Western Michigan University, Kalamazoo, MI), Nabeel T. Alshabatat (Mechanical Engineering, Tafila Technical University, Tafila, Jordan), and Koorosh Naghshineh (Mechanical and Aeronautical Engineering, Western Michigan University, 1903 W. Michigan Ave., M/S 5343, Kalamazoo, MI 49008, koorosh.naghshineh@wmich.edu)

This study presents a design method to minimize the radiated sound power from vibrating beams and plates. The method relies on altering the acoustic characteristics of beam and plate structures passively by forming dimples and beads on their surfaces. The dimples and beads change the local stiffness of the structure without changing its mass. Also, they alter the mode shapes so as to reduce sound

power. The vibration response of the dimpled and beaded beams and plates are calculated using the finite element method (i.e., ANSYS parametric design language). Then, the radiated sound power from vibrating structure is calculated based on the Lumped Parameter Model (LPM). Finally, the method of Genetic Algorithm (GA) is used to optimally locate and size the dimples or the beads on the plate to minimize the sound radiation. The sound radiation is minimized either at a single frequency, or over a broad frequency band. The results show that dimples or beads forming on thin beams and plates can achieve effective reductions in radiated sound power.

10:55

3aSA3. Using optimization for acoustic cloak design. Liang-Wu Cai and Chunyan Bao (Mechanical and Nuclear Engineering, Kansas State University, Manhattan, KS 66506, cai@ksu.edu)

Many of the acoustic cloak designs are based on the acoustic analogy of transformation optics. This process dictates some major challenges when realizing such cloaks: the materials are very otherworldly. One of the most significant obstacles, the mass-anisotropy, has been removed by using two isotropic acoustic materials to create an equivalent anisotropy. In this presentation, optimization is used as a tool for designing acoustic cloaks such that materials can be feasibly fabricated using naturally occurring materials. The initial designs are based on Cummer-Schurig prescription for acoustic cloak, rendered in a layered structure, and using the anisotropic-isotropic equivalency. The first is using unconstrained optimization to demonstrate that material singularity is not a requirement for perfect cloaking. The optimization is able to fine-tune material properties in the initial design to achieve perfect cloaking within a limited frequency range. The second work is using multi-objective optimization to expand the range of the frequency range in which the cloaking remains effective. The third are is to use constrained optimization to limit the material properties to ranges that are naturally available. Lastly, different optimization techniques are combined to design acoustic cloaks that are made of mixtures of conventional isotropic solid and fluid (acoustic) materials.

Contributed Papers

11:15

3aSA4. Fast wave source localization with sparse measurements. Anthony Sabelli and Wilkins Aquino (Duke University, Durham, NC 27705, ajsabelli@gmail.com)

Source localization problems are encountered in a variety of engineering dis- ciplines. Applications include earthquake localization, damage identification, speaker localization and structural testing. In most realistic settings, measure- ment points are sparse with respect to the physical domain. Moreover, the experimenter may not have control over where to place measurement points. It is in these imperfect settings that we still need to estimate the location of a wave source. In this talk we will outline a method for source localization inspired by the topological derivative used in shape identification. We will draw parallels to phase conjugation mirror techniques and gradient optimization. We will make no assumptions about the nature of the ambient media nor the computational domain. Specifically we allow for energy loss. We will also outline implemen- tation within an existing massively parallel finite element solver. Our proposed algorithm is minimally invasive and fully exploits the underlying optimization in the existing solvers. Moreover we can extend the method to other physical contexts.

11:30

3aSA5. Inverse acoustic source identification in a massively parallel finite element framework. Timothy Walsh (Computational Solid Mechanics and Structural Dynamics, Sandia National Laboratories, PO Box 5800, MS 0380, Albuquerque, NM 87185, tfwalsh@sandia.gov), Wilkins Aquino (Civil and Environmental Engineering, Duke University, Durham, NC), Denis Ridzal, and Joe Young (Uncertainty Quantification and Optimization, Sandia National Laboratories, Albuquerque, NM)

Characterizing the frequency spectrums of acoustic sources from measured accelerometer or microphone data is a common inverse problem in engineering acoustics. Applications include acoustic testing of aerospace structures, room acoustics, and underwater acoustics. Typically, accelerometer or microphone pressures are measured, and it is desired to characterize the acoustic sources that produced these measurements. Many of these applications of interest involve large acoustic domains and high frequency ranges, thus making a finite element solution an attractive option for the forward problem. In this talk we will present a partial differential equation (PDE) constrained optimization approach for solving the inverse problem that is based on a coupling between a finite element-based massively parallel structural dynamics code (Sierra-SD) and a massively parallel optimization code (Rapid Optimization Library (ROL)). Gradients and solution iterates are exchanged between the codes during the solution process. The gradients for the optimization solver are computed using the adjoint method, which translates to forward and adjoint Helmholtz solves in the frequency domain. We will present results on several problems of interest. Sandia is a multiprogram engineering and science laboratory operated by Sandia Corporation, a Lockheed Martin Company, for the US Department of Energy's National Nuclear Security Administration. (DE-AC04-94AL85000)

11:45

3aSA6. Sound radiation of double rotating dipole. John Wang and Hongan Xu (Volvo Construction Equipment, 312 Volvo Way, Shippensburg, PA 17257, john.wang@volvo.com)

Mechanical devices are modeled as single and double rotating dipoles in this study. The expressions for sound pressure, intensity and power are derived. Sound pressure and sound power level reduction is investigated. The reduction is quantified. The theory is compared with test data. The applications are discussed.

Session 3aSC

Speech Communication: Speech Production I: Segments and Suprasegmentals (Poster Session)

Jie Zhang, Chair
Linguistics, The University of Kansas, Lawrence, KS 66045

Contributed Papers

All posters will be on display from 8:00 a.m. to 12:00 noon. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 8:00 a.m. to 10:00 a.m. and contributors of even-numbered papers will be at their posters from 10:00 a.m. to 12:00 noon.

3aSC1. Speech production under real-time simulation of cochlear implant acoustic feedback. Elizabeth D. Casserly (Dept. of Linguistics, Indiana University, Memorial Hall Rm 322, Bloomington, IN 47406, casserly@indiana.edu) and David B. Pisoni (Dept. of Psychological & Brain Sciences, Indiana University, Bloomington, IN)

Although previous research on simulation of cochlear implant (CI) processing with normal-hearing listeners relied on offline transformation of prerecorded acoustic signals, the advent of new simulation technology using a Portable Real-Time Vocoder (PRTV) enables subjects to experience CI simulation not only of interlocutors' speech during face-to-face interaction, but also of their own speech acoustics. This paper explores the effects of this novel acoustic feedback manipulation on subjects' speech production. Nine normal-hearing speakers were recorded producing 114 isolated English words and 24 sentences during three time epochs: once under normal conditions, once immediately after being fitted with the PRTV, and again after experiencing a short, natural conversational interaction through the real-time (8-channel, noise-vocoded) CI simulation. Acoustic-phonetic analysis of subjects' speech revealed substantial, segment-specific shifts in vowel acoustics, alteration of sibilant frication frequencies, and evidence of increased cognitive load (e.g. slower speaking rates) during speech production under conditions of vocoded acoustic feedback. Speakers also appeared to alter their articulatory strategies, spending more time on production of consonants relative to vowels, possibly reflecting a flexible exploitation of reliable somatosensory versus aural feedback cues.

3aSC2. Developmental changes in voiceless fricative production: Influence of position in words. Kanae Nishi and Elizabeth C. Graham (Boys Town National Research Hospital, 555 N. 30th Street, Omaha, NE 68131, kanae.nishi@boystown.org)

Children master the production of fricative consonants later than other phonemes [Moeller et al., Ear Hear. 28, 605-627 (2007)]. Even though recognizable fricative categories are present before school age, fine-tuning of acoustic properties may continue throughout early adolescence [Nissen & Fox, J. Acoust. Soc. Am. 118, 2570-2578 (2005)]. Previous acoustic studies on the development of fricative production focused on those in word-initial position only. Even in adults' speech, acoustics of consonants in word-medial or word-final positions vary more compared to those in the word-initial position. The present study hypothesized that adult-like production of fricatives in the word-final position may be achieved later than those in the word-initial position due to the acoustic variability in the adult exemplars children hear. Thirty-six (six each of 4-, 6-, 8-, 10-, 12-year-olds and adults) female native speakers of American English recorded five tokens of 16 consonant-vowel-consonant monosyllabic real words containing voiceless fricative consonants /f θ s ʃ/ in initial or final position in /i/ and /ɑ/ vowel contexts. Each token was analyzed for frication duration, amplitude, and several spectral measures. Results will be discussed in terms of fricative position in word, vowel context, and speaker age. [Work supported by NIDCD R03 DC009334 and P30 DC004662.]

3aSC3. Effects of articulatory planning factors on children's production of plural *-s*. Rachel M. Theodore (University of Connecticut, 850 Bolton Road, Unit #1085, Storrs, CT 06269, rachel.theodore@uconn.edu), Katherine Demuth (Macquarie University, Sydney, NSW, Australia), and Stefanie Shattuck-Hufnagel (Massachusetts Institute of Technology, Cambridge, MA)

Children's early use of grammatical morphemes is notoriously variable. Recent findings indicate that some variability in early productions is systematically related to speech planning factors, suggesting that variability in morpheme production is not solely the consequence of impoverished syntactic representations. For example, research has shown that plural *-s* is produced more reliably in utterance-final compared to utterance-medial position. Here we examined the locus of the positional effect for plural *-s*. Productions of eight plural nouns in utterance-medial and utterance-final position were elicited from three groups of 2-year-olds. Across the groups, we manipulated articulatory difficulty of the medial context such that it consisted of a stop consonant (e.g., dogs bark), a stressed vowel (e.g., dogs eat), or an unstressed vowel (e.g., dogs arrive). Results showed a robust positional effect for the difficult context created by the stop consonant. The positional effect was not observed for the simple articulatory context created by the stressed vowel. However, planning difficulty for the unstressed vowel was sufficiently increased such that the positional effect again emerged in this context. These results suggest that production of grammatical morphemes is influenced by articulatory planning factors, which points to specific constraints for theoretical accounts of language acquisition.

3aSC4. Understanding speech acoustics in an era of extreme cue-integration: Multi-dimensional phonetics reveals individual differences in fricative production. Ariane E. Rhone (Neurosurgery, University of Iowa Hospitals and Clinics, E11 Seashore Hall, Dept of Psychology, Iowa City, IA 52242, ariane-rhone@uiowa.edu), Keith S. Apfelbaum, and Bob McMurray (Psychology, University of Iowa, Iowa City, IA)

Phonological features are indicated by many acoustic cues (Lisker, 1986). Listeners must thus combine multiple cues to recognize speech (Nearey, 1990). A recent survey of cues to English fricatives identified 24 distinct, useful cues (McMurray & Jongman, 2011). This multi-dimensional nature of speech raises methodological challenges: cues do not contribute equally, and multiple cues contribute to the same phonetic features (e.g. voicing). We offer a solution, using McMurray and Jongman's (2011) fricative database as a test. We used a logistic regression to predict the fricatives using measurements of 24 cues. We then sum the product of each cue value and its weight from the regression to determine the strength of the evidence for a given feature/phoneme (e.g. degree of "f-ness" vs. "v-ness") for each token. By computing this for different subsets of cues, we can measure how classes of cues work in tandem. We illustrate this by examining the relative contribution of cues within the frication and those within the vocoid, as well as spectral and temporal cues, to show individual differences in fricative productions. These analyses offer a straightforward approach to conceptualizing speech perception in high multi-dimensional space, while also giving insights on the production of English fricatives.

3aSC5. Linguistic effects on the timing of gestural coordination in Modern Greek consonant clusters. Jonathan C. Yip (Linguistics, University of Michigan, 455 Lorch Hall, 611 Tappan Street, Ann Arbor, MI 48109, jonyip@umich.edu)

Although position in word, order of place of articulation, and manner of articulation have been shown to influence gestural timing in CC clusters in various languages, there is no consensus on whether these timing patterns are due to biomechanical or perceptual-recoverability constraints (or both) on production. This study of Modern Greek speakers' CC productions investigates the effects of within-word position (initial, medial), place order (front-to-back, back-to-front), C1 manner (plosive, fricative), and C2 manner (plosive, fricative, liquid) on articulatory lag between C1 and C2. A perception-oriented account predicts an influence of both C1 and C2 manner on gestural lag when acoustic masking is most likely (i.e. when CC is word-initial and back-to-front), whereas a biomechanical account predicts no such effect. To assess relative degree of gestural lag, ultrasound imaging and lip video are used to track the timing patterns of labial, tongue-tip, and tongue-dorsum constrictions during production. Preliminary data on word-initial CC sequences show clear gestural onset lag and achievement lag in [kt] relative to [ks] and [kl], but no onset or achievement lag in [pt] relative to [ps] and [pl], consistent with the perceptual-recoverability hypothesis.

3aSC6. Prosodic position effects on the statistical relationships between distinctive features and acoustic-phonetic properties of English consonants. Noah H. Silbert (Center for Advanced Study of Language, University of Maryland, 7005 52nd Ave, College Park, MD 20742, nsilbert@umd.edu), Kenneth J. de Jong, Kirsten Regier, and Aaron Albin (Linguistics, Indiana University, Bloomington, IN)

Previous research has shown that distinctive features must interact extensively to account for the location and shape of phonological consonant categories in multidimensional acoustic space (de Jong, et al., 161st ASA Meeting). The current analysis focuses on how syllable position (onset vs. coda) modulates feature interactions in the consonants /p, b, t, d, f, v, s, z/. Statistical model comparisons indicate that models allowing pervasive interactions between features and syllable position fit better than do more restrictive models with few or no interactions. Some interactions between syllable position and features are well-documented, such as vowel duration distinguishing voicing more robustly in coda position than in onset position. Other such interactions are novel. For example, consonant duration can cue both voicing and manner contrasts, with duration differences corresponding more strongly to manner contrasts in onset position and more strongly to voicing contrasts in coda position. Similarly, measures of noise power distinguish coronals from labials in onset position, whereas place and voicing interact in codas. These results contribute to a picture of the acoustic distribution of consonants being not only segment-specific, but also determined substantially by the position of the consonant within a syllable.

3aSC7. Phonetic effects of distance in Burmese. Becky Butler (Cornell University, 203 Morrill Hall, 159 Central Ave., Ithaca, NY 14853, bbt24@cornell.edu)

This study investigates the phonetic effects of distance from phonological boundaries. Burmese, a Sino-Tibetan language, has a set of words of the shape Cə.Cə.(CV), in which the last syllable is footed, may contain any vowel in the inventory, and carries tone; whereas unfooted syllables contain only the vowel [ə] and do not have lexical tone (Green 1995, 2005). Under a purely phonological interpretation, we may expect all unfooted syllables to be identical in terms of duration and vowel quality. However, Chitoran and Hualde (2007) show that for Romance languages, distance from stress - a phonological entity - can cause significant durational differences between pretonic and pre-pretonic vowels. Similarly, the present study finds that although formant values between prefooted and pre-prefooted vowels in Burmese are not significantly distinct for any speakers - suggesting that vowel quality in all unfooted syllables is similar - distance from the tone-bearing syllable causes significant durational differences for three of five speakers ($p < 0.0001$). These results suggest that the data can be explained by neither phonological categorality nor phonetic gradience alone, but that both play a role in speech production and that the importance of each varies across speakers.

3aSC8. Language specificity in the perception of children's productions of /t/ and /k/. Benjamin Munson (Speech-Language-Hearing Sciences, University of Minnesota, 115 Shevlin Hall, 164 Pillsbury Drive SE, Minneapolis, MN 55455, munso005@umn.edu), Kiyoko Yoneyama (Department of English, Daito Bunka University, Tokyo, Kanto, Japan), and Jan Edwards (Communicative Disorders, University of Wisconsin, Madison, WI)

The age and order of acquisition of what are ostensibly the 'same' sounds can differ across languages. These differences relate to a number of factors, including frequency in the ambient language, language-specific phonetic instantiations of sounds, and language-specific parsing of children's emerging productions (Beckman & Edwards, 2010; Edwards & Beckman, 2008; Li et al, 2011). The current investigation examines the role of adults' perception of children's speech on the acquisition of /t/ and /k/ in English- and Japanese-speaking children. Previous work has shown that /t/ is acquired earlier than /k/ in English, but that the opposite is true in Japanese (Nakanishi et al., 1972; Smit et al., 1990). We examined whether this tendency was due to cross-linguistic differences in adults' perception of English- and Japanese-acquiring children's speech. Native speakers of English and Japanese labeled a large set of 2- to 5-year-old children's word-initial /t/- and /k/ productions. Japanese-speaking adults perceived English-speaking children's productions of sounds intermediate between /t/ and /k/ as more /k/-like than did English-speaking adults. This suggests that the earlier acquisition of /k/ in Japanese than in English may be due, in part, to Japanese-speaking adults' willingness to label ambiguous sounds as /k/-like.

3aSC9. Consonant-f_0 interaction under predictable voicing: f_0 lowering due to phonetic factors. Indranil Dutta (Computational Linguistics, English and Foreign Languages University, Department of Computational Linguistics, Tarnaka, Osmania University Campus, Hyderabad 500605, India, indranil.dutta.id@gmail.com), Jasmine M. George, and Minu S. Paul (Language Sciences, English and Foreign Languages University, Hyderabad, Andhra Pradesh, India)

We report on consonant-f_0 interactions in Malayalam. Crosslinguistically, voicing lowers f_0 in the following vowel (House and Fairbanks 1953, Hombert 1978, Clements 2002, Moreton 2006). While this lowering has been attributed to physiological and phonetic factors (Stevens 1998, Atkinson 1978, and Honda 2004), Ohde (1984), Svantesson and House (2006), and Keating (1984) have argued that f_0 lowering serves to maintain a phonological contrast between voiced and voiceless segments. Voicing in Malayalam is predictable; voiceless stops appear initially, and voiced intervocalically. We report on data from 6 native speakers. 3 repetitions of each word in a frame sentence were recorded. Since Malayalam words only appear with voiceless initial stops, we contrast these with nasals from corresponding places of articulation. We examine time-normalized f_0 perturbation due to voicing in Malayalam for all places of articulation for both initial and medial contexts. Our findings lend support to the physiological and phonetic account of f_0 lowering. Malayalam exhibits a pattern of f_0 lowering following voicing and raising during the production of voiceless segments. In spite of Malayalam voicing being predictable, the f_0 perturbation in the following vowel follows the cross-linguistic pattern of lowering following voiced segments. This finding dove-tails with the results from earlier studies that take physiological factors to be responsible for the lowering of f_0 following voiced segments.

3aSC10. Acoustic correlates of breathy voice in Marathi sonorants. Kelly Berkson (Linguistics, University of Kansas, Lawrence, KS 66049, keberkson@gmail.com)

Breathy voiced sonorants occur in fewer than 1% of the languages indexed in the UCLA Phonological Segment Inventory Database. Acoustic analysis of these sounds remains sparse, and our understanding of the acoustic correlates of breathy voice in sonorants is incomplete. The current study presents data from Marathi, an Indo-Aryan language which boasts a number of breathy voiced sonorants. Ten native speakers (five male, five female) were recorded producing Marathi words embedded in a carrier sentence. Tokens included plain and breathy voiced nasals, laterals, rhotics, and approximants before the vowel [a]. Measures reported for consonants and subsequent vowels include duration, F0, Cepstral Peak Prominence (CPP), and corrected H1-H2*, H1-A1*, H1-A2*, and H1-A3* values. As expected,

breathy sounds have lower CPP values than modal sounds, and larger positive values for the remaining spectral measures. The spectral effect of breathiness extends from the beginning of the consonant through the end of the vowel. While some breathy voiced sounds contain a salient breathy interval that is highly visible in the waveform and spectrogram, others don't, and in its absence the spectral differences between breathy and modal sounds are greatly increased.

3aSC11. Acoustic and aerodynamic characteristics of nasal diphthongs in Brazilian Portuguese. Rita Demasi (Linguística, USP, Rua Conde de Itu, 804, São Paulo, São Paulo 04741-001, Brazil, ritademasi@gmail.com) and Didier Demolin (GIPSA-LAB, Université Stendhal, Grenoble, Rhones-Alpes/Grenoble, France)

Previous studies describe acoustic and aerodynamic aspects of nasalized vowels in Brazilian Portuguese. However, there are few studies characterizing nasal diphthongs in this language. The aim of this work is to analyze acoustic and aerodynamic characteristics of nasal diphthongs of Brazilian Portuguese spoken in the city of São Paulo. We compared oral and nasal diphthongs to identify the main features of these sounds and to understand the timing of velum movements. Our data was recorded with the Portable EVA 2 workstation. The corpus of this experiment was made of ten oral and ten nasal diphthongs, with back and front glides at the end: /aw/ and /ej/; /ãw̃/ and /ẽj̃/. Words were inserted in the following sentence [ʤi.gʊ___kadɐ ʤiɐ] and [ʤi.gʊ___todʊ ʤiɐ]. The first phrase was repeated three times with six subjects and the second with three subjects. The corpus was analyzed with the Signal Explorer and Phonédit Softwares. The aerodynamic parameters analyzed were duration, peak and volume of nasal airflow. The acoustic parameters analyzed were formants patterns and FFT spectrum. Aerodynamic and acoustic data show that in nasalized diphthongs, the nasalized vowel is followed by a nasal glide and a short nasal appendix. The peak of nasal airflow at the end of the nasal diphthong is due to the fact that the oral closure, which is made for the following voiceless stop, is produced with an open velum.

3aSC12. Voicing, aspiration, and vowel duration in Hindi. Karthik Durvasula and Qian Luo (Michigan State University, East Lansing, MI 48824, durvasul@msu.edu)

There is extensive evidence that consonantal laryngeal features modulate adjacent vowel duration (Chen 1970). However, it is not clear if both consonant voicing and aspiration affect vowel duration. Previous studies (on Hindi) produced inconsistent results with respect to the effect of consonant aspiration on vowel duration, while finding a clear positive correlation with consonant voicing (Maddieson & Gandour 1976; Ohala & Ohala 1992; Lampp & Reklis 2004). We conducted an experiment on 7 native standard Hindi speakers, who produced 10 repetitions of 12 nonsense words ending in [d̪, d̪ʰ, t̪, t̪ʰ] that had 3 different CVCVV contexts. The results of the experiment show that there is a statistically significant main effect of coda voicing on vowel duration and a marginally significant main effect of aspiration on vowel duration. Furthermore, the effect of the aspirated coda consonants on vowel duration appears to be modulated by the surrounding segmental context. The results suggest that both consonant voicing and aspiration increase adjacent vowel duration. The results also suggest that the inconsistent findings in previous studies with respect to the effect of aspiration could be the result of differing phonetic contexts of the relevant segments.

3aSC13. Hierarchical Bayesian modeling of vowel formant data: Speaker-intrinsic and speaker-extrinsic approaches compared. Aaron L. Albin and Wil A. Rankinen (Department of Linguistics, Indiana University, Memorial Hall, 1021 E 3rd St, Bloomington, IN 47405-7005, aaalbin@indiana.edu)

Vowel formant data is traditionally normalized across speakers by transforming a set of 'raw' measurements into 'standardized' ones in one of two ways. With a *speaker-extrinsic* method, data from each individual is normalized with respect to external baseline measures calculated across the population of all speakers in a corpus, whereas a *speaker-intrinsic* method normalizes entirely with respect to speaker-dependent variables. The present study reports on implementations of both these methods in terms of hierarchical statistical models whereby probability distributions for various model parameters can be obtained using Bayesian analysis (rather than merely 'converting' the measurements). In this new framework, a speaker-extrinsic approach can estimate (1) the size and shape of each speaker's vowel space, (2) the locations of vowel categories across a speech community within a normalized space, and (3) individual speakers' deviations from the community norms. However, this process relies on a number of assumptions that are not needed with a speaker-intrinsic approach, which instead makes many low-level discrete 'decisions' on a speaker-by-speaker basis. By testing multiple models on the same dataset (a large corpus of vowel data collected from 132 speakers of American English), the present study explores the comparative merits of speaker-extrinsic and speaker-intrinsic Bayesian models.

3aSC14. The effect of physical appearance and accent on the elicitation of vowel hyperarticulation by British English native speakers in speech to foreigners. Jayanthiny Kangatharan (School of Social Sciences, Brunel University, 36 Abbots Drive, London, Harrow HA2 0RE, United Kingdom, hspgjjk@brunel.ac.uk), Maria Uther (School of Psychology, Univ. of New South Wales, London, Uxbridge, United Kingdom), and Fernand Gobet (School of Social Sciences, Brunel University, London, Uxbridge, United Kingdom)

Speech aimed at infants and foreigners has been reported to include the physical exaggeration of vowels, that is vowel hyperarticulation. Although infants have been demonstrated to experience hyperarticulated vowels in speech directed at them, little research has been done on whether vowel hyperarticulation occurs as a result of foreign appearance, foreign accent or as a consequence of both looking and sounding foreign. The present study explored if appearance and speech separately affect the native speakers' hyperarticulation. Fifty-two White British adult speakers communicated with one of four different confederate groups (2 types of appearance x 2 types of accent) to solve three modified versions of the DiapixUK tasks. Results indicate that not appearance but speech had an effect on native speakers' production of vowels. Specifically, vowel space was significantly larger in speech directed to foreign-accented individuals than to individuals with native accent irrespective of their physical appearance. The acquired samples of hyperarticulatory speech will be used in perceptual identification and clarity tasks to ascertain which speech samples help native speakers to understand speech better.

3aSC15. Dialectal and age-related acoustic variation in vowels in spontaneous speech. Ewa Jacewicz and Robert A. Fox (Speech and Hearing Science, The Ohio State University, 110 Pressey Hall, 1070 Carmack Rd., Columbus, OH 43210-1002, jacewicz.1@osu.edu)

Our knowledge of the acoustic characteristics of vowels is based mainly on productions obtained under fine experimental control (e.g., [hVd] tokens produced in isolation). Analyzing spontaneous speech data has its obvious challenges because the vowels of interest occur in various segmental and prosodic contexts and are additionally affected by sudden changes in speech tempo. These variations may compromise the accuracy of interpretation of linguistic phenomena such as sound change. In this study we examine if more natural productions of vowels in two dialects (as compared to those produced in citation form and read speech) show evidence of the existence of cross-generational vowel changes—including corresponding changes in spectral dynamics. A subset of vowels from a large corpus of spontaneous conversations was analyzed. The vowels occurred in variable consonantal contexts in both mono- and polysyllabic words. The obtained patterns of vowel change were consistent with those in read and citation form speech. Although the measured spectral changes were smaller due to shorter vowel durations, the dialect-specific nature of formant dynamics was maintained. These results demonstrate that the patterns of vowel change, including variation in formant dynamics, do not diminish under the circumstances of greater contextual/prosodic variability found in spontaneous speech.

3aSC16. Cross-dialectal and cross-generational changes in point-vowel locations in English. Robert A. Fox and Ewa Jacewicz (Speech and Hearing Science, The Ohio State University, 110 Pressey Hall, 1070 Carmack Rd., Columbus, OH 43210-1002, fox.2@osu.edu)

The positions of individual vowels in the acoustic vowel space often change over time in languages such as English. This study examines the changes in the location of four point vowels [i u ɑ æ] for three English dialects (North, Midland and Inland South) as produced by three generations of speakers (children,

adults aged 35-50 and 65+ years). We determined the speaker-specific spectral centroids of the midpoints of a set of 12 monophthongs in the F1 x F2 vowel space produced by each individual speaker. These formant values were then normalized using Nearey's formula. For both the raw and normalized F1 x F2 spaces, polar coordinates for productions of each point vowel were calculated with this centroid as the origin (producing a radius and an angle—a variation of Chung et al.'s [2012]) along with vowel space areas. Compass plots were drawn for the vowels of each speaker and angle (polar) histograms for each vowel were created for each speaker group. Analysis of the histograms and radii showed significant differences among the groups in terms of the location of /æ/ and /u/ as well as area differences. The implications these data have for sound change in progress will be discussed.

3aSC17. Relationship between articulatory acoustic vowel space and articulatory kinematic vowel space. Jimin Lee and Susan Shaiman (Department of Communication Science and Disorders, University of Pittsburgh, Pittsburgh, PA 15206, jiminlee@pitt.edu)

The current study examines the relationship between articulatory acoustic vowel space and kinematic vowel space with an emphasis on the range of tongue movement by utilizing electromagnetic articulography. Subject population is 20 healthy female speakers. Electromagnetic articulography (AG-200) and a synchronized separate digital audio recording system were utilized to obtain kinematic and high quality acoustic data. Three coils on the tongue (tip, body, and dorsum) and one coil on the lower lip were used. To examine both intra- and inter- speaker relationship between articulatory acoustic and kinematic spaces, speech samples of ten different tasks that elicited various articulatory space sizes were collected. Each speaker produced three repetitions of four corner vowels in /h/-vowel-/d/ and /d/-vowel-/d/ consonant environments embedded in a carrier phrase in five different speaking styles (habitual, fast, slow, loud, and soft). Articulatory working space was generated from coordinates of first and second formant frequencies from acoustic data, and position X and Y from each coil kinematic data. Both acoustic and kinematic coordinate data are obtained at the same time sampling point. Results will be discussed in terms of amount of variance of acoustic vowel space explained by kinematic articulatory space and issues of interpretation of acoustic vowel space.

3aSC18. Predictability effects on vowel realization in spontaneous speech. Michael McAuliffe and Molly Babel (Linguistics, University of British Columbia, Totem Field Studios, 2613 West Mall, Vancouver, BC V6T 1Z4, Canada, mcauliff@interchange.ubc.ca)

Previous research on vowel realizations within the formant space has found effects for lexical factors such as word frequency in both laboratory settings (Wright, 2004; Munson & Solomon, 2004; and others) and in spontaneous speech (Gahl, Yao, & Johnson, 2012). In addition to lexical factors, semantic context has also been found to influence vowel realizations in laboratory settings, such as emphatic/non-emphatic contexts (Fox & Jacewicz, 2009) and whether a word is predictable from the preceding words (Clopper & Pierrehumbert, 2008). The current project looks at whether effects on vowel realization for semantic context from the laboratory can be extended to spontaneous speech. As in Gahl, Yao, and Johnson (2012), the Buckeye Corpus (Pitt et al., 2007) will be used, with the same predictors used there with the addition of a semantic predictability measure. Semantic predictability for a given word will be calculated based on relatedness of that word to words five seconds before the word or less, where relatedness will be calculated based on WordNet (Princeton University, accessed 2012). As a listener can rely more on context for disambiguation, words that are predictable from their preceding context are hypothesized to contain less distinct vowels than words that are not predictable from context.

3aSC19. Acoustic correlates of vowel intelligibility in clear and conversational speech for young normal-hearing and elderly hearing-impaired listeners. Sarah H. Ferguson (Communication Sciences and Disorders, University of Utah, 390 South 1530 East, Room 1201, Salt Lake City, UT 84112, sarah.ferguson@hsc.utah.edu) and Hugo Quene (Utrecht Institute of Linguistics OTS, Utrecht University, Utrecht, Netherlands)

Previous reports on the relationship between clear speech acoustic changes and the clear speech intelligibility benefit for vowels have used an "extreme groups" design, comparing talkers who produced a large clear speech benefit to talkers who produced little or no clear speech benefit. In Ferguson and Kewley-Port (2007), 12 talkers from the Ferguson Clear Speech Database (Ferguson, 2004) were assigned to groups based on the vowel identification performance of young normal-hearing listeners, while Ferguson (2010) chose 20 talkers based on the performance of elderly hearing-impaired listeners. The present investigation is employing mixed-effects models to examine relationships among acoustic and perceptual data obtained for vowels produced by all 41 talkers of the Ferguson database. Acoustic data for the 1640 vowel tokens (41 talkers X 10 vowels X 2 tokens X two speaking styles) include vowel duration, vowel space, and several different measures of dynamic formant movement. Perceptual data consist of vowel intelligibility in noise as reflected by the performance of young normal-hearing and elderly hearing-impaired listeners. Analyses will explore the relative importance of the various clear speech acoustic changes to the clear speech vowel intelligibility effect as well as the degree to which this relationship varies between the two listener groups.

3aSC20. Acoustic and physiologic measures of register transitions sung by females. Richard J. Morris (Communication Science and Disorders, Florida State University, 201 West Bloxham Road, 612 Warren Building, Tallahassee, FL 32306-1200, richard.morris@cci.fsu.edu), David Okerlund (College of Music, Florida State University, Tallahassee, FL), and Claire E. Dolly (Communication Science and Disorders, Florida State University, Tallahassee, FL)

The purpose of this study was to examine the physiological and acoustical adjustments made by trained female singers to transition across vocal registers when singing a chord triad. Ten female singers participated in this study. A microphone was placed 30 cm from the corner of the subjects mouth and EGG electrodes were positioned over their thyroid laminae. Audio and EGG signals were channeled to Voce Vista Pro software. Each singer was presented the starting pitch of A3 or 220 Hz, by the experimenter and directed to sing a chord triad using an [a:] vowel. Each triad was sung in a single breath at an andante tempo. The EGG measurements included the closing quotient (CQ) and the FFT spectrum measurements include the harmonic number, frequency, and amplitude of the harmonic with the greatest amplitude. Four singers lowered CQ, two raised CQ, and four maintained a steady CQ when changing vocal register. The frequencies and amplitudes of the peak harmonic remained fairly steady across the register shift. These changes indicated that the singers were tracking their vocal tract resonance to their CQ. The more trained singers used the third harmonic in their chest register and the second harmonic in middle register.

3aSC21. An acoustic analysis of lexical stress in Uyghur. Mahire Yakup (Linguistics, University of Kansas, Lawrence, KS 66044, mylyakup@ku.edu)

In this study, the accent pattern in Uyghur, a Turkic language, was investigated. Two experiments provide a detailed phonetic analysis in order to determine the acoustic cues to stress in Uyghur. In Experiment 1, six disyllabic minimal pairs (e.g., A-cha, a-CHA), contrasting in location of stress, were produced by five native Uyghur speakers with three repetitions in a fixed sentence context. In order to generalize the results from the small set of minimal pairs in the first experiment, Experiment 2 examined the initial syllable of disyllabic nouns that contrasted in first-syllable stress (e.g., DA-ka, da-LA) while syllabic structure (CV versus CVC) was also manipulated. In both experiments, average fundamental frequency, duration, and average intensity were collected in the vowels in accented and unaccented syllables. The results from both experiments showed that there were significant differences in duration and intensity between stressed and unstressed syllables, with the intensity differences moderated by syllable structure. No differences were found in fundamental frequency. While previous studies have classified Uyghur as a pitch-accent and a stress-accent language, the present acoustic data suggest that native speakers make no use of pitch cues to signal stress in Uyghur.

3aSC22. The effect of segmental makeup on Mandarin tone coarticulation. Yuwen Lai and Hui-Ting Huang (Foreign Languages and Literatures, National Chiao-Tung University, Taiwan, 1001 University Road, Hsinchu 30010, Taiwan, yuwen.lai@gmail.com)

The effect of segmental make-ups on tonal coarticulation was investigated in Mandarin. Target syllables differ in coda (no coda, alveolar nasal or velar nasal) and initial sonorancy (obstruent or sonorant) were embedded in trisyllabic non-words templates, in the form of trisyllabic names. Twenty native speakers were recorded and the F0 contours of all syllables were measured at a 10-ms time step. The coarticulation effect of four Mandarin tones are examined in three different positions with 4 (initial position), 16 (medial position), and 4 (final position) tonal contexts. Moreover, the phonetic variations of each tone in different prosodic positions were examined. The preliminary results indicate that the coarticulation effect is less prominent when intervened by obstruents whereas nasal codas amplify the effect. The realization of this modulating effect on the directionality of tonal coarticulation (carryover and anticipatory) will be discussed. The magnitude of coarticulation in compatible (adjacent tones have similar registers) and conflicting (adjacent tones with large register difference) environments will also be compared.

3aSC23. Acoustic differences in adult-directed and child-directed monosyllabic Mandarin tone productions. Puisan Wong, Xin Yu, Guanjin Zhang, Jiulin Zhu, and Tina Yeung (Otolaryngology–Head and Neck Surgery, The Ohio State University, 915 Olentangy River Road, Columbus, OH 43212, pswResearch@gmail.com)

To investigate the acoustic differences in adult- and child-directed monosyllabic Mandarin tone productions, twenty-one mothers of preschool children labeled pictures that represented monosyllabic words to their children and to another adult. Their productions were low-pass filtered to eliminate lexical information. Five judges determined the target tones based on the filtered stimuli. Acoustic analyses were performed on the productions in which the target tones were correctly identified by four or more of the judges. Preliminary results showed no duration difference in the four tones in adult-directed and child-directed productions. Overall, all the four tones were produced at significantly higher f0s in child-directed productions than in adult-directed productions. Specifically, child-directed Tone 1 productions were produced at significantly higher f0s, and the f0 contours exhibited higher positive f0 slopes and were not as level as in adult-directed Tone 1 productions. Child-directed Tone 2 productions were produced at higher f0s, spanned at larger f0 ranges but maintained the same rising slopes as in adult-directed productions. Child-directed Tone 3 and Tone 4 productions had the same f0 shapes as in adult-directed productions but were produced at higher f0s (Work supported by NIDCD).

3aSC24. A production and perception study on tonal neutralization in Nanchang Chinese. Jiang Liu and Jie Zhang (Linguistics, The University of Kansas, 1541 Lilac Lane, Lawrence, KS 66044, liujiang@ku.edu)

In a production study of tonal contrasts in lexically stressed but grammatically stressless syllables vs. lexically stressless syllables in Nanchang, a Gan dialect spoken in southeastern China, we found that tonal neutralization only occurs in lexically stressless syllables. We argue that the main phonetic ground for such a tonal contrast distribution lies in the rhyme duration difference between syllables with and without lexical stress, namely, lexically stressless syllables have shorter rhyme duration than lexically stressed but grammatically stressless syllables, and the shorter the rhyme duration of a syllable is the fewer tonal contrasts the syllable allows. In terms of perception, we found that different tonal contrasts indeed become neutralized in lexically stressless syllables. However, the neutralization pattern at the perception level is not the same as the one at the production level due to word specific effects.

3aSC25. Perception and production of Mandarin tones by English vocalists and instrumentalists. Shuang Lu, Joe Kirkham, Ratree Wayland, and Edith Kaan (Department of Linguistics, University of Florida, P.O. Box 115454, Gainesville, FL 32611-5454, shuanglu@ufl.edu)

Musical training has been found to positively affect non-native lexical tone perception and production (e.g. Wong et al., 2007). While there are comprehensive studies on trained instrumental musicians, relatively little is known about formally trained *vocal* musicians. The present study compares English vocalists and instrumentalists to see which type of musical training is more advantageous to lexical tone perception and production. Stimuli consisted of six syllables ([t^{h}i], [li], [mi], [t^{h}o], [lo], [mo]) associated with four Mandarin tones (high-level, high-rising, low dipping, and high falling). Native Mandarin non-musicians, native English non-musicians, vocalists, and instrumentalists (n = 15 per group) were tested in a same/different discrimination task and an imitation task. In the discrimination task, the English vocalists [d' = 3.12] performed significantly better than the English non-musicians [d' = 2.41; p = 0.04]. The vocalists also numerically outperformed the instrumentalists [d' = 2.84], who in turn outperformed the English non-musicians. Analyses on the "different" and "same" tone-pairs showed that the vocalists are marginally more accurate than the instrumentalists for T2-T2 pair-type only [p = 0.067]. In the imitation task, the three English groups did not differ in how their productions were evaluated by the native Mandarin judges.

3aSC26. Pitch and intensity in the speech of Japanese speakers' of English: Comparison with L1 speakers. Jun Okada, Ian L. Wilson, and Miyuki Yoshizawa (CLR Phonetics Lab, University of Aizu, Tsuruga, Ikki-machi, Aizuwakamatsu, Fukushima 965-8580, Japan, s1170182@u-aizu.ac.jp)

The speech of L2 learners of English is often difficult to understand because of intonation problems and misplaced word stress. In this research, we investigated whether or not the intonation patterns of Japanese speakers of English show common patterns based on proficiency level. First, we recorded "The North Wind and the Sun" from 50 Japanese undergraduate students (aged 18 to 24). We recorded native English speakers and also obtained such native data online. Next, we labeled each word and analyzed the pitch and intensity using Praat. Data was separated by gender and by proficiency in English, results were plotted, and statistical analysis was undertaken. Preliminary results show that pitch (and to a lesser extent, intensity) showed a common pattern across native speakers, but that L2 speakers relied on intensity much more than pitch in the production of stress.

3aSC27. Prosodic characteristics of three sentence types in Thai. Alif Silpachai (Linguistics, University of Southern California, Los Angeles, CA 90089, asilpachai@ucla.edu)

This study presents an acoustic analysis of three sentence types in Thai (declarative, interrogative, and emphatic) with the goal of providing a basic characterization of their prosody. To investigate prosodic realizations of sentence final syllables, we placed, in a sentence-final position, a target word which varied in one of the 5 lexical tones in Thai. We also varied the tonal context before the target word so that the pre-target word ends with low (21), mid (31), or high (45) tones. Preliminary results from one speaker show that F0 measures, especially f0 maximum, minimum, and range, differed across sentence types. In particular, emphatic sentences were distinguished from non-emphatic sentences by expanded F0 range, whereas target words in questions were distinguished from those in declarative sentences by both higher F0 maximum and minimum. Syllable duration also played a role in signaling emphasis and question: emphatic sentences were significantly longer than non-emphatic sentences, and questions were significantly shorter than declarative sentences. Interestingly, the tonal pattern of the target word changed for the case of emphasis when the target word had 31 and 45 tones. We will present findings from four additional Thai speakers and discuss their relevance to the intonational phonology of Thai.

3aSC28. Brazilian Portuguese intonation: A comparison between automatic and perceptual analyses. Waldemar Ferreira Netto, Marcus Vinicius Moreira Martins, Daniel Oliveira Peres, Renata Cezar de Moraes Rosa, and Joice Medeiros (Dept. de Letras Clássicas e Vernáculas, Universidade de São Paulo, Av. Professor Luciano Gualberto, 403, São Paulo 05508-900, Brazil, marcusvmmartins@gmail.com)

The present work aims to determine the most appropriated automatic analysis of phrasal intonation patterns in Brazilian Portuguese, understood as the variation of F0. In order to evaluate the automatic analysis done by the software ExProsodia a perceptual analysis was carried out and its results were compared to the ones given by the software. The corpus consisted of one sentence produced three times by 11 female subjects. In the perceptual analysis an intuitive evaluation was done considering the syllabic nuclei as intonation units (control group). The data from the automatic analysis were analyzed in three different ways: i) the cumulative mean of all points of intonation curve; ii) the cumulative mean of points higher than 0Hz; iii) and the cumulative mean of the points selected by ExProsodia. The acoustic parameters considered for the automatic analysis were: F0- from 50Hz to 700Hz; duration- from 20ms to 600ms; intensity- higher than 10% of RMS mean of intonation curve. A Dunnett's test compared the control group with other groups from

the automatic analysis ($\alpha = 95\%$, $n = 33$). In the comparison between the control group and i & ii, it was obtained $p < 0.05$. It was verified that the units established by the interval given by ExProsodia (iii) were the only ones that showed no significant differences when compared to the control group. The results indicate a similarity between the automatic and perceptual analyses. The automatic analysis done by ExProsodia seems thus trustworthy.

3aSC29. Language grouping based on pitch interval variability. Diana Stojanovic (Linguistics, University of Hawai'i at Manoa, 1811 East-West Rd., 933, Honolulu, HI 96848, dswongkaren@gmail.com)

Language rhythm has been discussed so far as a consequence of regularity, phonological differences, and in most recent literature as a result of durational variability. Original support coming from perception experiments is recently often questioned at least in terms of how well listeners can classify language samples stripped of all but durational information. In particular, pitch has been suggested to play a significant role in perceiving rhythmic differences. In Patel 2006, language and music rhythm differences between English and French were measured by means of durational and melodic measures. It was shown that the variability of pitches did not distinguish between two languages, but the variability of pitch excursions was significant. In this paper, we use pitch excursion variability as a measure applied to read samples of several languages. Preliminary results on 30 sec samples show that Brazilian Portuguese, French, Hawaiian, and Indonesian group together compared to English and German. The first group has more frequent but less prominent excursions (60 Hz) while the second group has less frequent but larger excursions (100Hz). Based on these results, originally proposed regularity can at least partly be explained by the variability of pitch intervals, grouping languages into "smooth" vs. "more prominent".

3aSC30. Discriminating languages with general measures of temporal regularity and spectral variance. Kathy M. Carbonell (Speech, Language & Hearing Sciences, University of Arizona, 1131 E 2nd St, Tucson, AZ 85721, kathyc@email.arizona.edu), Dan Brenner (Linguistics, University of Arizona, Tucson, AZ), and Andrew J. Lotto (Speech, Language & Hearing Sciences, University of Arizona, Tucson, AZ)

There has been a lot of recent interest in distinguishing languages based on their rhythmic differences. A common successful approach involves measures of relative durations and duration variability of vowels and consonants in utterances. Recent studies have shown that more general measures of temporal regularities in the amplitude envelope in separate frequency bands (the Envelope Modulation Spectrum) can reliably discriminate between English and Spanish [Carbonell et al. J. Acoust. Soc. Am. 129, 2680.]. In the current study, these temporal structure measures were supplemented with measures of the mean and variance of spectral energy in octave bands as well as with traditional linguistic measures. Using stepwise discriminant analysis and a set of productions from Japanese, Korean and Mandarin speakers, this suite of both acoustic and linguistic measures were tested together and pitted against each other to determine the most efficient discriminators of language. The results provide insight into what the traditional linguistic measures of speech rhythms are telling us about how language type structures the acoustic signal.

3aSC31. An experimental study of Korean rhythm structure on the basis of rhythm metrics. Eun-Sun Tark (Linguistics, University of Kansas, Lawrence, KS 66045, estark713@gmail.com)

This paper investigates the rhythm structure of Korean. Metrics used in this study included %V, ΔC, Varco V, nPVI-V, and rPVI-C, which have been shown to reflect differences in rhythm structure (stress-, syllable-, and mora-timing) across languages. Ten female native Koreans each produced 20 short declarative Korean sentences. The rhythm metric results of Korean were compared to those of English, Spanish, and Japanese using raw data from previous studies [Ramus et al. 1999, Cognition 73, 265-292; White and Mattys, 2007, Journal of Phonetics 35, 501-522; Grenon and White, 2008, BUCLD 32, 155-166]. Results indicate that Korean combines aspects of both syllable timing and mora timing. Korean has a similar ΔC, Varco V, and nPVI-V to syllable-timed languages. Korean has a similar %V and nPVI-V to mora-timed languages. These data show that instead of belonging to one of three distinct rhythm categories, languages may be placed along a rhythm structure continuum. On such a continuum, Korean is placed between syllable-timed and mora-timed languages, and distinct from stress-timed languages.

3aSC32. Phonetic characteristics of syllable reduction and enhancement in American English. Keiichi Tajima (Department of Psychology, Hosei University, Tokyo, Japan, tajima@hosei.ac.jp) and Stefanie Shattuck-Hufnagel (Speech Communication Group, Research Laboratory of Electronics, Massachusetts Institute of Technology, Cambridge, MA)

Syllables have been argued to play an important role in the prosodic organization of spoken language, but the number of syllables that speakers produce or listeners hear in an utterance is not always clear-cut, and may vary in subtle ways. For example, speakers may reduce or delete unstressed vowels in casual speech, e.g., producing *support* as *s'port*, on one hand, or insert vowels in careful speech, e.g., producing *please* as *puh-lease*, on the other. Relatedly, duration differences in a non-vowel segment may lead to changes in word identity based on the presence/absence of an unstressed vowel, as when different durations of /l/ in *blow* are sufficient to lead listeners to perceive either *blow* or *below*. The present study investigates how often such apparent changes in syllable count occur in spoken English, which phonetic contexts they tend to occur in, and to what extent these changes are probabilistic, by analyzing productions from a phonetically annotated corpus of conversational American English. Preliminary results suggest that unstressed vowels are often reduced enough to be omitted from the transcription, and this occurs more in certain phonetic contexts than others. Further results, and implications for theories of speech production and perception, will be discussed. [Work supported by JSPS.]

3aSC33. Video recordings of L1 and L2 jaw movement: Effect of syllable onset on jaw opening during syllable nucleus. Yusuke Abe, Ian L. Wilson (CLR Phonetics Lab, University of Aizu, Tsuruga, Ikkimachi, Aizuwakamatsu 965-8580, Japan, s1170175@gmail.com), and Donna Erickson (Showa Music University, Kawasaki, Kanagawa, Japan)

Video is a convenient, inexpensive method of recording data for jaw movement during speech. However, when using markers attached to the chin, it is possible that the data will not represent actual mandible motion, because of the skin stretching over the mandible - especially true for labial consonants. In this study, we made video recordings of L1 and L2 speakers of English saying 5 trials of 34 sentences each, and we automatically measured the distance between paper markers attached to the chin and glasses. We compared jaw opening during syllable nucleus for syllables with and without labial onsets, for L1 and L2 English speakers of various proficiencies. Although speakers must stretch the lower lip upwards for a labial constriction, preliminary results show that there are no statistically significant differences for any speaker's jaw opening during the nucleus of non-labial- versus labial-onset syllables. There is also very little intra-subject variation in the metrical structure (as measured by jaw opening) for a given sentence across trials. However, across-trial variability in the time between jaw movement peaks is a lot less for L1 than for L2, presumably because these L2 speakers have not yet mastered the metrical structure of English.

3aSC34. Acoustic contrastivity in soft, conversational, and loud speech. Yunjung Kim and Ali Beslin (Dept. of Comm Sci & Disor, Louisiana State University, Baton Rouge, LA 70803, ykim6@lsu.edu)

Increasing vocal effort is frequently used in clinical practice as a strategy to enhance speech intelligibility of individuals with various speech problems, particularly dysarthria. However, it is not straightforward why this strategy might yield better speech intelligibility, although some potential contributors have been suggested including hyperarticulation of vowels, greater presentation level of sounds (for consonant identification accuracy) or both. This presentation focuses on the change of *relative* contrastivity within utterances that were produced with gradually increasing vocal efforts (soft, conversational and loud, see Kim, ASA, 2011) to examine whether acoustic contrastivity is exaggerated with increasing vocal intensity and to identify the acoustic variables that produce contrastivity that are more or less sensitive to changes in vocal intensity. In this presentation, data on the ratio of vowel durations (long vs short), formant structures of vowels (tense vs lax) as well as the ratio of M1 of /s/ vs /ʃ/ will be compared across soft, conversational and loud speech conditions produced by young female adult speakers.

Session 3aUW

Underwater Acoustics and Signal Processing in Acoustics: Random Matrix Theory

Kathleen E. Wage, Cochair
George Mason University, 4400 University Dr., Fairfax, VA 22030

James C. Preisig, Cochair
WHOI, Woods Hole, MA 02540

Chair's Introduction—8:10

Invited Papers

8:15

3aUW1. Random matrix theory in signal processing: Performance analyses and algorithm design. Christ D. Richmond (MIT Lincoln Laboratory, 244 Wood Street, Lexington, MA 02420, christ@ll.mit.edu)

Estimation of covariance matrices from a finite sample set of data observations plays a central role in signal processing. The true data covariance is rarely known in practice, but optimized algorithms inevitably depend on such statistics to enable robust system performance; for example, environments plagued by dominant interference, and/or those challenged by complex propagation. Classical (finite) random matrix theory (RMT) facilitates assessment of finite sample effects surrounding covariance estimation. Implements shown to be very useful in this regard under a circular complex Gaussian data assumption are reviewed. Recent advances in RMT explore limiting behavior of eigenvalues and eigenvectors of random matrices as dimensions become increasingly large (referred to as infinite RMT). Defining the notion of an empirical distribution for the eigenvalues deviates from classical treatments, but yields amazing convergence toward deterministic distributions. Coupled with the Stieltjes transform, powerful tools emerge that can provide new insights especially for signal processing methods intimately tied to the eigen-decomposition, e.g. diagonal loading and dominant mode rejection used in adaptive beamforming. Although many of the theorems are based on asymptotic convergence as dimensionality increases, they describe performance of finite systems quite well. Aspects of infinite RMT are also reviewed and contrasted with classical RMT.

8:45

3aUW2. Constructing acoustic timefronts using random matrix theory. Katherine C. Hegewisch and Steven Tomsovic (Physics, Washington State University, Deparment of Physics, Pullman, WA 99164-2814, tomsovic@wsu.edu)

In a recent letter [Europhys.Lett. 97, 34002 (2012)], random matrix theory is introduced for long-range acoustic propagation in the ocean. The theory is expressed in terms of unitary propagation matrices that represent the scattering between acoustic modes due to sound speed fluctuations induced by the ocean's internal waves. The scattering exhibits a power-law decay as a function of the differences in mode numbers thereby generating a power-law, banded, random unitary matrix ensemble. This talk describes that approach and extends the methods to the construction of an ensemble of acoustic timefronts. The result is a very efficient method for studying the statistical properties of timefronts at various propagation ranges that agrees well with propagation based on the parabolic equation. It helps identify which information about the ocean environment survives in the timefronts and how to connect features of the data to the surviving environmental information. It also makes direct connections to methods used in other disordered wave guide contexts where the use of random matrix theory has a multi-decade history. [This work was supported by ONR and NSF.]

9:15

3aUW3. Random matrix theory and performance prediction of subspace methods. Raj R. Nadakuditi (University of Michigan, Ann Arbor, MI 48109, rajnrao@umich.edu)

Subspace methods constitute a powerful class of techniques for detection, estimation and classification of signals buried in noise. Recent results in random matrix theory precisely quantify the accuracy of subspaces estimates from finite, noisy data in both the white noise and colored noise setting. This advance facilitates unified performance analysis of signal processing methods that rely on these empirical subspaces. We discuss the pertinent theory and its application to the characterization of the performance of direction-of-arrival estimation, matched subspace detection and subspace clustering for large arrays in the sample-starved setting for both white and colored noise.

9:45

3aUW4. Cross-correlations of diffuse ocean noise using eigenvalue based statistical inference. Ravi Menon, Peter Gerstoft, and William S. Hodgkiss (SIO, 9500 Gilman Dr, UC San Diego, La Jolla, CA 92093, rmenon@ucsd.edu)

Cross-correlations of diffuse noise fields can be used to extract environmental information. The influence of directional sources (usually ships), often results in a bias of the travel time estimates obtained from the cross-correlations. Using an array of sensors, insights from random matrix theory on the behavior of the eigenvalues of the sample covariance matrix (SCM) in an isotropic noise field are used to isolate the diffuse noise component from the directional sources. A sequential hypothesis testing of the eigenvalues of the SCM reveals eigenvalues dominated by loud sources that are statistical outliers for the assumed diffuse noise model. Travel times obtained from cross-correlations using only the diffuse noise component (i.e., by discarding or attenuating the outliers) converge to the expected travel times (i.e., unbiased estimates) and are stable temporally. Data from the Shallow Water 2006 experiment demonstrates the effectiveness of this approach and that the SNR builds up as the square root of time, as predicted by theory.

10:00

3aUW5. Eigenvalues of the sample covariance matrix for a towed array. Peter Gerstoft, Ravishankar Menon, William S. Hodgkiss (SIO Marine Phys Lab, Univ of California San Diego, 9500 Gillman Drive, La Jolla, CA 92093-0238, gerstoft@ucsd.edu), and Christoph Mecklenbrauker (Vienna University of Technology, Vienna, Austria)

Observations of the spatial sample covariance matrix (SCM) reveal that the ordered noise eigenvalues of the SCM decay steadily. Using a stochastic model for the sample covariance matrix, the empirical eigenvalue distribution can be derived using random matrix theory. The eigenvalue spectrum is directly related to the quantile function of the empirical eigenvalue distribution. These ordered eigenvalues have a decay that resembles what is observed in real data. Noise on the array is considered either incoherent self-noise or propagating acoustic noise that is coherent across the array. Using conventional 2D or 3D isotropic noise models, realizations of the SCM eigenvalues are generated using random matrix theory. Deep-water towed-array data are analyzed and it is shown that the eigenvalues of the SCM and compares well with theory.

10:15

3aUW6. Random matrix theory analysis of the dominant mode rejection beamformer. Kathleen E. Wage (Electrical and Computer Engineering Dept., George Mason University, 4400 University Drive, MSN 1G5, Fairfax, VA 22030, kwage@gmu.edu) and John R. Buck (Electrical and Computer Engineering Dept., University of Massachusetts Dartmouth, North Dartmouth, MA)

The Dominant Mode Rejection (DMR) beamformer developed by Abraham and Owsley [Proc. Oceans, 1990] determines the beamformer weights from the sensor covariance matrix eigendecomposition. The weights are designed to reject signals contained in the dominant subspace, which is defined by the eigenvectors associated with the largest eigenvalues. In previous work, we developed a model for the mean notch depth (ND) of the DMR beamformer from random matrix theory (RMT) results on the sample eigenvector fidelity [IEEE Stat. Sig. Proc. workshop, 2012]. While ND is useful, other metrics such as white noise gain (WNG) and signal to interference and noise ratio (SINR) are of great interest. WNG characterizes the beamformer robustness to mismatch, and SINR quantifies overall performance. SINR loss is defined as the ratio of the SINR for a beamformer designed using sample statistics to the SINR for the optimal beamformer designed using ensemble statistics. This talk extends our previous work by considering the relationship among ND, WNG, and SINR for the DMR beamformer. A surprising result obtained from RMT is that for a single loud interferer and twice as many snapshots as sensors, the expected SINR loss depends only on the number of snapshots. [Work supported by ONR.]

10:30

3aUW7. Computational model for the eigenvalue density function of a cylindrically isotropic noise sample covariance matrix. Saurav R. Tuladhar, John R. Buck (ECE Dept, University of Massachusetts Dartmouth, 285 Old Westport Rd, North Dartmouth, MA 02747, stuladhar@umassd.edu), and Kathleen E. Wage (ECE Dept, George Mason University, Fairfax, VA)

Adaptive beamformers (ABFs) rely on the knowledge of ensemble covariance matrix (ECM), which is usually not known a priori. The ECM is often estimated from the sample covariance matrix (SCM). When the sample size is limited, the SCM may not converge to the ECM. ABF performance is then determined by the SCM eigenstructure. Random Matrix Theory (RMT) provides a helpful framework to analyze the asymptotic behavior of the SCM eigenstructure. This talk presents a computational model for the SCM's asymptotic eigenvalue density function (EDF) for a uniform linear array in a cylindrically isotropic noise field. Cylindrically isotropic noise fields are common models for shallow water environments, including the Kuperman-Ingenito noise model. The proposed method employs Nadakuditi and Edelman's polynomial method to model the EDF as the product of a deterministic matrix and a Wishart matrix. The model exploits properties of free multiplicative convolution to reduce the required polynomial order, resulting a substantial computational savings. The model EDF exhibits good agreement with eigenvalue histograms from simulations. A model for the SCM EDF is a necessary first step for accurate bearing dependent detection thresholds in cylindrically isotropic noise environments. [Supported by ONR.]

Meeting of Accredited Standards Committee (ASC) S2 Mechanical Vibration and Shock

A. T. Herfat, Chair ASC S2
Emerson Climate Technologies, Inc., 1675 West Campbell Road, PO Box 669, Sidney, OH 45365-0669

C. F. Gaumond, Vice Chair ASC S2
Naval Research Laboratory, Code 7142, 4555 Overlook Ave. SW, Washington, DC 20375-5320

Accredited Standards Committee S2 on Mechanical Vibration and Shock. Working group chairs will report on the status of various shock and vibration standards currently under development. Consideration will be given to new standards that might be needed over the next few years. Open discussion of committee reports is encouraged.

People interested in attending the meeting of the TAG for ISO/TC 108, Mechanical vibration, shock and condition monitoring, and four of its five subcommittees, take note—that meeting will be held in conjunction with the Standards Plenary meeting at 9:00 a.m. on Tuesday, 23 October 2012.

Scope of S2: Standards, specifications, methods of measurement and test, and terminology in the field of mechanical vibration and shock, and condition monitoring and diagnostics of machines, including the effects of exposure to mechanical vibration and shock on humans, including those aspects which pertain to biological safety, tolerance and comfort.

WEDNESDAY AFTERNOON, 24 OCTOBER 2012 JULIA LEE A/B, 1:00 P.M. TO 3:00 P.M.

Session 3pAB

Animal Bioacoustics: Vocalization, Hearing, and Response in Non-Human Vertebrates II

Michael A. Stocker, Chair
Ocean Conservation Research, P.O. Box 559, Lagunitas, CA 94938

Contributed Papers

1:00

3pAB1. Aerial hearing sensitivity in a southern sea otter (*Enhydra lutris nereis*). Asila Ghoul and Colleen Reichmuth (Institute of Marine Sciences, Long Marine Laboratory, University of California, Santa Cruz, 100 Shaffer Rd., Santa Cruz, CA 95060, asila@ucsc.edu)

The lack of information concerning auditory sensitivity in sea otters has been recognized by biologists and resource managers as a priority research need for this threatened species. Noise-generating human activity in near-shore marine environments occurs as a result of construction, transportation, exploration and recreation. These activities may degrade critical habitat or cause behavioral or auditory effects that are harmful to individuals. As direct measures of hearing are not presently available for sea otters, we obtained psychophysical hearing thresholds from a trained individual. Audiometric testing was conducted with an adult male sea otter using 500 ms frequency-modulated narrow-band sweeps under quiet conditions. Absolute aerial thresholds were collected at eleven frequencies ranging from 0.125 to 45 kHz. The sea otter showed a broad functional range of hearing, extending from 0.250 to ~40 kHz, with best sensitivity between 2 and 16 kHz. The lowest measured threshold was -1 dB re 20 μPa at 8 kHz. The high-frequency hearing data was similar to that of terrestrial carnivores, while hearing thresholds below 1 kHz showed a relative decrease in sensitivity. Measurements of underwater sensitivity in the same sea otter are ongoing, and will inform explorations of amphibious hearing capabilities in marine mammals, as well as provide insight into the effects of anthropogenic noise on this vulnerable species.

1:15

3pAB2. Auditory thresholds in marine vertebrates conform to natural ambient noise levels. Michael A. Stocker (Ocean Conservation Research, P.O. Box 559, Lagunitas, CA 94938, mstocker@OCR.org) and John T. Reuterdahl (Ocean Conservation Research, Mill Valley, CA)

Auditory thresholds are often displayed in a manner that reveals what is commonly called a “U-curve.” But if the threshold curves are displayed on the x axis on a true Log10 scale the profile is shaped differently. For marine mammals the shape is more like a “hockey stick.” If these curves are overlaid on the “Wenz ambient noise spectra curves” there appears to be shape conformance. This makes sense as auditory sensitivity would naturally evolve to exclude ambient environmental noise. This paper evaluates 120 legacy auditory threshold curves from 18 species of marine mammals and 60 threshold curves from 32 species of fish. The auditory threshold curves from the fish do not conform to the Wenz curves. Given that both the auditory thresholds and the Wenz curves were expressed as pressure gradient energy it is possible that the profile of the fish threshold curves express

sound in either the particle velocity, or both particle velocity and pressure gradient energy. This paper extrapolates the particle velocity data from the fish threshold conditions to determine if there is some conformity to ambient noise levels in either or both the particle and pressure gradient realms.

1:30

3pAB3. High-frequency hearing in seals and sea lions and the implications for detection of ultrasonic coded transmitters. Kane A. Cunningham (Department of Ocean Sciences, University of California at Santa Cruz, 100 Shaffer Rd., Santa Cruz, CA 95060, kacunningham413@yahoo.com), Sean A. Hayes (Fisheries Ecology Division, NOAA National Marine Fisheries Service, Southwest Fisheries Science Center, Santa Cruz, CA), Michelle W. Rub (Fish Ecology Division, NOAA National Marine Fisheries Service, Northwest Fisheries Science Center, Seattle, WA), and Colleen Reichmuth (Institute of Marine Sciences, Long Marine Laboratory, University of California, Santa Cruz, CA)

In order to better understand the ability of pinnipeds to detect acoustic signals from ultrasonic coded transmitters (UCTs) commonly used in fisheries research, high-frequency hearing thresholds were obtained from a trained Pacific harbor seal (*Phoca vitulina*) and a trained California sea lion (*Zalophus californianus*). Using a 69 kHz, 500 ms, narrow-band FM sweep stimulus, detection thresholds for the harbor seal and the sea lion were determined to be 106 dB and 112 dB re 1 μPa respectively. While the harbor seal threshold falls within the range of existing data, the sea lion threshold is 33 dB lower than expected based on previous reports. This finding indicates that sea lions may be more sensitive to the output of UCTs than previously thought, and allows for the possibility that acoustically tagged fish may be selectively targeted for predation by sea lions as well as seals. These hearing thresholds, combined with ongoing work on the effect of signal duration on high-frequency hearing, will help estimate the ranges at which certain UCTs can be detected by these species. Detection range estimations, in turn, will allow fisheries researchers to better understand how fish survivorship data obtained using UCTs may be skewed by pinniped predation.

1:45

3pAB4. Animal-borne active acoustic tags: A new paradigm to conduct minimally invasive behavioral response studies? Holger Klinck (Cooperative Institute for Marine Resources Studies, Oregon State University and NOAA Pacific Marine Environmental Laboratory, Hatfield Marine Science Center, 2030 SE Marine Science Drive, Newport, OR 97365, Holger.Klinck@oregonstate.edu), Markus Horning, David K. Mellinger (Oregon State University, Newport, OR), Daniel P. Costa (University of California, Santa Cruz, CA), Selene Fregosi (Oregon State University, Newport, OR), David A. Mann (Loggerhead Instruments, Sarasota, FL), Kenneth Sexton (The Sexton Company, Salem, OR), and Luis Huckstadt (University of California, Santa Cruz, CA)

In 2011 a pilot study was begun to evaluate the potential of animal-borne active acoustic tags for conducting minimally-invasive behavioral response studies on pinnipeds. A basic prototype tag was developed and tested on juvenile northern elephant seals (Mirounga angustirostris) during translocation experiments at Año Nuevo State Park, CA, USA in spring 2012. The principal scientific questions of this pilot study were these: (1) do sounds emitted from an animal-borne low acoustic intensity tag elicit behavioral responses, and (2) are potential animal responses related to signal content (e.g., threatening vs. non-threatening). Although the sample size was small, preliminary results indicate that (1) low-intensity sounds emitted by animal-borne tags elicit distinct behavioral responses, (2) these responses appear related to signal content, and (3) the responses may differ based on depth, bathymetry, and location. The results of the conducted study show the promise of this approach as a minimally-invasive and cost-effective method to investigate animal responses to underwater sounds, as well as a method to develop mitigation strategies. Future efforts would increase the sample size, range of acoustic stimuli, and age/sex classes of tagged seals. [Funding from NOAA/NMFS Ocean Acoustics Program.]

2:00

3pAB5. Tracking calling depths and movements of North Atlantic right whales using multipath localization. Robert D. Valtierra (Mech. Engineering, Boston University, 110 Cummington St., Boston, MA 02215, rvaltier@bu.edu), Sofie M. VanParijs (Northeast Fisheries Science Center, National Oceanic and Atmospheric Administration, Woods Hole, MA), R. G. Holt (Mech. Engineering, Boston University, Boston, MA), and Danielle M. Cholewiak (Northeast Fisheries Science Center, National Oceanic and Atmospheric Administration, Woods Hole, MA)

The track and calling depths of a North Atlantic right whale (NARW) recorded by 10 bottom-mounted Autonomous Acoustic Recording Units (ARUs) in the Stellwagen Bank National Marine Sanctuary was determined using the Direct Reflected Time Difference of Arrival (DRTD) localization method. An autocorrelation technique was used to extract direct-reflected time difference of arrival information from recorded NARW up-calls containing several overlapping multipath signal arrivals. The method's feasibility was tested using data from play back transmissions to localize an acoustic transducer at a known depth and location. The method was then used to track an hour of movements and depths of a single NARW using periodic up-calls for localization purposes.

2:15

3pAB6. Passive acoustic monitoring on the North Atlantic right whale calving grounds. Melissa Soldevilla, Lance Garrison (NOAA-NMFS Southeast Fisheries Science Center, 75 Virginia Beach Dr., Miami, FL 33149, melissa.soldevilla@noaa.gov), and Christopher Clark (Bioacoustics Research Program, Cornell University, Ithica, NY)

Shallow water environments, such as the North Atlantic right whale calving grounds, pose a challenge to cetacean passive acoustic monitoring due to high variability in ambient noise and environmental conditions. In this region of high shipping traffic and increased ship-strike risk, passive acoustic monitoring may reduce right whale ship strikes. This study describes temporal variability in right whale call detections, ambient noise sources, and environmental conditions on the right whale calving grounds during 2009-2010 and 2010-2011. Right whale detections occurred between November 19 and March 11, on up to 25% of days per deployment with increased nocturnal call detections, and increased acoustic presence off Jacksonville, FL during 2010-2011. Shipping noise was most common off Jacksonville, detected in up to 74% of minutes, with a diurnal peak, while tidally-associated broadband impulses were detected in up to 43% of minutes off Savannah GA. Environmental conditions including SST, wind, waves, and tidal height varied on daily and semi-diurnal scales. While sightings were higher in 2009-2010, the fewer sightings in 2010-2011 were more narrowly distributed within the depth range of the acoustic instruments. Passive acoustic monitoring is effective for detecting right whales in this environment, especially at night when they cannot be seen.

2:30

3pAB7. Comparison of the first-year response of beaked and sperm whale populations to the Northern Gulf oil spill based on passive acoustic monitoring. Natalia Sidorovskaia (Physics, Univ. of Louisiana, P.O. Box 44210, Lafayette, LA 70504-4210, nas@louisiana.edu), Azmy Ackleh (Mathematics, Univ. of Louisiana, Lafayette, LA), Christopher Tiemann (Applied Research Laboratories, UT Austin, Austin, TX), Juliette Ioup, and George Ioup (Physics, Univ of New Orleans, New Orleans, LA)

This paper continues a discussion on using passive acoustic methods to study the environmental impact of the recent oil spill in the Northern Gulf of Mexico on resident populations of marine mammals. The Littoral Acoustic Demonstration Center, possessing several broadband acoustic datasets collected near the spill site before and after the event, is in a unique position for monitoring long-term environmental impacts in the vicinity of the incident The pre-spill recordings provide a baseline which, when combined with post-spill measurements, give important indicators of changes in the local populations. Ackleh et al., J. Acoust. Soc. Am. 131, 2306-2314, provide a comparison of 2007 and 2010 measurements showing a decrease in acoustic activity and abundance of sperm whales at the 9-mile distant site, whereas acoustic activity and abundance at the 25-mile distantsite has clearly increased. This may indicate that some sperm whales have relocated farther away from the spill subject to food source availability. This paper reports on applying

3p WED. PM

developed population estimation techniques to monitor beaked whale response that appears to be different from that of sperm whales. Follow-up experiments will be critical for understanding the long-term impact on different species. [Research supported by SPAWAR, NSF, and Greenpeace.]

2:45

3pAB8. Population density of sperm whales in the Bahamas estimated using non-linked sensors. Elizabeth T. Küsel (Northwest Electromagnetics and Acoustics Research Laboratory, Portland State University, 1900 SW 4th Ave., Portland, OR 97201, kusele@alum.rpi.edu), David K. Mellinger (Cooperative Institute for Marine Resources Studies, Oregon State University and NOAA Pacific Marine Environmental Laboratory, Newport, OR), Len Thomas (Centre for Research into Ecological and Environmental Modelling, University of St. Andrews, St. Andrews, Fife, United Kingdom), and Tiago A. Marques (Centre for Research into Ecological and Environmental Modelling, University of St. Andrews, Campo Grande, Lisboa, Portugal)

Estimates are presented of sperm whale click detection probability and sperm whale population density at the U.S. Navy's Atlantic Undersea Test and Evaluation Center (AUTEC) in the Bahamas. The estimation of the probability of detecting whale echolocation clicks at multiple non-linked sensors uses estimates of sperm whale source level distribution, beam pattern of click emission, distribution of whale locations and orientations with respect to the sensors while clicking, acoustic transmission loss from source (whale) to receiver (bottom hydrophone), and noise levels at the receiver. These data are combined in a Monte Carlo model that propagates simulated clicks from whales at various random positions to each receiving hydrophone to estimate the signal-to-noise ratio at the receiver and the detection function, the probability of detecting clicks as a function of distance. The estimated detection function for each receiving hydrophone is then combined with information on the detector's rate of missed calls and false detections as a function of signal-to-noise ratio, average sperm whale click rates, and the actual number of clicks detected in a given period of time in order to estimate population density. Results are compared to multi-sensor cases where detection functions were estimated analytically.

WEDNESDAY AFTERNOON, 24 OCTOBER 2012 — TRUMAN A/B, 1:00 P.M. TO 3:00 P.M.

Session 3pBA

Biomedical Acoustics: Best Student Paper Award Poster Session

Kevin J. Haworth, Chair
University of Cincinnati, Cincinnati, OH 45209

The ASA Technical Committee on Biomedical Acoustics offers a Best Student Paper Award to eligible students who are presenting at the meeting. Each student must defend a poster of her or his work during the student poster session. This defense will be evaluated by a group of judges from the Technical Committee on Biomedical Acoustics. Additionally, each student will give an oral presentation in a regular/special session. Up to three awards will be presented to the students with $500 for first prize, $300 for second prize, and $200 for third prize. The award winners will be announced at the meeting of the Biomedical Acoustics Technical Committee. Below is a list of students competing, with their abstract numbers and titles listed. Full abstracts can be found in the oral sessions associated with the abstract numbers.

All entries will be on display and all authors will be at their posters from 1:00 p.m. to 3:00 p.m.

2aBA6. Modeling acousto-optic sensing of high-intensity focused ultrasound lesion formation. Student author: Matthew T. Adams

2pBA14. Compound manipulation of micro-particles using a single device: Ultrasonic trapping, transporting and rotating. Student author: Kun Jia

3aBA7. Focused, radially-polarized shear wave beams in tissue-like media. Student author: Kyle S. Spratt

3aBA8. Shear wave generation using hybrid beamforming methods. Student author: Alireza Nabavizadeh

3aBA13. Rayleigh wave propagation method for the characterization of viscoelastic properties of biomaterials. Student author: Siavash Kazemirad

4aBA1. Effects of encapsulation damping on frequency dependent subharmonic threshold for contrast microbubbles. Student author: Amit Katiyar

4aBA2. Pulse duration dependence of cavitation emissions and loss of echogenicity from ultrasound contrast agents insonified by Doppler pulses. Student author: Kirthi Radhakrishnan

4aBA3. Echogenicity and release characteristics of folate-conjugated echogenic liposomes for cytosolic delivery of cancer drugs. Student author: Shirshendu Paul

4aBA4. High-frequency harmonic imaging with coded excitation: Implications for the assessment of coronary atherosclerosis. Student author: Himanshu Shekhar

4aBA6. Acoustic emissions associated with ultrasound-induced rupture of *ex vivo* blood vessels. Student author: Cameron L. Hoerig

4aBA7. Cavitation mechanisms in ultrasound-enhanced permeability of *ex vivo* porcine skin. Student author: Kyle T. Rich

4aBA8. Laser-induced-cavitation enhanced ultrasound thrombolysis. Student author: Huizhong Cui

4aBA9. Ethanol injection induced cavitation and heating in tissue exposed to high intensity focused ultrasound. Student author: Chong Chen

4pBA1. Effect of skull anatomy on intracranial acoustic fields for ultrasound-enhanced thrombolysis. Student author: Joseph J. Korfhagen

4pBA6. Histological analysis of biological tissues using high-frequency ultrasound. Student author: Kristina M. Sorensen

4pBA8. Parametric imaging of three-dimensional engineered tissue constructs using high-frequency ultrasound. Student author: Karla P. Mercado

WEDNESDAY AFTERNOON, 24 OCTOBER 2012 BASIE A1, 1:30 P.M. TO 2:20 P.M.

Session 3pED

Education in Acoustics: Acoustics Education Prize Lecture

Preston S. Wilson, Chair
Applied Research Lab., Univ. of Texas at Austin, Austin, TX 78712-0292

Chair's Introduction—1:30

Invited Paper

1:35

3pED1. Physclips: Multimedia, multi-level learning, and teaching resources. Joe Wolfe and George Hatsidimitris (University of New South Wales, School of Physics, University of New South Wales, Sydney, NSW 2052, Australia, j.wolfe@unsw.edu.au)

Physclips provides multimedia resources to physics students and teachers at the levels of senior high school to introductory university. Completed volumes cover mechanics, waves and sound. Each chapter includes a rich multimedia lesson of about 10 minutes, including film clips, animations, sound files and images of key experiments and demonstrations. Contextually embedded links lead to html pages providing broader and deeper support and, where needed, to tools such as calculus and vectors. The ongoing development of the interface reflects learner feedback and our own experience and research. The architecture and presentation of Physclips is largely consistent with evidence-based guidelines in the field of educational multimedia. Often, animations and labeling are superimposed on film clips to indicate abstract quantities, thus providing the novice with the insight of the expert's 'mind's eye'. The scrollbar is indexed with keywords and images to assist learners to find and to relocate conceptually discrete segments, which facilitates revision and reference usage. Together with extensive cross-linking, this allows students to construct individual learning pathways. Teachers download animations singly or in compressed folders for inclusion in lessons, blogs etc. Physclips is supported by Australia's Office of Learning and Teaching and the University of New South Wales.

Session 3pID

Interdisciplinary: Hot Topics in Acoustics

Lily M. Wang, Chair
Durham School of Architectural Engineering and Construction, University of Nebraska - Lincoln, Omaha, NE 68182-0816

Chair's Introduction—1:30

Invited Papers

1:35

3pID1. Hot topics in speech communication: Listening to foreign-accented speech. Tessa Bent (Department of Speech and Hearing Sciences, Indiana University, 200 S. Jordan Ave., Bloomington, IN 47405, tbent@indiana.edu)

There are currently more non-native English speakers in the world than there are native speakers. Most of these second language users will speak with a detectable foreign accent. Foreign-accented speech differs from native language norms along many acoustic-phonetic dimensions including the realization of vowel, consonant, and prosodic features. An important question for researchers in the field of speech communication is how this type of language variation influences speech perception and perceptual processing. Numerous findings have shown that foreign-accented speech is generally less intelligible, receives lower comprehensibility ratings, and is processed more slowly than native-accented speech. Further, these negative perceptual effects can be exacerbated by noisy listening conditions or listener variables such as age or hearing loss. However, research over the past several years has shown the amazing flexibility of the speech perception mechanism in its ability to adapt to this form of variability. Through experience and training, listeners can improve their word identification skills with specific foreign-accented talkers, particular foreign accents, and foreign-accented speech in general. New directions in this research area include perception of foreign-accented speech by infants and children as well as how a foreign accent may influence memory.

2:00

3pID2. New directions for manipulation of sound using acoustic metamaterials. Christina J. Naify, Gregory J. Orris, Theodore P. Martin, and Christopher N. Layman (Code 7160, Naval Research Laboratory, 4555 Overlook Ave SW, Washington, DC 20375, christina.naify.ctr@nrl.navy.mil)

Manipulation of sound waves using acoustic metamaterials has expanded significantly in recent years. Acoustic metamaterials are a class of materials that use sub-wavelength structures to achieve effective bulk properties under acoustic excitation. Unusual effective physical properties, including negative bulk modulus, negative mass density, and negative index have been achieved using metamaterials. Additionally, the development of structures based on transformational acoustics has resulted in designs for scattering reduction and sound focusing. Current research emphases include expansion from narrowband, resonant structures to broadband structures, as well as the design and construction challenges of three-dimensional structures. Active or tunable structures are also being explored. Examples will be given of negative index and three-dimensional metamaterial structures. [Work is supported by the Office of Naval Research.]

2:25

3pID3. Photoacoustic tomography: Ultrasonically breaking through the optical diffusion limit. Lihong Wang (Department of Biomedical Engineering, Washington University, One Brookings Drive, P.O. Box 1097, St. Louis, MO 63130-4899, lhwang@biomed.wustl.edu)

Photoacoustic tomography (PAT), combining optical and ultrasonic waves via the photoacoustic effect, provides in vivo multiscale non-ionizing functional and molecular imaging. Light offers rich tissue contrast but does not penetrate biological tissue in straight paths as x-rays do. Consequently, high-resolution pure optical imaging (e.g., confocal microscopy, two-photon microscopy, and optical coherence tomography) is limited to depths within the optical diffusion limit (~1 mm in the skin). In PAT, pulsed laser light penetrates the tissue and generates a small but rapid temperature rise, which induces emission of ultrasonic waves due to thermoelastic expansion. The ultrasonic waves, ~1000 times less scattering than optical waves in tissue, are then detected to form high-resolution images at depths up to 7 cm, breaking through the optical diffusion limit. PAT is the only modality capable of imaging across the length scales of organelles, cells, tissues, and organs with consistent contrast. Such a technology has the potential to enable multiscale systems biology and accelerate translation from microscopic laboratory discoveries to macroscopic clinical practice. PAT may also hold the key to the earliest detection of cancer by in vivo label-free quantification of hypermetabolism, the quintessential hallmark of cancer. The technology is commercialized by several companies.

Session 3pNS

Noise, ASA Committee on Standards, and Psychological and Physiological Acoustics: Passive and Active Noise Reduction in Hearing Protection

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

Hilary L. Gallagher, Cochair
Air Force Research Lab., Wright-Patterson AFB, OH 45433-7901

Chair's Introduction—1:00

Invited Papers

1:05

3pNS1. Development of an advanced hearing protection evaluation system. Kevin Shank and Josiah Oliver (Adaptive Technologies Inc., 2020 Kraft Dr Ste 3040, Blacksburg, VA 24060, kevin@adaptivetechinc.com)

Acoustic Test Fixtures (ATFs) are practical and often necessary tools for testing Hearing Protection Devices (HPDs) especially with extremely loud impulsive and/or continuous noise, for which the use of live subjects might not be advisable. Although there have been various standardized and laboratory ATFs from past research, there still exists large uncertainty in the correlation between the attenuation results obtained from ATFs and those obtained from actual human subject tests, particularly for intraaural HPDs. It is suspected that one of the main factors contributing to the discrepancy may be insufficient fidelity in the circumaural/intraaural flesh and bone. Human subject testing was performed to obtain median parameters of ear canal geometry and eardrum reflectance, which are considered to be critical parameters for circumaural/intraaural HPD attenuation performance. This presentation discusses the research methodologies and design implementation of these important subsystems in this advanced Hearing Protection Evaluation System (HPES).

1:25

3pNS2. Two case studies for fit testing hearing protector devices. William J. Murphy, Christa L. Themann, Mark R. Stephenson, and David C. Byrne (Hearing Loss Prevention Team, Centers for Disease Control and Prevention, National Institute for Occupational Safety and Health, 4676 Columbia Parkway, Cincinnati, OH 45226-1998, wjm4@cdc.gov)

Hearing protection devices (HPDs) are typically selected based upon the Noise Reduction Rating (NRR) and, until recently, were rarely tested for attenuation in real-world environments. The National Institute for Occupational Safety and Health has developed a fit-testing system (HPD Well-Fit^{TM}) that performs attenuation tests with a large circumaural earmuff, a portable computer and a computer mouse with a scroll wheel. HPD Well-Fit was used to estimate the attenuation of employees working in two different settings: inspectors for off-shore drilling rigs and sandblasters at a hydroelectric facility. The highest exposure levels for the inspectors and sandblasters were estimated to be 110 and 130 dBA, respectively. Fit testing and training were used to achieve a 25-dB Personal Attenuation Rating (PAR) for the inspectors. Fit testing before and after the sandblaster work shift demonstrated PARs of 30 to 42 dB using HPD Well-Fit. The average time to complete the fit tests was 10 minutes. If retraining was necessary, then an additional 3 to 6 minutes were required.

1:45

3pNS3. Continuous and impulsive noise attenuation performance of passive level dependent earplugs. Richard L. McKinley, Hilary L. Gallagher (Air Force Research Laboratory, 2610 Seventh Street, Wright Patterson AFB, OH 45433, richard.mckinley@wpafb.af.mil), Melissa Theis (Oak Ridge Institute for Science and Education, Dayton, Ohio), and William J. Murphy (National Institute for Occupational Safety and Health, Cincinnati, OH)

Level dependent hearing protectors, earplugs and earmuffs, have advanced in technology due to the needs of military personnel and others to reduce the risk of hearing damage from impulsive noise. These hearing protectors were developed to preserve ambient listening capabilities therefore improving situational awareness while reducing the risk of noise induced hearing loss by attenuating both continuous and impulsive noise. Four commercially available passive level dependent earplugs were assessed for both continuous noise attenuation and impulsive insertion loss performance. The continuous noise attenuation results were collected using American National Standard Institute (ANSI) S12.6-2008 Methods for Measuring the Real-Ear Attenuation of Hearing Protectors while the impulsive insertion loss results were collected using ANSI S12.42-2010 Methods for the Measurement of Insertion Loss of Hearing Protection Devices in Continuous or Impulsive Noise Using Microphone-in-Real-Ear (MIRE) or Acoustic Test Fixture Procedures. The presentation will include the passive noise attenuation performance of level dependent earplugs for both continuous and impulsive noise. The impulsive insertion loss results for these particular hearing protectors will be applied to impulsive noise damage risk criteria for an estimate of allowable impulsive noise exposure.

2:05

3pNS4. Effective attenuation performance of passive hearing protectors: A temporary threshold shift study. Richard L. McKinley, Hilary L. Gallagher (Air Force Research Laboratory, 2610 Seventh Street, Wright Patt AFB, OH 45433, richard.mckinley@wpafb.af.mil), and Melissa Theis (Oak Ridge Institute for Science and Technology, Dayton, Ohio)

Passive hearing protectors have been used for decades to reduce the risk of noise induced hearing loss. Hearing protectors (earmuffs, earplugs, helmets) have traditionally been the first line of defense for personnel working in hazardous noise environments. According to ANSI S12.68-2007, the "gold standard" method of estimating effective A-weighted sound pressure levels when hearing protectors are worn is the classical octave band method. The octave band method subtracts the hearing protector noise attenuation from the ambient noise level for each relevant octave band to estimate the noise exposure at the ear, under the hearing protector. ANSI S12.6-2008 Methods for Measuring the Real-Ear Attenuation of Hearing Protectors was used to measure the attenuation of the hearing protectors. The purpose of this study was to measure the effective attenuation of a hearing protector in terms of temporary threshold shift (TTS) response for individual human subjects with and without hearing protection. This presentation will include the TTS response curves for subjects exposed to various noise levels and durations in a controlled laboratory environment. The passive hearing protectors evaluated in this study included an earplug, earmuff, and a headphone with minimal attenuation as determined by REAT.

Contributed Papers

2:25

3pNS5. Measurements of bone-conducted impulse noise from weapons using a head simulator. Odile H. Clavier, Anthony J. Dietz, Jed C. Wilbur (Creare Inc., 16 Great Hollow Rd, Hanover, NH 03755, ohc@creare.com), Edward L. Zechmann, and William J. Murphy (Hearing Loss Prevention Team, National Institute for Occupational Safety and Health, Cincinnati, OH)

High-intensity impulse sounds are generally considered to be more damaging than continuous sounds, so understanding the attenuation performance of hearing protection devices against impulse noise is key to providing adequate protection for exposed persons. The maximum attenuation of hearing protection devices is limited by bone-conducted sound. Weapon fire noise in the form of short duration impulses can reach peak levels of 170 dB SPL at the shooter's ear, a sound level for which maximum hearing protection is recommended and for which bone-conducted sound will be a significant factor. However, current acoustic test fixtures do not capture the bone-conducted sound paths. In this study, an anatomically correct head simulator built specifically to measure bone-conducted sound was used to evaluate the effects of impulse noise generated by hand guns and rifles at several peak sound pressure levels ranging between 120 dB SPL and 170 dB SPL. Time histories of the acceleration of the temporal bones and the sound pressure transmitted into the cranial cavity were recorded. Results investigating the linearity of the bone-conducted response to impulse noise at high peak levels and the effects of hearing protection on the sound level and vibrations inside the head are presented.

2:40

3pNS6. Adaptive feedforward control for active noise cancellation in-ear headphones. Sylvia Priese, Christoph Bruhnken (Institute of Measurement and Automatic Control, Leibniz Universität Hannover, Nienburger Straße 17, Hannover, 30167, Germany, sylvia.priese@imr.uni-hannover.de), Daniel Voss, Jürgen Peissig (Technology and Innovation, Sennheiser Electronic GmbH & Co. KG, Wedemark, NI, Germany), and Eduard Reithmeier (Institute of Measurement and Automatic Control, Leibniz Universität Hannover, Hannover, NI, Germany)

Noise can be disturbing, stressful or even harmful. Headphones with active noise cancellation (ANC) can enhance the user's comfort, especially when travelling. On a plane or a train, in the street or at work, these headphones give the possibility to reduce unwanted noise. The range of ANC headphones on the market is constantly increasing. Circumaural and supra-aural headphones with different control strategies have been available for a long time; over the last few years the product lines have been expanded to in-ear headphones. These headphones already have quite a good passive attenuation and are equipped with feedforward control for active noise cancellation. The best results in attenuation are achieved by semi-adaptive digital controls, which choose the best filter depending on the noise spectrum and can be manually adapted to the user. A fully adaptive control has already been proven to be very effective in aviation headsets and other ANC applications. Besides the market analysis of ANC headphones we would like to present an adaptive feedforward control for in-ear headphones and highlight the advantages compared to a static feedforward control.

2:55

3pNS7. Design of a feedback controller for active noise control with in-ear headphones. Christoph Bruhnken, Sylvia Priese (Institute of Measurement and Automatic Control, Leibniz Universität Hannover, Nienburger Straße 17, Hannover, 30167, Germany, christoph.bruhnken@imr.uni-hannover.de), Hatem Foudhaili, Jürgen Peissig (Technology and Innovation, Sennheiser Electronic GmbH & Co. KG, Wedemark, NI, Germany), and Eduard Reithmeier (Institute of Measurement and Automatic Control, Leibniz Universität Hannover, Hannover, NI, Germany)

Nowadays mobility is an important factor in many jobs. Therefore, there is an increased use of planes, trains and cars, and the associated exposure to noise. Good acoustic insulation is often hard to realize due to the involved extra weight. Ear protection or headphones with active noise control (ANC) may be a possible solution. Today circumaural and supra-aural ANC headphones with good attenuation are commercially available. However, their weight and the necessary headband can impair the wearing comfort. ANC in-ear headphones do not have these disadvantages and, therefore, there is a need of further research in the field of ANC. In ANC headphones, disturbing noise is minimized by an out-of-phase anti-noise. Therefore, the noise is recorded by microphones next to each ear, and filtered by an analog or digital platform to generate the anti-noise. There are two main control strategies depending on the position of the microphones, feedforward control with an external reference microphone and feedback control with an internal error microphone. The presentation will focus on the design of feedback controllers and the main problem regarded to in-ear headphones, interpersonal variances, which make the design of stable controllers with high noise attenuation difficult. A model-based solution will be presented.

Session 3pUW

Underwater Acoustics and Signal Processing in Acoustics: Advances in Underwater Acoustic Communication

Hee-Chun Song, Chair
Scripps Institution of Oceanography, La Jolla, CA 92093-0238

Contributed Papers

1:15

3pUW1. Active average intensity based on single vector sensor. Pengyu Du, Xiao Zhang, Jingwei Yin, and Xiao Han (College of Underwater Acoustic Engineering, Harbin Engineering University, Harbin, Heilongjiang 150001, China, yinjingwei@hrbeu.edu.cn)

Code divided multiple access underwater communication based on single vector sensor is studied in this paper. The most common methods to estimate azimuth with self-directivity of single vector sensor are average sound intensity method and complex sound intensity method, however, for the same frequency band multi-users, theoretical limitation for these methods is only two users.Spread spectrum communication is featured with strong anti-multipath, anti-interference, secret-keeping and communication web composing ability. Active average intensity method, which measures azimuths of multi-users simultaneously with the excellent correlative characteristics of pseudo-random code in spread spectrum communication, is proposed in this paper. Simulation and experiment for same frequency band spread spectrum multi-user communication testify the feasibility and utility of active average sound intensity method. With the estimated azimuth, vector combination can be generated to adjust the directivity of vector sensor, achieve multi-user beam communication, inhibit multi-path interference, and enhance processing gain and lower error rate. Key words: underwater acoustic communication; CDMA; single vector sensor; active acoustic intensity average

1:30

3pUW2. The application of differential spread spectrum technology in underwater acoustic communication. Xiao Han, Jingwei Yin, Pengyu Du, and Xiaoyu Guo (College of Underwater Acoustic Engineering, Harbin Engineering University, Mudanjiang, Harbin, Heilongjiang 150001, China, yinjingwei@hrbeu.edu.cn)

In underwater acoustic channel, the Doppler effect produced by relative movement between Source and information destination is very complex. Currently, the spread spectrum system typically uses PSK modulation. As the transmission characteristic in water sound channel is phase rapid changing, spread spectrum systems based on PSK modulation need high precision in estimating the carrier and need continuous tracking of the carrier, which make the performance in practical applications limited. Differential spread spectrum acoustic communication technology is studied in this paper. Using differential coherent demodulation method at the receiving end, which solves the problem of estimating the carrier in underwater acoustic communication, can overcome the frequency and phase error due to the drift of the carrier in transfer process. This method is verified Through computer simulation studies and Lake test.

1:45

3pUW3. Research on multilevel differential amplitude and phase-shift keying in convolution-coded orthogonal frequency division multiplexing underwater communication system. Yuheng Zhang, Chi Wang, Jingwei Yin, and Xueli Sheng (College of Underwater Acoustic Engineering, Harbin Engineering University, Harbin, Heilongjiang 150001, China, yinjingwei@hrbeu.edu.cn)

With the increasing demands of underwater source development and the increase of users, underwater transfer information has also greatly increased and high-bit-rate underwater acoustic communication has become a hot topic in underwater acoustic communication research. MDAPSK (Multilevel Differential Amplitude and Phase-shift Keying) is a modulation technique having high efficiency in spectrum utilization, which transfers information by using differential amplitude and phase code, demodulates information by adopting coherent demodulation. It reduces the difficulty of system, and improves the speed of transmission. While OFDM (Orthogonal Frequency Division Multiplexing) has advantages including high efficiency in spectrum and faster communication speed. After describing the schemes of the two technologies, a design scheme which concerns application of MDAPSK in the OFDM based underwater communication was given. The convolutional codes are also used in this system to realize the effectiveness and reliability in high-bit-rate underwater wireless communication. The computer simulation and the channel pool experimentation show that the system has a better performance. Key words: MDAPSK; OFDM; convolution coding; high-bit-rate communication

2:00

3pUW4. Application of orthogonal frequency division multiplexing in cognitive underwater communication. Chi Wang, Jingwei Yin, Pengyu Du, and Longxiang Guo (College of Underwater Acoustic Engineering, Harbin Engineering University, Mudanjiang, Harbin, Heilongjiang 150001, China, yinjingwei@hrbeu.edu.cn)

With the development of underwater acoustic communication in military and commercial field and the urgent need for underwater wireless Ad Hoc networks, developing an intelligent and high-bit-rate underwater communication system is imminent. OFDM(Orthogonal Frequency Division Multiplexing) technology could be a good platform for cognitive underwater communication, which has advantages including high efficiency in spectrum utilization, faster communication speed and flexibility in choosing frequencies. A design scheme of the OFDM based cognitive underwater communication and block diagram are given. The system can intelligently choose NC-OFDM (Non-Contiguous OFDM), DOFDM (Differential OFDM) or Pilot-added OFDM communication schemes in order to meet different channel conditions and different rate requirements and to overcome the problem of data conflict and the waste of spectrum resources in multi-users' competitive communication. Meanwhile, the system also can intelligently choose parameters in each scheme, such as sub-channel, pilot interval and error-correcting codes. The simulation results prove the feasibility and effectiveness of the OFDM based cognitive underwater communication.

2:15

3pUW5. Volterra series-based non-linearity analysis of shallow water acoustic channels. Xiaopeng Huang (Dept. of Electrical and Computer Engineering, Stevens Institute of Technology, Castle Point on Hudson, Hoboken, NJ 07030, xiaopeng.huang0508@gmail.com)

Most of existing underwater acoustic (UWA) communication systems are based on the linear UWA channels. However, some environmental factors (e.g., bubble plumes) in complicated shallow water environments will contribute to the non-linearity of channels. Therefore, In order to fully understand the properties of shallow water acoustic channels, and develop more bandwidth-efficient communication systems in complicated shallow

water environments, we adopt the Volterra series to analyze the non-linearity of shallow water acoustic channels for the first time, and its completed theoretical derivations will be presented. Volterra series combines the representations of a nonlinear system without memory and a linear, casual system with memory to describe a nonlinear system with memory. Generally speaking, the central problem in using a Volterra approach to the analysis of nonlinear channels with momory consists of estimating the Volterra kernels, which represent a nonparametric characterization of the channel.

2:30

3pUW6. Shallow water acoustic channel modeling with adaptive communications. Xiaopeng Huang (Dept. of Electrical and Computer Engineering, Stevens Institute of Technology, Castle Point on Hudson, Hoboken, NJ 07030, xiaopeng.huang0508@gmail.com)

The underwater acoustic channel is known to be severely bandwidth limited due to sound attenuation by sea water, and interaction with the ocean surface and bottom. Yet, shallow water acoustic channels at high frequencies are little understood, particularly in shallow water environments, and hence the quest for achieving a viable adaptive communication solution has been a challenge that perplexed scientists for a long time. In this abstract, we first take Hodson River estuary as an example to investigate the characterizations of shallow water environments, which mainly comprises the evaluation of key channel parameters such as the scattering function, Doppler shift, coherent bandwidth, coherent time, 2D (i.e., Time-Frequency) time-variant channel impulse response (CIR). The study will also cover channel fading statistics, and water conditions that affect the CIR (e.g., bubble plumes and mddium inhomogeneities). Finally, the models developed will be used to evaluate the achievable performance of channel estimation and adaptive communication systems in shallow water acoustic media.

2:45

3pUW7. Bidirectional equalization for underwater acoustic communications. Hee-Chun Song (Scripps Institution of Oceanography, 9500 Gilman Drive, La Jolla, CA 92093-0238, hcsong@mpl.ucsd.edu)

The bi-directional decision feedback equalizer (BiDFE) that combines the outputs of a conventional DFE and backward DFE can improve the performance of the conventional DFE by up to 1-2 dB based on simulations. In this paper, the BiDFE concept is extended to multi-channel time reversal communications involving a DFE as a post-processor. Experimental data collected in shallow water (10-20 kHz) show that the performance can be enhanced by 0.4-1.8 dB in terms of output SNR. In particular, a larger improvement (e.g., 1.8 dB) is achieved for time-varying channels where the channel diversity in opposite directions is more profound.

Plenary Session, Business Meeting, and Awards Ceremony

David L. Bradley, President
Acoustical Society of America

Business Meeting

Presentation of Certificates to New Fellows

Peter F. Assmann
Yang-Hann Kim
David A. Eddins
William J. Murphy
John A. Hildebrand
Scott D. Pfeiffer
Peter Howell
John R. Preston
Andrew J. Hull
Ronald C. Scherer

Presentation of Awards

Medwin Prize in Acoustical Oceanography to John A. Colosi

Rossing Prize in Acoustics Education to Joe Wolfe

Silver Medal in Animal Bioacoustics to Richard R. Fay

Silver Medal in Noise to Keith Attenborough

Silver Medal in Physical Acoustics to Andrea Prosperetti

Session 3eED

Education in Acoustics and Women in Acoustics: Listen Up and Get Involved

Marcia J. Isakson, Cochair
Applied Research Laboratories, University of Texas at Austin, Austin, TX 78713

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

Contributed Paper

5:30

3eED1. Hands-on demonstrations for Project Listen Up: Education outreach Part IV. Jacqueline A. Blackburn and Murray S. Korman (Physics Department, U.S. Naval Academy, Chauvenet Hall Room 295, Annapolis, MD 21402, korman@usna.edu)

Acoustical demonstrations geared to promote a hands-on learning experience for middle- and high-school age Girl Scouts are setup. The participants will be free to explore, control the apparatus and make their own scientific discoveries. The hands-on demonstrations will include (1) a home-made electric slide guitar using easy to find parts, (2) a safe smoke ring generator and (3) a portable ripple tank with plastic eyedroppers for simple excitation of waves.

WEDNESDAY EVENING, 24 OCTOBER 2012 7:30 P.M. TO 9:30 P.M.

OPEN MEETINGS OF TECHNICAL COMMITTEES

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

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

Committees meeting on Wednesday are as follows:

Biomedical Acoustics	Mary Lou Williams
Signal Processing in Acoustics	Lester Young A