The just noticeable difference (JND) or smallest detectable increment of clarity index (C80) has been investigated due to the lack of consensus in the existing literature. The purpose of this study was to determine how the JND of C80 varies as a function of the test procedure. Test signals, with varying amounts of clarity, were generated and combined with short anechoic recordings of orchestral music. The testing took place in the University of Hartford’s anechoic chamber, and the signals were played back over eight spatially arranged loudspeakers. Two testing methods were compared, which both consisted of the subject hearing two signals, A and B, and deciding if the signals were the same or different in terms of clarity. For Test Method 1, the subjects were required to listen to all of signal A and then all of signal B before selecting their response. For Test Method 2, the subjects were allowed to switch between signals A and B in real-time. The difference in the JND of C80 for the two test methods, along with a comparison to previously published results, will be discussed. [Work is supported by the Paul S. Veneklasen Research Foundation.]

1:15
5pAA2. The effect of motif length in reverberation-time listening tests using the double-blind testing method ABX. Scott S. Edwards, Daniel A. Ignatiuk, Robert D. Celmer, and Michelle C. Vigeant (Dept. Mech. Eng., Acoust. Prog. and Lab., Univ. of Hartford, 200 Bloomfield Ave., West Hartford, CT 06117, vigeant@hartford.edu)

Subjective listening tests are often conducted to determine the relationship between an objective parameter and human perception. These tests can consist of auralizing measured or predicted room impulse responses (RIRs) that have been convolved with anechoic recordings. However, no previous research exists that provide guidelines in terms of how auditory memory affects the chosen length of the musical motif convolved with the RIR. Research in the field of auditory memory suggests that there is both short auditory memory of approximately 200 ms and long auditory memory lasting several seconds [Cowan, Psychol. Bull. 96, 341–370 (1984)]. A study was carried out using two motifs with three different lengths: 5, 7, and 10 s, where the successively longer motifs contained the same passages as the shorter ones. A relatively straightforward parameter of reverberation time was the independent variable, and signals were created with reverberation times between 1.0–1.5 s. The subjects were presented three signals, A, B, and X, where the difference in reverberation time ranged between 0.3–0.5 s. The subjects were required to identify which signal A or B matched the test signal X. The effect of motif length on the percent correct will be discussed. [Work is supported by the Paul S. Veneklasen Research Foundation.]
Every acoustic floor or wall design requires an analysis in order to provide a sustainable design. This is true whether the design is a single family residence with the goal of achieving a desired level of quietude, or a multifamily, or commercial dwelling required to meet various codes and/or standards. Since IIC and STC measurements taken in the field produce a lower STC or IIC than laboratory controlled values, this presentation will begin by reviewing the variables that are not present in the controlled laboratory setting for their acoustic impact. This presentation will further provide field readings of well known documented basic floor and wall configurations. With this foundation standard floor and wall configurations will be reviewed for their FSTC and FIIC improvement with standard readily available acoustic materials to address variables consistent with the degradation of the configuration’s acoustic efficacy. The FIIC and FSTC values of the same floors and walls will be further compared to walls and floors with unique and innovative installation methodologies and configuration upgrades. The summary will be a detailed approach to the best field practices and innovative configuration upgrades, inclusive of a few new patented and patent pending products, for optimum field acoustic efficacy.

For steady-state broadband sound fields in enclosures, surfaces may be modeled as diffuse reflectors, specular reflectors, or some combination. Modeling of reflection surfaces when using energy-intensity based boundary elements is reviewed, and the proper energy-intensity source characteristics are derived. These characteristics depend on local correlation effects, the angular distribution of the incident field, and the angular dependence of reflection coefficient. Specular reflection surfaces subjected to fairly random reverberant sound fields exhibit behavior similar to diffuse reflection surfaces to lowest order. Simple solutions are obtained to illuminate the basic difference in interior sound fields caused by different reflection types. For example, random incidence sound propagating down a long corridor with non-absorbing walls exhibits different behavior depending on whether the surfaces are diffuse or specular reflectors. The diffuse case exhibits a spatial gradient in mean-square pressure along the corridor, whereas the specular case gives a uniform field. These solutions are also used to clarify the relationship between the flow of intensity and the gradient of mean-square pressure. This relationship has been represented simplistically and incorrectly in some energy based methods. The importance of benchmarking energy based methods against analytical solutions, rather than experimental data, is stressed.

Case study reviewing the speech transmission index and voice intelligibility design and implementation of two atrium spaces where dedicated public address systems were implemented. The architectural design and selection of finishes has a great impact on the capability of any public address loudspeaker system distinct from any system performance characteristics. The presentation will review the design process, acoustical modeling, and testing results outlining the issues associated with meeting the NFPA 72 Common Intelligibility Score suggested goals. Architectural design requirements such as day-lighting and views are increasing large areas of glass in atrium spaces. These designs are making intelligibility increasingly difficult and the electronic systems and loudspeaker designs can only accomplish so much without architectural acoustical finishes as part of the design.

Classrooms have been shown to be prone to elevated occupied noise levels that reduce speech comprehension and inhibit learning. This presentation discusses a current study that includes three parts: (1) the characterization of masking invoked by noise commonly found in today’s classrooms, (2) the formulation of a metric that intends to associate masking risk to noise features, and (3) the attempt to quantify any increase in speech comprehension resulting from interior acoustic designs that aspire to reduce noise in occupied classrooms. Calibrated recordings made during classroom activities in 9 schools are post processed to allow noise characteristics to be rated based on attributes related to the noise source or type, duration, rate of recurrence, spectra, level, envelope, and peak energy. Binaural and monaural recordings are compared. Speech to noise ratios are statistically weighted over various time frames and activities in an attempt to refine reproducibility among different activities. All recordings were made in occupied rooms both before and after the various treatments were integrated. The results suggest that there are misconceptions in the literature in terms of today’s classroom design trends and a few novel principles emerged as being highly effective.
Session 5pAB

Animal Bioacoustics and Psychological and Physiological Acoustics: Auditory Attention, Learning and Memory: From Neurons to Behavior

Cynthia F. Moss, Chair
Univ. of Maryland, Dept. of Psychology, College Park, MD 20742

Invited Papers

1:00

5pAB1. Focusing, maintaining, and switching attention. Barbara G. Shinn-Cunningham (Hearing Res. Ctr., Boston Univ., 677 Beacon St., Boston, MA 02215)

Humans and other animals have an amazing ability to selectively attend to whatever sound source is most relevant, a task that requires the brain to separate a sound mixture into distinct perceptual objects. Results from a number of recent behavioral and neuroimaging studies from our laboratory and others have begun to tease apart the processes and mechanisms that enable us to select an important source from a sound mixture and attend to it as its meaning unfolds, as well as to switch attention if the need arises. As reviewed in this talk, the processes of focusing, maintaining, and switching auditory attention involve dynamics that directly impact our ability to understand sounds in complex settings. For instance, there is a cost associated with the process of disengaging attention from one source and focusing on another. On top of this switching cost, there is a benefit of maintaining attention to an ongoing source that yields improvements in performance over time. These processes interact with the way we store and remember signals and directly influence how we function in complex auditory scenes, especially social settings like the Speech Communication Poster Session or the Thursday Evening Buffet Social.

1:25

5pAB2. The role of innate and attentional mechanisms in parsing complex acoustic scenes: A neural and behavioral study. Mounya Elhilali (Dept. of Elec. & Comput. Eng., Johns Hopkins Univ., 3400 N. Charles St., Baltimore, MD 21218, mounya@jhu.edu)

The mechanisms by which a complex auditory scene is parsed into coherent objects depend on poorly understood interactions between task-driven and stimulus-driven attentional processes. We use a simultaneous psychophysical-neurophysiological experimental paradigm to manipulate human listeners attention to different features of auditory scenes. In a series of experiments, our findings reveal a role of attention in enhancing the sustained neural representation of the foreground. This enhancement, in both power and phase coherence, originates in auditory cortex, occurs exclusively at the frequency of the target rhythm, and is only revealed when contrasting two attentional states that direct subjects focus to different features of the acoustic scene. It also interacts with innate processes of the auditory system, particularly its differential sensitivity to temporal dynamics of sounds. These results have substantial implications for models of foreground/background organization and mechanisms mediating auditory object formation.

1:50

5pAB3. Intention and attention: Top-down influences on the representation of task-relevant sounds. Jonathan B. Fritz (Inst. for Systems Res., ECE, Univ. of Maryland, College Park, MD 20747, ripple@isr.umd.edu), Stephen V. David, Daniel Winkowski, Pingbo Yin (Univ. of Maryland, College Park, MD 20747), Mounya Elhilali (Johns Hopkins Univ., Baltimore, MD 21218), and Shihab A. Shamma (Univ. of Maryland, College Park, MD 20747)

To explore the role of top-down projections from frontal cortex (FC) to auditory cortex (AC), which may play a role in adaptive reshaping of A1 receptive fields to attended acoustic stimuli during behavior, simultaneous recordings were made from single neurons in FC and AC in behaving ferrets, trained on multiple auditory detection and discrimination tasks in positive and negative reinforcement paradigms. Performance required selective attention to different salient spectral frequency and/or temporal cues. Previous observations revealed task-specific transformations in receptive fields in AC [Fritz et al. (2003); (2005); (2007); (2009); Atiani et al. (2009)]. In contrast, FC neurons showed recognition responses, which categorically distinguished between acoustic foreground and background stimuli. FC responses to targets were often independent of the acoustic properties of the target and thus encoded an abstract representation of the class of target stimuli [Fritz et al. (2009)]. Stimulation of FC, paired with tones, lead to receptive field transformations similar to those observed in behavior. These results emphasize the importance of interactions between multiple areas during selective attention, and the tight coupling of target recognition and auditory attention that enhances auditory cortical filters for attended acoustic stimuli, thus creating a functional representation of task-relevant sounds during behavior. [This work was supported by grants from NIDCD, NIH.]
5pAB4. Learning alters the coding capacity of single neurons and populations in the auditory forebrain. James Jeanne (Neurosci. Graduate Program, Univ. of California San Diego, La Jolla, CA 92093), Tatyana Sharpee (Salk Inst. for Biological Studies, La Jolla, CA 92037), and Timothy Gentner (Univ. of California, San Diego, La Jolla, CA 92093)

To control adaptive behaviors, vertebrate neural systems must extract information from large numbers of diverse, physically complex signals in the natural world. Sensory experience helps neural systems to meet this challenge by evoking long-lasting changes that bias cortical circuits toward representations of behaviorally relevant signals. Such circuits may encode stimuli using highly selective individual neurons or by larger populations of neurons, thereby greatly expanding coding capacity. I will describe studies investigating the representation of a complex acoustic communication signal, birdsong, across multiple forebrain regions that are analogous to secondary auditory cortices in mammals. I will show that single neurons and populations of neurons in these regions are tuned to the complex spectro-temporal features in songs that birds have learned to recognize. These results demonstrate that learning mechanisms can act and different organizational levels in the brain, and provide biological support for specific population coding schemes.

2:40

5pAB5. Top-down vocal feedback control in active hearing. Xiaoqin Wang (Dept. of Biomedical Eng., Johns Hopkins Univ., 720 Rutland Ave., Baltimore, MD 21205, xiaoqin.wang@jhu.edu)

Speaking is a sensory-motor process that involves constant self-monitoring to ensure accurate vocal production. This monitoring of vocal feedback allows a speaker to quickly adjust speech production to correct perceived errors between intended and actually produced vocal sounds. The self-monitoring in speaking is crucial for learning to speak native as well as foreign languages. An important behavior in vocal feedback control for both human speech and animal vocalizations is the compensatory change in speech or vocalizations (e.g., pitch, frequency, and intensity) when there is a mismatch between intended and perceived vocal sounds. Such a behavior requires mechanisms for continuously monitoring auditory feedback during vocal production. We have found that disruption of auditory feedback during vocalization alters coding properties of auditory cortex neurons in marmosets (a highly vocal primate species). Furthermore, when marmosets compensate for changes in vocal feedback, there are corresponding changes in their cortical neural activity. These findings suggest that the neural network underlying self-monitoring of vocal production likely consists of both sensory processing and top-down modulations via higher-cortical areas involving planning and memory.

3:05—3:20 Break

Contributed Papers

3:20
5pAB6. Production and perception of sequential patterns in budgerigar (Melopsittacus undulatus) warble: Evidence for a magic number. Hsiao-Wei Tu and Robert J. Dooling (Dept. of Psych., Univ. of Maryland, College Park, MD 20742, hsiaowei@umd.edu)

The warble song of budgerigars (Melopsittacus undulatus) is composed of a large number of elements uttered in streams lasting from a few seconds to a few minutes without obvious repetition of particular patterns. Previous work showed that warble elements can be classified into eight acoustic categories for which budgerigars have corresponding perceptual categories. Here, we analyzed long sequences of natural warble to determine the proportion of these categories in warble as well as any sequential patterns of these categories in warble. Results showed that elements were not randomly arranged and that warble has at least a fifth-order Markovian structure. This suggests that budgerigars may have a motor control of approximately five elements during the production of warble. Investigations of the budgerigars’ ability to perceive warble sequences of various lengths showed convergent evidence that budgerigars are able to master a novel sequence between four and seven elements in length. In other words, budgerigars may have an attention span of about five elements which may be a limit of their capacity of neural processing. Through gradual training with chunking (about five elements), birds are able to learn sequences up to 50 elements. [Work supported by NIH/NIDCD R01DC001198 to R.J.D.]

3:35
5pAB7. Perception of alterations in natural budgerigar (Melopsittacus undulatus) warble: Implications of animal “syntactical capability.” Hsiao-Wei Tu and Robert J. Dooling (Dept. of Psych., Univ. of Maryland, College Park, MD 20742, hsiaowei@umd.edu)

The warble song of male budgerigars (Melopsittacus undulatus) is an extraordinarily complex, multisyllabic, learned vocalization that is produced continuously often for minutes at a time. Previous work has shown that warble elements form acoustic and perceptual categories for budgerigars and these birds can detect ordering changes in artificial sequences of warble elements. Using operand conditioning and a psychophysical paradigm, we examined the sensitivity of budgerigars for detecting different types of insertions (e.g., targets) in a running background of warble up 1000 elements in length. When the inserted targets are warble calls taken directly from the background, budgerigars show a species-specific ability to detect them solely based on sequence violations in the natural ordering of warble elements. Moreover, budgerigars, but not other species, are especially sensitive to temporally reversed warble elements inserted in natural warble sequences, indicating that the acoustic details of warble elements are also perceptually significant to budgerigars besides sequential cues. Although it is still unclear whether budgerigars perceive their warble as a rule-governed sequence or a pattern-based vocalization, the findings here open the door to studies of serial order learning in a natural, non-human communication system. [Work supported by NIH/NIDCD R01DC001198 to R.J.D.]

3:50
5pAB8. Bioacoustic and behavioral correlates of spatial memory in echolocating bats. Jonathan R. Barchi (Dept. of Neurosci., Brown Univ., 185 Meeting St., Providence, RI 02912, barchi@brown.edu), Jason E. Gaudette, Jeffrey M. Knowles, and James A. Simmons (Brown Univ., Providence, RI 02912)

Echolocating bats face a unique orientation problem due to their reliance on echolocation over visual orientation. Each observation (sonar vocalization) only contains information about objects in front of and within at most 5–10 m of the animal. Because their behavior extends over much larger distances, spatial memory might be particularly important during foraging activity and free flight to integrate the shorter views provided by sonar. To investigate memory for object locations, Eptesicus fuscus were allowed to fly freely in an instrumented flight room populated with sparse obstacles (hanging chains). Bat position and vocalization were monitored with a stereo-registered pair of thermal cameras and an array of ultrasonic microphones synchronized to the video. Data were analyzed for correlates of spatial memory-flight path dynamics, head aim, and temporal structure of echolocation signals. Data recorded from bats free-flying in the chain array are consistent with memory of obstacle locations and high-level planning of a flight path through the entire space. These data reflect the behavior of bats flying in natural conditions. [Work supported by ONR and NSF.]
5pAB9. Effect of ototoxic drugs on *Rana catesbeiana* tadpole orientation behavior and caspase-3 expression. Erika Alexander, Brian Schmidt, and Andrea Simmons (Dept. of Psych., Brown Univ., 89 Waterman St., Providence, RI 02912, erikaealexander@gmail.com)

Treatment with the ototoxic antibiotic gentamicin results in specific damage to hair cells of the inner ear and possibly of the lateral line system. In amphibians, damaged hair cells may spontaneously recover. *Rana catesbeiana* tadpoles were treated with different dosages of gentamicin solution, including a nontreated control, for 24 h. Animals were tested behaviorally and then sacrificed either immediately or after a 7 day recovery period.

Their brains were processed for immunohistofluorescent labeling of caspase-3, a marker for apoptosis. Behavioral changes associated with gentamicin treatment included a disruption of balance, altered swimming, and an inability to detect water currents. Animals treated with gentamicin showed increased caspase-3 expression in multiple brain areas compared to non-treated controls. Animals allowed a 7 day recovery period that showed lower caspase-3 label across brain areas than animals sacrificed immediately after treatment. These results suggest that ototoxic damage is associated with increased caspase-3 expression in the central nervous system. In tadpoles, affected behaviors undergo spontaneous recovery, which is itself correlated with decreases in caspase-3 expression. [Work supported by NSF GRFP, NIH, and RI Space Grants.]

FRIDAY AFTERNOON, 23 APRIL 2010

**Session 5pBB**

**Biomedical Ultrasound/Bioresponse to Vibration: Ultrasonic Characterization of Bone II**

Keith A. Wear, Chair

*Food and Drug Administration, 10903 New Hampshire Ave., Silver Spring, MD 20993-0002*

**Invited Papers**

1:00


Reliable quantification of microstructural and mechanical bone properties remains an open issue with relevance for the diagnosis of bone quality disorders, such as osteoporosis. The reconstruction of such parameters from nondestructive testing based on ultrasound testing and model-based solution of the identification inverse problem have been suggested by several investigators as novel techniques with high potential not only due to the reduced cost and its non-ionizing nature, but for the direct relationship and sensitivity of the propagation characteristics estimates to relevant bone properties that determine mechanical strength. Recently, approaches taking advantage of the poroelastic nature of cancellous bone have been developed. These approaches assume that the Biot theory is a valid model of wave propagation and are aiming at either (i) recovering the properties of the fast and slow waves or (ii) directly reconstructing the bone properties that govern Biot’s theory. However, the most recent literature reveals inconsistencies between Biot model and experimental results, thereby suggesting that Biot model may not be the most appropriate theoretical framework for solving the inverse problem. The most recent results will be reviewed and future research directions will be highlighted to overcome the observed inconsistencies.

1:15

5pBB2. Structural and mechanical assessment of trabecular bone by *in vivo* magnetic resonance imaging. Felix Wehrli (Dept. of Radiology, 3400 Spruce St., Philadelphia, PA 19104, wehrli@mail.med.upenn.edu)

Bone is a complex composite material whose strength is determined by a combination of the material’s intrinsic mechanical properties, its overall volume fraction, and architecture at the macro-, micro- and ultrastructural levels. Trabecular bone, the type of bone prevalent in the vertebrae and ends of long bones, is particularly prone to fracture in osteoporosis. It consists of a network of interconnected plates and struts of about 100-µm thickness immersed in a matrix of marrow. Advances in magnetic resonance imaging technology, including development of high-field magnets and radiofrequency coil technology, along with improved image processing and analysis techniques, now allow acquisition of images from which the three-dimensional (3D) microarchitecture can be retrieved, at least at peripheral skeletal locations such as the distal radius and tibia. From these data measures of topology, scale and orientation of the trabecular network, all contributing to the bone’s mechanical competence, can be derived with high-serial reproducibility. The processed images can also be used to create 3-D meshes and mechanical parameters such as elastic and shear moduli computed by micro-finite-element analysis. The translation of this technology to clinical medicine is highlighted with examples from recent drug intervention studies, allowing direct assessment of structural and mechanical consequences of treatment.

**FRIDAY AFTERNOON, 23 APRIL 2010**
5pBB3. The effect of structural anisotropy on the fast wave propagation in cancellous bone. Mami Matsukawa, Katsunori Mizuno (Lab. of Ultrasonic Electron, Doshisha Univ., Kyotanabe 610-0321, Japan, mmatsuka@mail.doshisha.ac.jp), and Yoshiki Nagatani (Kobe City College of Technol., Kobe, 651-2194, Japan)

QUS parameters are closely related to the structural properties and elastic properties of bone, which can provide important information related to bone quality and bone strength. One should be very careful, however, that bone contains complicated structure from microscopic to macroscopic levels. For example, the cancellous bone inside the epiphysis is composed of a complicated trabecular network in the bone marrow, showing strongly anisotropic and heterogeneous structure. We have then observed the longitudinal wave propagation in this complicated medium, paying special attention to the two wave phenomenon. The fast wave, which mainly propagates in the trabecular part of the cancellous bone, is more sensitive to the network structure. With the help of x-ray micro-computed tomography, three dimensional anisotropic trabecular structure and fast wave propagation were investigated using bovine cancellous bones. We can find the strong effect of anisotropic trabecular structure on the fast wave velocity. The mean intercept length (the information of average trabecular length) was a good parameter to describe the three dimensional trabecular structure and showed good correlation with the fast wave velocity.

Contributed Papers

1:45

Different quantitative ultrasound techniques are currently developed for clinical assessment of human bone status. This paper is dedicated to axial transmission: emitter(s) and receiver(s) are linearly arranged on the same side of the skeletal site. A multiparameter approach might be relevant to improve bone diagnosis and this be could achieved by accurate measurement of guided wave phase velocities. Clinical requirements and bone/soft tissue heterogeneities constrain the length probe to about 10 mm. Thus efficiency of conventional spatio-temporal Fourier transform is reduced. Signal processing to obtain reliable guided wave velocities is a key point. Here the guided mode phase velocities are obtained using a projection in the singular vectors basis determined by the singular values decomposition of the transmission matrix between the two arrays. This method has been first validated on metallic plates. A set of excised human radius were then probed. Attention was paid to observation of cut-off frequencies because of their high power of discrimination of relevant bone properties (thickness and elasticity). The analysis of the whole spectrum including several branches of intermediate phase velocities is under progress. The so-called bi-directional correction of the soft tissue thickness variations under the probe is used for the in vivo measurements.

2:00
5pBB5. Discrimination of bone fractures by low-frequency axial transmission velocity in the radius and tibia. Petro Moilanen (Dept. of Phys., Univ. of Jyväskylä, P.O. Box 35, 40014 Jyväskylä, Finland, petro.moilanen@jyu.fi), Mikko Määttä (Univ. of Oulu, 90014 Oulu, Finland), Vantte Kilappa, Leiting Xu (Univ. of Jyväskylä, 40014 Jyväskylä, Finland), Timo Jämsä (Univ. of Oulu, 90014 Oulu, Finland), Jussi Timonen, and Sulin Cheng (Univ. of Jyväskylä, 40014 Jyväskylä, Finland)

Low-frequency axial transmission velocity of the first arriving signal \(V_{LF}\) has a wavelength (8–10 mm) sufficient to probe osteoporotic changes in subcortical bone. Purpose of the present study was to evaluate \(V_{LF}\) on retrospective fracture discrimination, using a custom-made ultrasonometer in the midshaft radius and tibia. Preliminary data for 49 non-fractured (NF; 45–87 years) and 16 fractured (F; 56–81 years) postmenopausal females were analyzed. The fractures included were caused by low or moderate trauma and mostly occurred in the forearms or lower legs. Subj ects with disease or medication affecting bone metabolism were excluded. Areas under receiver operating characteristic curve were 0.76 for the radius and 0.60 for the tibia \(V_{LF}\). When adjusted for age and BMI, odds ratio for the radius \(V_{LF}\) was 1.97 (95% CI: 1.11–3.48), while that for the tibia \(V_{LF}\) was not statistically significant. In the contrary, dual-energy x-ray absorptiometry (DXA; whole-body, femoral neck, and L2–L4) was unable to differentiate between the F and NF groups. Despite the small study population, these results suggest that \(V_{LF}\) in the radius discriminates osteoporotic fractures in cases of peripheral fractures, in particular, for which DXA has a limited sensitivity.

2:15
5pBB6. Decomposition of two-component pulses: Simulation and phantom experiment. Keith A Wear (FDA Ctr. for Devices and Radiological Health, Bldg. 62, Rm. 3108, 10993 New Hampshire Ave., Silver Spring, MD 20993)

Porous media often support propagation of two compressional waves. When cancellous bone samples are interrogated in through-transmission with broadband sources, these two waves often overlap in time. A method for measuring attenuation and velocity of the two component waves was developed. The method (1) assumes that the transfer function is the sum of two exponentially modulated sinuosoids and (2) minimizes the sum of the squared error between a model fit and the data. The method was tested for decomposing a 500-kHz-centre-frequency signal containing two overlapping components: one passing through a polycarbonate plate (fast wave) and another passing through a cancellous-bone-minicking phantom (slow wave). The method yielded estimates of attenuation slopes accurate to within 7% (polycarbonate plate) and 2% (cancellous bone phantom). The method yielded estimates of phase velocities accurate to within 1.5% (both media). The method was tested on simulated data generated using attenuation slopes and phase velocities corresponding to bovine cancellous bone. Throughout broad ranges of signal-to-noise ratio and fast-slow-wave-velocity differential, the method yielded estimates of attenuation slope that were accurate to within 10% and estimates of phase velocity that were accurate to within 5% (fast wave) and 2% (slow wave).

2:30
5pBB7. Recovery of structural degradation indicators of human bones using elastic waves. Erick Ogam, Zine El Abeddine Fellah (Laboratoire de Mcanique et d’Acoustique, UPR 7051 CNRS, 31 chemin Joseph Aiguier, Marseille 13402, France), Jean-Philippe Groby (Universit du Maine, 72085 Le Mans Cedex 09, France), Robert Gilbert (Univ. of Delaware, Newark, DE 19716), and Armand Wirgin (Laboratoire de Mcanique et d’Acoustique, Marseille 13402, France)

Non-ionizing techniques of bone characterization have mostly been based on ultrasonic wave propagation. In this study, bone is characterized via its linear response to vibratory solicitations. The analysis of the response of bones to transient mechanical excitation is made both theoretically and experimentally. An orthotropic three-dimensional finite element interaction model (FEIM) using computerized tomography scan image geometries of excised dry human trabecular bones is developed. The model parameters (nine elastic constants) are recovered by solving an inverse eigenvalue problem using resonance frequencies from vibration spectroscopy. Transient waves are excited and signals acquired along the diaphysis of the tibiae, experimentally by employing piezoelectric transducers and numerically using FEIM and a time integration scheme. The elastic waves propagating as
The effective properties of a two-phase composite depends both on the constituent materials and the microstructure. Starting from David J. Bergman’s work on the integral representation formula (IRF) approach for deriving bounds on dielectric properties of composites with isotropic constituents, followed by Golden & Papanicolaou’s work on establishing the link between IRF and spectral theory, the IRF is on now a firm mathematical background for further generalization from isotropic constituents to general constituents. The key feature we are looking for is the separation of information on material contrast from that of microstructure in the IRF. This is especially important for (visco)elastic composites because their current available IRFs do not have this feature, even when constituents are isotropic. In this talk, we will present a recently derived IRF for general (visco)elastic composites with the above feature. The relation between IRF and the quantification of microstructure (as spectrum of a generalized Helmholtz projection operator) will be explained. If time permits, numerical results will be presented and the connection between the IRF and the fabric tensor introduced by Stephen C. Cowin will also be discussed.

3:00—3:30 Break

Invited Papers

3:30

5pBB9. Multiscale structure-functional modeling of lamellar bone. Kay Raum (Julius Wolff Institut, & Berlin-Brandenburg School for Regenerative Therapies, Charité Universitätsmedizin Berlin, Augustenburger Platz 1, 13353 Berlin, Germany, kay.raum@charite.de), Quentin Grimal (Universités Pierre et Marie Curie-Paris 6, F75006 Paris, France and CNRS, F75006 Paris, France), and Alf Gerisch (Universität Darmstadt, Darmstadt, Germany)

Bone is a natural example of achieving a unique combination and variability of stiffness and strength. One of the striking features of bone tissue is the ability to adapt to variable loading conditions by multiple but well organized structural arrangements of mineralized collagen fibrils at several levels of hierarchical organization. A profound understanding of the structure-function relations in bone requires both experimental assessment of heterogeneous elastic and structural parameters and theoretical modeling of the elastic deformation behavior. A bottom-up approach for experimental assessment and numerical modeling of the hierarchical structure from the nanoscale to the macroscale will be presented. Experimental data are obtained by scanning acoustic microscopy between 50 MHz and 1.2 GHz and provide anisotropic elastic and structural information at the lamellar (nanoscale) and at the tissue matrix (microscale) level. These data are directly translated into a finite element mesh. By numerical deformation analyses, the homogenized elastic stiffness tensor of the next hierarchical levels (micromodel to macroscale) is derived. At each level the numerical results are cross-validated by experimental data. It will be shown that local variations in elastic anisotropy within the femoral shaft are related to an inhomogeneous strain distribution resulting from external stresses by weight and muscle forces.

3:45

5pBB10. Role of absorption mechanisms for ultrasound attenuation in cancellous bone: Macroscopic modeling and experiment. Michal Pakula and Mariusz Kaczmarek (Inst. of Mech. and Comp. Sci., Kazimierz Wielki Univ., Kopernika 1, Bydgoszcz, Poland, michalp@ukw.edu.pl)

Evaluation of the relative contribution of physical mechanisms responsible for attenuation of ultrasonic wave in cancellous bone is one of the crucial issues from the point of view of modeling elastic wave propagation and related model-based identification of the structural and mechanical properties of bone material. Considering trabecular bone as a porous material filled with fluid, the wave attenuation may stem from (i) intrinsic absorption in the fluid and solid phase, (ii) friction at the fluid-solid interface, as well as (iii) wave scattering by inhomogeneities (pores/trabeculae). The commonly used for modeling ultrasound propagation in cancellous bone of the macroscopic Biot’s theory will be discussed in context of its potential applicability for prediction of wave parameters: phase velocity and attenuation coefficient as functions of frequency. Since the model was introduced for long wavelength range, the scattering effects are neglected, and the analysis will be focused on the absorption mechanisms responsible for attenuation of ultrasonic waves in bone material. The suitability of the model will be verified by comparison of results of sensitivity analysis of the model with in vitro experimental ultrasonic data obtained for cancellous bones filled with different fluids.

Contributed Papers

4:00

5pBB11. Characterization of cortical bone fracture with scanning confocal ultrasound and longitudinal acoustic velocity. Yi-Xian Qin, Jiqi Cheng, Suzanne Ferreri, and Wei Lin (Dept. of Biomedical Eng., Stony Brook Univ., 350 Psych.-A Bldg., Stony Brook, NY 11794, yi-xian.qin@sunysb.edu)

Non-invasive assessment of fracture, particularly in non-typical fracture, is a critical health problem. As a promising alternative to the x-ray, ultrasound has demonstrated potentials in early fracture diagnosis. A real-time scanning confocal ultrasound image was developed to evaluate bone defect and bone loss. The objectives of this study were to evaluate the cortical fracture gap size using quantitative ultrasound imaging and the longitudinal ultrasound velocity in bone to predict the fracture gap size. The total fractures were created by MTE compression at the middle diaphysis, with gap size varied from 1–5 mm (N=4). Fractures were tested with the ultrasound scanning with 0.5-mm resolution. The measured ultrasound signals were analyzed to calculate BUA (dB/MHz), ATT (dB), and transverse velocity (m/s). The longitudinal wave velocity was tested using three surface transducers. Strong correlations were observed between ultrasound and x-ray images in fracture size (R2=0.91). High correlation was found between gap size and the longitudinal velocity from 4000 to 3000-m/s for 0.5–5-mm gaps (R2=0.93). These results suggest that ultrasound is capable to predict bone fracture
and provide useful information for longitudinal assessment of complications, such as non-union fracture, and for evaluating healing. [Work supported by NSBRI/NASA and NIH.]

4:15 5pBB12. Dynamic acoustoelastic testing for noninvasive detection of microdamage in cortical tissue of long bones. Guillaume Renaud, Maryline Talmant, and Pascal Laugier (Laboratoire d’Imagerie Paramétrique, CNRS UMR7623, Université Pierre et Marie Curie, 15 rue de l’École de médecine, 75006 Paris, France, renaud_gu@yahoo.fr)

First developed for trabecular bone, dynamic acoustoelastic testing (DAT) is based on the measurement of ultrasound (US) pulses time of flight and energy modulations (TOFM and EMs) induced by a low-frequency (LF) acoustic wave, synchronously injected in the probed medium. The technique needs the LF wave to be quasi-uniform in the investigated region and quasi-static in time (compared with the US TOF). US TOFM and EM are related to elastic and dissipative nonlinearities, respectively. Here is presented the application of the technique to cortical tissue of long bones. A LF vibration is generated along the axis of the sample whose frequency corresponds to the first compressional resonance mode. Simultaneously, US pulses are emitted and received by a dedicated probe, after propagation along the axis of the sample. The emitter-receiver distance is chosen and the position of the US probe is adjusted so that the head wave arrives first in time. US TOFM and EM are thus computed using the head wave. After validation in an aluminum cylinder where no dissipative nonlinearity but weak quadratic elastic nonlinearity is observed, in vitro results are presented for bone. DAT is performed on diaphyseal bone sections before and after mechanical damage was induced.

4:30 5pBB13. Bone quality assessment at multiple skeletal sites using quantitative ultrasound imaging. Frederick Serra-Hsu, Jiqi Cheng, Wei Lin, and Yi-Xian Qin (Dept. of Biomed Eng., SUNY Stony Brook, HSC T18, Rm. 030, Stony Brook, NY 11794, fserrahs@ic.sunysb.edu)

Osteoporosis is responsible for nearly 1.5 × 10^8 fractures annually in U.S. with the most typical sites at the hip, spine, and wrist. Early diagnosis is essential for the prediction of fracture risk. While BMD only represents quantitative information of bone, current quantitative ultrasound (QUS) typically employs index using attenuation (BUA) and velocity (UV) measurements at the calcaneus. The objective of this study aims to develop an image based QUS technology for enhanced diagnostic readings at multiple anatomical sites. With a newly developed scanning confocal acoustic diagnostic system, five embalmed cadaver forearms were disarticulated from the wrist joint, cut at the mid shaft, leaving the interosseous membrane intact, and scanned by the ultrasound. Acoustic phantoms scanned with the new system and basic water coupling show low variation (<7% of mean BUA) with highly linear BUA curves (mean R^2=0.96). BUA and UV were analyzed at the distal radius using existing algorithms; however, the distal radius geometry provides much different BUA and UV schemes than the calcaneus so site-specific calculation algorithms are needed. DXA, mCT, and mechanical testing are currently under evaluation. These preliminary data demonstrated feasibility of ultrasound imaging in critical skeletal site, i.e., wrist. [Work supported by NSBRI/NASA and NIH.]

4:45 5pBB14. Separating individual modes from multimodal guided wave signals in long cortical bones. K. L. Xu, D. A. Ta (Dept. of Electron. Eng, Fudan Univ., Shanghai 200433, China), P. Moilanen (Univ. of Jyväskylä, Finland), and W. Q. Wang (Fudan Univ., Shanghai 200433, China)

Time-frequency representation (TFR) is one of the classical methods used for guided wave signal processing in assessment of long cortical bones. In this paper, we first used a crazy-climber ridge detection method to separate individual modes from multimodal guided wave signals in the time-frequency domain. Second, a penalization algorithm, based on spline smoothing, was employed to reconstruct the individual modes separately in the time domain. Robustness of these methods was evaluated by analyzing highly noisy simulated signals and experimental signals on bovine tibiae. It was found that the separated TFR ridges, provided by the crazy-climber algorithm, were clearer than using the TFR alone and these corresponding trajectories were in good agreement with theoretical dispersion curves. The penalization algorithm, on the other hand, showed promising performance by yielding good accordance between the reconstructed individual modes and the theoretical predictions. Both the crazy-climber and the penalization algorithms were thus useful for TFR analysis of multimodal guided wave signals, even under strongly noisy conditions, by separating and reconstructing the individual contributions in the time domain. These methods can thus significantly improve ultrasonic guided wave signal processing by providing a feasible quantitative assessment of individual modes in temporal multimode signals, recorded in long bones.
Session 5pEA

Engineering Acoustics, Acoustical Oceanography, and Underwater Acoustics: Computer Modeling for Complex Acoustic Environments

Kenneth M. Walsh, Chair
K. M. Engineering, Ltd., 51 Bayberry Ln., Middletown, RI 02842

Chair’s Introduction—1:00

Invited Papers

1:05


Synthetic aperture processing is applied to seabed reflection data to improve images of stratified sediment layering. Vertical profiles of sediments are limited in dynamic range by sediment scattering. Synthetic aperture processing, which increases the effectiveness along track length of the sonar transmitter and receiver, suppresses scattering noise, thereby allowing detection of stratified layering with weaker reflection coefficients. The image improvement resulting from the application of synthetic aperture processing is determined by comparing reflection profiles generated with and without synthetic aperture processing. The processing algorithm used in the analyses is not the conventional synthetic aperture processing algorithm which is coherent for point scatterers, but is an algorithm that provides coherent processing for planar reflectors.

1:25

5pEA2. Fast array processor for the noiselet reverberation model. Edmund J. Sullivan, Robert P. Goddard (Prometheus Inc., 21 Arnold Ave., Newport, RI 02840, ed@prometheus-us.com), Hyman A. Greenbaum (Information Technol., Middletown, RI 02842), and Carlos Godoy (Naval Undersea Warfare Ctr., Newport, RI 02841)

The noiselet reverberation model, instead of transmitting the pulse itself, transmits a superposition of many copies of the pulse, each copy with a random amplitude and phase. This allows the simulation to mimic the point scatterer method without requiring a prohibitively large number of scatterers. It uses the GRAB ray model to introduce the propagation information. It provides a real-time high-fidelity simulation, providing element-level time series as an output for a planar array of receivers. However, it is presently limited by the phase shift calculations required by the frequency-domain beamformer to about 20 receivers without an unacceptable trimming of the number of rays used. Here, an approach is demonstrated that significantly improves the speed of this computation by identifying those receivers requiring the same phase shift to within a given specified error, such that only a single phase shift computation is required. These receivers fall onto a finite width strip, or quasi stave, that is not required to be parallel to element rows or columns. Along with providing a significant speed-up to the calculation, the approach becomes more efficient as the number of receivers increases, thereby allowing real time simulations using receiver arrays of over 100 elements.

1:45


Multidomain modeling, recently also called multiphysics modeling, is appropriate for systems in which different parts of the system are modeled using different equations and different modeling variables. Each such domain is modeled using the variables and equations that are appropriate for it, and the domains are coupled by additional equations at their boundaries and in regions of overlap. Acoustic transducers are an obvious example of multidomain systems. Historically, electrical circuit analog modeling was an approach for multidomain analysis. In addition, specific domain couplings have been hard-coded into particular FEA codes for the analysis of some kinds of coupled systems. Today there are FEA codes, as well as codes based on simpler ordinary differential and algebraic equation analysis, that provide a robust framework for this kind of modeling. This paper will discuss these analysis methods and provide computational examples from some simple multidomain systems that include acoustic transducers. [Work supported in part by the Office of Naval Research.]

2:05

5pEA4. Ducted propagation in the atmosphere, from audible sound to infrasound. Roger Waxler, Carrick Talmadge, Kenneth Gilbert, Xiao Di, Phillip Blom, Jelle Assink, and Claus Hetzer (NCPA, Univ. of Mississippi, University, MS 38677)

Earth’s atmosphere supports complicated and highly variable winds and temperature gradients. Temperature inversions and wind jets provide ducts in which sound can propagate efficiently to great distances from the source. Due to the intrinsic variability of the atmospheric winds, these sound ducts are also quite variable, both temporally and spatially. Further, the direction of the atmospheric winds...
can change dramatically with increasing altitude so that the behavior of the propagated field can become quite complex as the range from the source increases. Despite this, some systematic behavior has been observed in long range sound propagation. An overview will be presented, with an emphasis on the systematic behavior. The current state of our understanding of the sources and consequences of the atmosphere’s variability will be discussed.

Contributed Papers

2:25

5pEA5. Atmospheric sound scattering model to test signal coding methods for acoustic wind profiling. Paul Kendrick and Sabine von Hünerbein (School of Computing, Sci. and Eng., Univ. of Salford, Salford M5 4WT, United Kingdom, p.kendrick@salford.ac.uk)

This presentation concerns simulation of atmospheric sound propagation, including the backscatter of acoustic waves from moving temperature fluctuations. The model has been developed to evaluate the performance of acoustic wind speed profiling techniques, in particular, the use of pulse coding methods which aim to increase range resolution, accuracy, and data availability. Stepped frequency chirps are propagated through the model and the results used to optimize matched filtering algorithms. Results from simulations are presented and the performance of several different signal detection algorithms evaluated. This methodology provides insight into particular problems encountered by the signal detection algorithm and will be utilized to aid the design of an experimental acoustic wind speed profiler. [Work funded by the Engineering and Physical Sciences Research Council (EPSRC).]

2:40

5pEA6. A finite difference model for low frequency sound measurement in kraft recovery boilers. Robert Hildebrand (Lake Superior State Univ., 650 W. Easterday Ave., Sault Ste. Marie, MI 49783, rhlhildebrand@lssu.edu), Matthew Carroll (Texas A & M Univ. at Galveston, Galveston, TX 77553-1675), Ville Järvinen, and Juha Miettinen (Tampere Univ. of Technol., Tampere, Finland)

Kraft recovery boilers greatly enhance the efficiency of paper pulp mills by burning the organic wastes recovered from the pulp making process and generating electricity, but significant smelt-water explosion hazards exist when steam is used as the working fluid due to sodium compounds in the sludge from the waste incineration process. Current practice is to use acoustic emission sensors to monitor for leaks in the bottom wall and to initiate an automatic shutdown sequence for the boiler once a leak is detected. A high-frequency model previously developed by the authors and verified by measurements performed in Skoghall, Sweden indicated strong attenuation in the sludge between the point of leakage and the detectors, but experimental results demonstrated a pass band at low frequencies (under 20 kHz). In this paper the authors develop a two-degree-of-freedom finite difference model for the prediction of acoustic attenuation at low frequencies. The bottom wall is modeled as a periodic tube and fin arrangement, and a time-domain finite difference method is used whereby the fin and tube are simple lumped elements. The model is then compared to previous experimental results, with which excellent agreement is obtained.

2:55


Nonlinear acoustic interactions such as acoustic radiation forces or acoustic streaming require high-amplitude acoustic pressure or velocity terms to generate observable effects. While at the macro-scale these effects are not widely used; due to the highly compliant microstructures, high-amplitude acoustic pressure and velocity terms have been reached in a microscopic cavity even at very low-voltage drives (1–10 Vpp). Here, vibrational and acoustic modes of a thick-walled fluid-filled hollow cylindrical glass capillary are introduced. Vibrations are coupled to the capillary through a laser cut, C-shaped lead zirconate titanate plate. Mode shapes and dispersion relationships are obtained both analytically and from finite element modeling in ANSYS. Analytical modeling of the microcavity resonator shows excellent agreement with the computational models developed in ANSYS and the experimental results. Coupling of the vibrational modes of the capillary to the fluid enclosed plays an essential role in manipulation and separation of micro- and nanoparticles and biological entities in acoustically excited portable microfluidic platforms. In this regard, proper modeling of the vibroacoustic modes enables calculation of acoustic radiation force field generated inside the microfluidic capillary. Experimental results on particle collection and separation will be presented at the meeting.

3:10

5pEA8. Shock wave formation on turbulent coanda surfaces. Thomas Dowd (Dept. of Mathematics, James Madison Univ., 800 S Main St., Harrisonburg, VA 22807, dowd.thomas@gmail.com)

In the petroleum industry a procedure called flaring is used, which is the burning off of unwanted gas. Modern day flares use the Coanda effect, which states “when a jet of fluid is passed over a curved surface it bends to follow the surface entraining large amounts of air” to achieve smokeless combustion, increased combustion efficiency, and decreased thermal radiation due to the entrainment of large amounts of air. These advantages are at the cost of increased noise pollution, due to shock wave formation caused by the difference in the nozzle exit pressure and the ambient pressure. Sound called shock-associated noise is generated by the interaction of downstream propagating turbulent eddies and the stationary quasi-periodic shock-cell structure contained in the supersonic jet. A model of shock-associated noise near turbulent Coanda surfaces will be presented and compared with experimental data along with suggestions for reducing noise. Also a visualization of these shock wave formations and their intricacies will be presented if time allows.

3:25


While receiving less attention in the literature than in the field of electromagnetic scattering, theoretical efforts to define and create acoustic skins capable of reducing scattered energy from obstacles by way of mimicking coordinate transformations through use of meta-materials have begun. The present work extends recent analysis of Norris by considering a variety of acoustic skins, from those comprised of fluid layers which are isotropic in bulk moduli with anisotropic density to those having anisotropic bulk moduli and isotropic density. In all but pure inertial types, fluid layers comprising the skins are pentamode materials governed by a special scalar acoustic equation for pseudo-pressure derived by Norris. In most cases presented, material properties of the fluid/pentamode layers are based upon target values specified by continuously varying properties resulting from theoretical coordinate transformations geared to minimize scattered pressure limited by realistic goals. The present work analyzes acoustic skins for the specific case of plane wave scattering from an acoustically hard sphere. An initial exploration of the parameter space defining such skins (for example, material properties of their constituent layers and operating frequency) is undertaken with a view toward “optimal” design. [Work supported by the NAVSEA Newport In-House Laboratory Individual Research (ILIR) Program.]
Power consumption is one of the major design considerations for recent computing systems. MP3 players require MP3 decoding performed on the processor. We analyze the power consumption for MP3 decoding varying the processor type [central processing unit (CPU) and graphics processing unit (GPU)]. In general, the MP3 decoding is performed on CPU. GPU, which is a specialized processor that offloads three dimensional graphics rendering from the CPU, can be used for MP3 decoding since it is used for various computations as well as graphics rendering. At first, we analyze the power consumption for the MP3 decoding performed on CPU by using Inspector tool. And then, we analyze the power consumption for the MP3 decoding performed on GPU. To analyze the power efficiency of the GPU for MP3 decoding, we use CUDA for translating the MP3 decoding process from CPU to GPU. Computer unified device architecture (CUDA) is the computing engine in NVIDIA GPUs that is accessible to software developers through industry standard programming languages. According to our experimental results, the MP3 decoding performed on GPU saves power consumption by 23% compared to that on CPU. [Work supported by the MKE, Korea, under the ITRC supervised by the IITA (IITA-2009-00903-0008)].

Session 5pPAa

Physical Acoustics: Outdoor Sound Propagation

D. Keith Wilson, Cochair
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W. C. Kirkpatrick Alberts, Cochair
U.S. Army Research Lab., 2800 Powder Mill Rd., Adelphi, MD 20783

Contributed Papers

5p PAa1. Ground effects in time-domain simulations of outdoor sound propagation. Didier Dragna, Philippe Blanc-Benon (LMFA, UMR CNRS 5509, Ecole Centrale de Lyon, 69134 Ecully Cedex, France; didier.dragna@ec-lyon.fr), and Franck Poisson (SNCF, Direction de l’Innovation et de la Recherche, 75379 Paris Cedex 08, France)

Finite-difference, time-domain (FDTD) methods are precise and powerful tools to solve linearized Euler equations. Interaction between the acoustic waves with local wind or temperature gradients as well as a complex topography can be taken into account. Recently, a time-domain boundary condition has been proposed [Cottet et al., AIAA J. 47, 2391–2403 (2009)] and has been implemented in a FDTD solver using methods developed in the computational aeroacoustics community [Bogey and Bailly, J. Comput. Phys. 194, 194–214 (2004)]. Propagation of an initial pulse over a distance of 500 m in a two-dimensional geometry is considered in a frequency band up to 1200 Hz. Surface waves which propagate close to and parallel to impedance grounds are exhibited. To validate the obtained results, a comparison is made in the time-domain with an analytical solution. Ground effects are discussed depending on the impedance of the ground. [This work was performed using HPC resources from GENCI-IDRIS (Grant 2009-022203).]


The Crank–Nicholson parabolic equation (CNPE) method was used to calculate speech intelligibility for long range propagation of voice messages. How intelligible a voice message will be at long range is very sensitive to atmospheric conditions and parameters such as source height and background noise levels. In order to identify this sensitivity, four independent variables were identified and varied throughout the study. These independent variables are the atmospheric refraction index, which takes into account both temperature and wind gradients within the atmosphere, ambient temperature, source height, and background noise level. CNPE computations were performed for 31 optimally chosen cases where all four of the independent variables were random variables and were inputted into a metamodel. This allows for curve fitting and extrapolation of the existing data to determine intermediate results for any operation scenario defined within the bounds of the current study. Upon doing this, it was determined that the atmospheric refraction index is the most sensitive variable. In addition, there is virtually no dependence on ambient surface temperature.

5p PAa3. Wind velocity measurements with acoustic daylight. Oleg A. Godin (Cires, Univ. of Colorado and NOAA/Earth System Res. Lab., Boulder, CO 80305, oleg.godin@noaa.gov), Vladimir G. Irisov, and Mikhail I. Charnotskii (ZelTechnologies LLC and NOAA/Earth System Res. Lab., Boulder, CO 80305-3328)

Ambient acoustic noise in ocean and atmosphere provides acoustic illumination, which can be used, akin to daylight, to visualize objects and characterize the environment [Buckingham et al., Nature(London) 356, 327–329 (1992)]. It has been shown theoretically [O. A. Godin, Phys. Rev. Lett. 97, 054301 (2006)] that deterministic travel times of acoustic waves propagating in opposite directions between two points in an inhomogeneous, moving, time-independent medium can be retrieved from the cross-correlation function of non-diffuse acoustic noise recorded at the two points. Thus, the noise cross-correlation function contains information on non-reciprocity of sound propagation, which can be utilized to measure flow velocities small compared to the sound speed. This paper presents an experimental verification of the theoretical predictions. Deterministic travel times have been retrieved from cross-correlations of noise recorded at three microphones about 20 m apart and inverted for the sound speed and two horizontal components of the wind velocity. Traffic noise served as the acoustic daylight in the experiment. Accuracy of the passive acoustic measurements of sound speed and wind velocity has been confirmed by comparison with contact measurements of wind, air temperature, and humidity. Possible uses of acoustic daylight for a low-cost environmental monitoring will be discussed.
Large wind noise reduction outdoors can be achieved by the use of metal foam layers mounted flush with the surface. In this research, a triangular array for source direction determination is mounted under different thickness metal foams. The performance of the array in locating a spark source is determined by measurements with and without metal foam covers. The performance is quantified by agreement with the actual direction and by the chi squared values of the fit to the direction and the speed of sound. Some geometries produced large errors in speed of sound but the corresponding directions were reasonably accurate. Relatively thin layers of metal foam produced accurate estimates for the direction and sound speed. With the microphones mounted just under a 0.5-in.-thick foam, the direction could be measured to within a few degrees. [This work is funded by the Armaments Research, Development, and Engineering Command (ARDEC)].

2:00
5pPAa5. Gunshot localization using distributed acoustic sensors. Xiao Di, W. Garth Frazier, and Kenneth E. Gilbert (Nat'l. Ctr. for Physical Acoust., Univ. of Mississippi, University, MS 38677)

For detecting and localizing gunshot sound, distributed acoustic sensors with various baseline separations (several meters to dozens of meters) are used in our field experiments. The large baselines of the distributed sensor networks, based on the time difference of arrival method, provide improved localization accuracy and have distinct advantages over small arrays. Results are reported and compared for various baseline separations, detection distances, numbers of sensors, and weather conditions. The sound source used is a propane cannon. Several least squares methods are used in the data analysis and source localization. Results from the experiment are presented and discussed. The localization errors are analyzed as a function of combined error sources (GPS, timing, weather, etc.). Possible ways to improve the localization accuracy are suggested. [Research supported by the U. S. Army TACOM-ARDEC at Picatinny Arsenal, NJ.]

2:15
5pPAa6. Investigation of dominant sources of infrasonic wind noise in ground level microphones. John Paul R. Abbott, Richard Raspet, Jeremy Webster, and Jiao Yu (Dept. of Phys. and Astronomy, Natl. Ctr. for Physical Acoust, Univ. of Mississippi, University, MS 38677)

A two part project has investigated whether the pressure fluctuations (infrasonic wind noise) at the ground, as measured in an open field, can be predicted from the vertical wind velocity measured above the ground. The dominant region contributing to the pressure on the ground is calculated theoretically and displayed with contour plots. Experimentally, time dependent correlations between the measured pressure and wind velocities were done. The correlation coefficients were very low, typically between 0.08–0.15, for broadband frequencies up to 5 Hz. Cross spectrum analysis and application of octave band pass filters showed the dominant frequencies for correlations to be around 0.07–0.1 Hz, with correlation coefficients typically between 0.3–0.5. [Project funded by the US Army Space and Missile Defense Command.]
Session 5pPAb

Physical Acoustics: Ultrasonics

Andrea Prosperietti, Cochair
Johns Hopkins Univ., Dept. of Mechanical Engineering, 3400 N. Charles St., Baltimore, MD 21218

Andrew A. Piacsek, Cochair
Central Washington Univ., Dept. of Physics, 400 E. University Way, Ellensburg, WA 98926-7422

Contributed Papers

1:00
5pPAb1. Interaction of shear horizontal ultrasonic waves with a surface grating: Analytical and experimental results. Tony Valier-Brasier (Lab. d’Acoustique de l’Univ. du Maine (LAUM), UMR CNRS 6613, Ave. Olivier Messiaen, 72000 Le Mans Cedex 09, France, tony.valier-brasier.etu@univ-lemans.fr), Damien Leduc (University du Havre, 76610 Le Havre, France), Catherine Potel (Univ. du Maine, 72000 Le Mans Cedex 09, France), Bruno Morvan (Univ. du Havre, 76610 Le Havre, France), Michel Bruneau (Univ. du Maine, 72000 Le Mans Cedex 09, France), and Jean-Louis Izbioki (Univ. du Havre, 76610 Le Havre, France)

This paper deals with the propagation of shear horizontal (SH) ultrasonic waves on a plate with one surface grating. The theoretical acoustic field in the corrugated plate is obtained from a linear combination of the Neumann eigenmodes of a plane plate waveguide which bounds outwardly the perturbed surface of the plate considered. Coupling of the eigenmodes with the rough walls is studied using the integral formulation. The effect of the roughness is expressed in such a way that two intermodal coupling mechanisms are highlighted: a bulk coupling and a surface coupling. The first one depends only on the depth of the roughness and the second one on both its depth and its slope. Moreover, the coupling between SH modes is studied experimentally. An electromagnetic acoustic transducer enables a non-contact emission of the SH waves. Notably, the energy transfer from a mode to others so appears to be strongest at frequencies where a relation between the period of the grating and the wavenumbers of the coupled SH modes is verified.

1:15
5pPAb2. High-quality acoustic cavities based on radial sonic crystals. Daniel Torrent (Dept. Electron Eng., Polytechnic Univ. of Valencia, datorma1@upvnet.upv.es) and José Sanchez-Dehesa (Polytechnic Univ. of Valencia, Spain.)

Radial sonic crystals are radially inhomogeneous systems that present a dispersion relation similar to one dimensional sonic crystals, with the peculiarity of having low-frequency band gaps. These structures are here employed to design high-quality circularly symmetric acoustic cavities. The presence of band gaps in their dispersion relation allows to localize sound in the cavity centers, leading to strong resonances whose frequency depends on the geometry of the field. Therefore, it is possible to design cavities with monopolar, dipolar, quadrupolar, etc., symmetry. A proposal for its physical realization is also described, showing that radial sonic crystals are possible to build by using the concept of acoustic metamaterials. [Work supported by MCIIN of Spain and ONR.]

1:30
5pPAb3. Ultrasonic characterization of fluid saturated absorbing polymethylacrylate porous plates. Serge Derible (LOMC GOA, FRE CNRS 3102, FANO FR CNRS 3110, Univ. Le Havre, 76600 Le Havre, France), Pierre Campistron, Georges Freiha (Univ. de Valenciennes et du Hainaut Cambrésis, 59313 Valenciennes, France.), Hervé Franklin (Univ. Le Havre, 76600 Le Havre, France), and Bertrand Nongaillard (Univ. Valenciennes et du Hainaut Cambrésis, 59313 Valenciennes, France)

Experiments are led on five manufactured porous polymethylmethacrylate (PMMA) samples, numbered 1–5. Their constituting materials are packed beads whose diameters run from 200–300 µm for plate 1 up to 600–700 µm for plate 5. The plates are normally insonified by wide band transducers acting within the frequency range 100–800 kHz. The samples obey Biot’s theory taking into account the viscoelasticity of the solid frame. The reflection and transmission coefficients are measured and then added or subtracted to give the two transition terms, issued from S-matrix theory. They are connected to the symmetrical or asymmetrical motions of the faces of the plates. Those terms allow us a characterization of the porous media by means of a computer program that seeks the values of the Biot parameters giving the best agreement between theory and experiments.

1:45
5pPAb4. Evaluation of surface waves in bone-conducted sound. Margaret Wisner, William O’Brien (BRL, Univ. of Illinois, Urbana, IL), Odile Clavier, and Anthony Dietz (Creare, Hanover, NH)

In order to better understand the basic physical principles of bone-conducted sound, the skull is approximated as a thin spherical fluid-filled shell. Results for sound waves propagating through and around this structure have been determined for a range of frequencies, media, and shell sizes. Since the early 1950s researchers have studied the problem of what happens when waves hit solid targets. Most of the research is concerned with the back-scatter signal but recognize that the interaction of the wave with the object influences these results. For the bone-conducted sound study, the interaction between the incident signal and the object is the critical information. For acoustic noise interacting with the human skull, the ka is below 5. The structure supports the existence of surface waves as the predominant mechanism of energy transfer within the skull. Multiple types of surface wave for this structure of a thin, curved fluid-solid interface exist. Signals which propagate on the air side of the surface due to the curvature are known as Franz or Stoneley waves. Since the shell is thin compared to the typical wavelength of sound, waves within the shell are lower order Lamb or plate waves. All of these waves are dispersive in nature with group speeds considerably smaller than the speed of sound in bone.

2:00

Tissue heating by high-intensity focused ultrasound (HIFU) is a promising modality for minimally invasive therapy. However, real-time treatment monitoring still poses significant challenges, particularly at the lower exposure levels where stable cavitation and/or boiling does not result. Bubble-
-free HIFU lesions offer little acoustic contrast; however, one does observe significant contrast in both optical scattering and absorption. We employ acousto-optic (AO) imaging to sense, in real time, optical changes induced by lesion formation. By using a transducer to simultaneously heat a tissue volume and pump the AO interaction, lesions generated in excised chicken breast are monitored in real time. The change in AO response with time is linearly related to the time-dependent lesion volume, provided the diameter of the lesion does not exceed the width of the optical beam. Therefore, AO sensing can be used to both determine the onset of lesion formation and the resulting volume of the necrosed region. The feasibility of using the observed change AO signal amplitude as the criteria to guide HIFU exposure in real time is demonstrated. [Work supported by the Center for Subsurface Sensing and Imaging Systems, NSF ERC Award No. EEC-9986821.]

FRIDAY AFTERNOON, 23 APRIL 2010

Session 5pSC

Speech Communication: Machine Learning Techniques for Speech Recognition

Mark A. Hasegawa-Johnson, Chair

Univ. of Illinois, ECE, 405 N. Matthews, Urbana, IL 61801

Chair's Introduction—1:00

Invited Papers

1:05

5pSC1. A novel algorithm for sparse classification. Sujeeth Bharadwaj and Mark Hasegawa-Johnson (Beckman Inst., Univ. of Illinois at Urbana-Champaign, sbhara3@illinois.edu)

A recent result in compressed sensing (CS) makes it possible to perform non-parametric speech recognition that is robust to noise and that requires few training examples. By taking fixed length representations of training examples and stacking them in a matrix, a frame or an over-complete basis can be constructed. Gemmeke and Cranen showed that sparse projections onto this frame recover the correct transcription with 91% accuracy at -5-dB SNR. The goal of speech recognition is not sparse projection onto training tokens, but onto training types. Sparse projection onto types can be achieved by building a frame for each word in the dictionary and stacking the frames to form a rank 3 tensor. Speech recognition is performed by convex linear projection onto the tensor, with sparsity enforced only in the index that specifies type. A mixed L1/L2 relaxation was derived, leading to a Newton descent algorithm that converges to the global optimum.

1:25

5pSC2. Speech classification using penalized logistic regression with hidden Markov model log-likelihood regressors. Øystein Birkenes (TANDBERG, Philip Pedersens vei 20, 1366 Lysaker, Norway, oystein.birkenes@tandberg.com), Tomoko Matsui (The Inst. of Statistical Mathematics, Tokyo, Japan), Kunio Tanabe (Waseda Univ., Tokyo, Japan), and Magne Hallstein Johnsen (Norwegian Univ. of Sci. and Technol., Trondheim, Norway)

Penalized logistic regression (PLR) is a well-founded discriminative classifier with long roots in the history of statistics. Speech classification with PLR is possible with an appropriate choice of map from the space of feature vector sequences into the Euclidean space. In this talk, one such map is presented, namely, the one that maps into vectors consisting of log-likelihoods computed from a set of hidden Markov models (HMMs). The use of this map in PLR leads to a powerful discriminative classifier that naturally handles the sequential data arising in speech classification. In the training phase, the HMM parameters and the regression parameters are jointly estimated by maximizing a penalized likelihood. The proposed approach is shown to be a generalization of conditional maximum likelihood (CML) and maximum mutual information (MMI) estimation for speech classification, leading to more flexible decision boundaries and higher classification accuracy. The posterior probabilities resulting from classification with PLR allow for continuous speech recognition via N-best or lattice rescoring.
Many acoustic events, particularly those associated with speech events, can be thought of as events in a rich descriptive subspace where the dimensions of the subspace can be thought of as a sort of decomposition of the original event space. In phonetic terms, we can think of how phonological features can be integrated to determine phonetic identity; for auditory scene analysis we can look how features like harmonic energy and cross-channel correlation come together to determine whether a particular frequency corresponds to target speech versus background noise. Some success has been achieved by thinking of these problems as probabilistic detection of acoustic (sub-)events. However, event detectors are typically local in nature and need to be smoothed out by looking at neighboring events in time. This talk describes using conditional random fields models within the automatic speech recognition setting to combine bottom-up speech event detectors. The talk will explore some of the successes and limitations of this log-linear method which integrates local evidence over time sequences.

2:05

Current ASR uses two main sources of information, the top-down prior knowledge in the form of a statistical language and the bottom-up information from acoustic speech signal. The information from acoustic signal is present in so called speech attributes that are derived from the signal. Any information that gets lost during attribute extraction cannot be recovered. Any preserved information that is irrelevant to the ASR task could complicate further processing. Attributes in conventional ASR are derived by some transformation of the short-term spectrum of speech that represents magnitude frequency components of a short segment of the signal. Origins of this representation can be traced to speech vocoding. For a past decade, we are working on attributes that are derived from posterior probabilities of speech sounds that are estimated by multilayer perceptron trained on large amounts of labeled data. This speech representation is now used in most leading state-of-the-art experimental systems. The talk will describe several research issues that influence the effectiveness of these posterior-based attributes. These issues include the choice of classes of speech sounds, input evidence for the posterior estimation, architecture and training of the posterior probability estimator, and post-processing of posterior estimates for use with conventional GMM-HMM ASR.

2:25

Articulatory information can improve the performance of automatic speech recognition systems. Unfortunately, since such information is not directly observable, it must be estimated from the acoustic signal using speech-inversion techniques. Here, we first compare five different machine learning techniques for inverting the speech acoustics generated using the Haskins Laboratories speech production model in combination with HLsyn. In particular, we compare the accuracies of estimating two forms of articulatory information (a) vocal tract constriction trajectories and (b) articulatory flesh-point pellet trajectories. We show that tract variable estimation can be performed more accurately than pellet estimation. Second, we also show that estimated tract variables can improve the performance of an autoregressive neural network model for recognizing speech gestures. We compare gesture recognition accuracy for three different input conditions: (1) generated acoustic signal and estimated tract variables, (2) acoustic signal and the original (or groundtruth) tract variables, and (3) acoustic signal only. Results show that gesture recognition accuracy was, not surprisingly, best for condition (2) and worst for condition (3). Importantly, however, condition (1) yielded better performance than (3), demonstrating that estimated tract-variable articulatory information is indeed helpful for automatic speech recognition.

2:45
5pSC6. Margin based discriminative training techniques for automatic speech recognition. Fei Sha (Dept. of Comput. Sci., Univ. of Southern California, Los Angeles, CA 90089), Chih-Chieh Cheng, and Lawrence Saul (Univ. of California, La Jolla, CA 92093)

Recently, large margin techniques have gained popularity for discriminative training of continuous-density hidden Markov models. These techniques are motivated by statistical learning theory, aiming to strike a balance between model complexity and generalization error. When used in speech recognition, these techniques separate discriminant scores from the correct transcriptions from all incorrect recognizer outputs. The parameters of acoustic models are adjusted to maximize the separation—margin—that is proportional to recognition errors. Many of such techniques have been successfully applied to large vocabulary speech recognition tasks and have improved the performance of existing systems with traditional approaches for parameter estimation. In this talk, I will review some of the work that my collaborators and I have undertaken in this direction. I will also describe a few new advances, including online learning of large margin separation as well as applying margin based techniques to adaptively transform acoustic features jointly with acoustic models.
3:05

**Contributed Paper**

**5pSC7. Spoken Hindi paired word recognition using probabilistic neural network.** Dinesh Kumar Rajoriya, R. S. Anand, and R. P. Maheshwari (Dept. of Elec. Eng., Indian Inst. of Technol. Roorkee, Roorkee, Uttarakhand 247 667, India, dineshajoriya@gmail.com)

Automatic speech recognition systems play a significant role in the development of real-world applications. This paper demonstrated spoken Hindi (Indian national language) paired word recognition (SHPWR) system, which has been examined with the help of a probabilistic neural network as a classifier. This type of network is a combination of a radial basis layer and a competitive transfer function layer which picks the maximum probabilities as a final result. SHPWR is pattern recognition (PR) problem. A PR technique encompasses two fundamental tasks; description and classification. In the description process, features are extracted from the Hindi paired word templates, and probabilistic neural network is used as a classifier. For the experimental point of view 1000 Hindi paired word templates have been recorded from individuals with different environment condition, gender, and age groups. Speaker dependent SHPWR systems achieved close to 100% while the accuracy of speaker independent SHPWR systems is quite adequate.

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**FRIDAY AFTERNOON, 23 APRIL 2010**

**ESSEX A/B/C, 1:00 TO 3:35 P.M.**

**Session 5pSP**

**Signal Processing in Acoustics, Underwater Acoustics, and Animal Bioacoustics: Classification Methods in Acoustics and Non-Gaussian Noise II**

R. Lee Culver, Cochair

Pennsylvania State Univ., Applied Research Lab., P.O. Box 30, State College, PA 16804-0030

Brett Bissinger, Cochair

Pennsylvania State Univ., Applied Research Lab., P.O. Box 30, State College, PA 16804-0030

**Invited Papers**

**1:00**

**5pSP1. Formulation for statistics of echoes due to a finite number of scatterers and patches of scatterers in a directional sonar beam.** Timothy K. Stanton (Dept. Appl. Ocean Phys. and Eng., Woods Hole Oceanograph. Inst., MS 11, Woods Hole, MA 02543, tstanton@whoi.edu)

A general theoretical formulation is developed for echo statistics associated with a finite number of scatterers and patches of scatterers, each being randomly located in a directional sonar beam. This is the case of a direct-path sonar where there are no echoes from the seafloor and sea surface. The formulation is derived by combining the equations of Ehrenberg and colleagues (single scatterer randomly located in sonar beam) and Barakat (sum of finite number of random variables). Each scatterer or patch of scatterers is independent and can have its own echo probability density function (PDF) (before beampattern effects). Examples show that the PDF of the echo (as received by the sonar) is strongly non-Rayleigh due to beampattern effects and finite number of scatterers or patches. Numerical simulations are made for validation. [Work supported by ONR.]

**1:20**

**5pSP2. Featureless classification for active sonar systems.** Mary E. Soules and Joshua B. Broadwater (The Johns Hopkins Appl. Phys. Lab., 11100 Johns Hopkins Rd., Laurel, MD 20723)

Active sonar systems depend on classification algorithms to identify target echoes and suppress false alarms. Historically, classifiers use a set of empirically derived features that exhibit some statistical separation between background clutter and target echoes. In an ideal scenario, these features would form a sufficient set of statistics capturing all of the information required to classify an echo return. Unfortunately, due to their empirically derived nature, features are rarely provably sufficient. To overcome this drawback, a featureless classifier will be presented. Instead of features, the raw data samples, which form a trivial but provable set of sufficient statistics, are used. The adaptive cosine estimate algorithm has a history of success with featureless classification in other applications and is well suited for underwater acoustics. An in-depth look at the featureless algorithm that will include comparisons to several traditional active sonar classifiers will be provided.

**1:40**

**5pSP3. Clustering classification methods for acoustic applications including the analysis of clicking whale data.** Juliette W. Ioup, George E. Ioup, Lisa A. Plugh (Dept. of Phys., Univ. of New Orleans, New Orleans, LA 70148, jioup@uno.edu), Natalia A. Sidorovskaia (Univ. of Louisiana at Lafayette, Lafayette, LA), Christopher O. Tiemann (Univ. of Texas at Austin, Austin, TX), and Charles H. Thompson (NOAA/NMFS/SEFSC, Stennis Space Ctr., MS)

Several clustering methods are available to group like objects into classes. The K means approach is well established. Both standard and homegrown algorithms have been tested in the following applications. One neural network technique is self-organizing map clus-
ter [T. Kohonen, *Self-Organizing Maps*, 2nd ed, Springer, New York (1997)]. Although it has some similarities to K means, it has notable differences which can be quite helpful in classification. An advantage is that it has the ability to limit the number of clusters automatically. This can be an important aspect when the number of classes is unknown. After a review of the basics of these techniques, the power of the methods is illustrated by applications to the study of clicking whales. The Littoral Acoustic Demonstration Center has acquired underwater acoustic data from clicking whales in the Gulf of Mexico, and it has access to data from the third and fourth International Workshops on Detection, Classification and Localization of Marine Mammals Using Passive Acoustics. Application of K means and SOM clustering to these data is an important component of identifying individual whales acoustically. Additional details on these results are being presented in Animal Bioacoustics sessions at this meeting. [Research supported by ONR and SPAWAR.]

2:00

5pSP4. Waveguide invariant minimum variance scatterer depth classification for active sonar. Ryan Goldhahn, Granger Hickman, and Jeffrey Krolik (Dept. of ECE, Duke Univ., 130 Hudson Hall, Durham, NC 27708-0291)

Active sonar systems are plagued by false alarms due to confusion between returns from water-column targets and backscatter from the bottom. Both feature-based and physics-based classifiers are notoriously susceptible to mismatch of the environment used for training and/or modeling active sonar returns. In this paper, in order to achieve more robust classification, uniformly sub-sampled DFT coefficients from a single snapshot of the wideband active sonar return are used to define a waveguide-invariant spectral density matrix (WI-SDM). The WI-SDM facilitates adaptive matched-filtering based approaches for target depth estimation, where the waveguide invariant property is exploited to obtain uncorrelated snapshots without inflating covariance matrix rank. Depth classification is then performed by designing a waveguide-invariant minimum variance filter (WI-MVF) with adaptive weights which minimize ambiguous deep sidelobes. Simulation and real data results in a shallow-water Mediterranean environment are presented to illustrate the approach.

[Work sponsored by ONR.]

Contributed Papers

2:20

5pSP5. Testing the temporal robustness of an automatic aural classifier. Stefan Murphy and Paul C. Hines (Defence R&D Canada, P.O. Box, 1012 Dartmouth, Nova Scotia, B2Y 3Z7, Canada stefan.murphy@drcd-rddc.gc.ca)

Military sonar systems must detect and classify submarine threats at ranges safely outside their circle of attack. However, in littoral environments, echoes from geological features (clutter) are frequently mistaken for targets of interest, resulting in degraded performance. Perceptual signal features similar to those employed in the human auditory system can be used to automatically discriminate between target and clutter echoes, thereby improving sonar performance. [J. Acoust. Soc. Am. 122, 1502–1517 (2007)]

The present work examines the temporal robustness of the aural classifier using data from two field trials: the first in 2007 and the second in 2009. The experiments were conducted on the Malta Plateau using a cardiod towded-array receiver, and a broadband source transmitting linear FM sweeps from 600–3500 Hz. The data set consists of hundreds of pulse-compressed echoes from several surrogate targets and geological clutter objects. The echoes are examined using an automatic classifier that processes each echo to extract perceptual features. Each echo is classified as target or clutter based on the position vector formed by these features. The classifier establishes a boundary between clutter and target echoes in the feature space using the 2007 experiment. Temporal robustness is investigated by testing the classifier on echoes from the 2009 experiment. In this work, the experiments are reviewed and initial results are presented.

2:35

5pSP6. On the use of instantaneous acoustic intensity for the classification of defects in air-filled pipes. Kirill Horoshenkov, Tareq Bin Ali, Simon Tait (School of Eng., Univ. of Bradford, Bradford BD7 1DP, United Kingdom), and Alexandra Tolstoy (Afolstoy Sci., McLean, VA 22101)

The sound pressure in an air-filled pipe has been measured using six pairs of matched microphones. These data have been then analyzed to determine the six vectors of instantaneous particle velocity and acoustic intensity. The acoustic intensity vectors have been processed coherently using the matched field processing (MFP) technique to study the effect of changes such as blockages and cracks on the ambiguity surface. The results show that the technique is sensitive to very small changes in the pipe cross-section and the boundary conditions on the pipe wall. It enables us to locate a defect to within 1% of the measurement range. It also enables us to discriminate between blockages and wall cracks using the spectral characteristics of the ambiguity surface predicted via the MFP. This technique does not require acoustic modeling of the sound field in the pipe, but it relies on the sound pressure data recorded on three or more microphones in the absence and presence of a defect.

2:50

5pSP7. Source depth classification of passive sonar signals using amplitude statistics. R. Lee Culver, Colin W. Iemomm (Appl. Res. Lab. and Grad. Program in Acoust., Penn State Univ., P.O. Box 30, State College, PA 16804), Brett E. Bissinger, and N. K. Bose (Penn State Univ., State College, PA 16804)

A number of at-sea measurements have shown that fluctuations in the amplitude of passive sonar signals are affected by source depth, range, and the propagation path from the source to the receiver. However, to date, received signal statistics have not been incorporated into passive sonar classification algorithms because the field results have not been accompanied by development of a statistics-based classifier architecture whose performance can be assessed using accepted signal processing metrics, e.g., a receiver operating characteristic curve. The performance of a statistics-based classifier depends critically upon statistical knowledge of the received signal, which in turn depends upon statistical knowledge of relevant environment parameters (sound speed in the water column and sediment, for example). Thus the classifier architecture must make use of the environmental information which in reality is only approximately known and must be described in statistical terms. Given a statistical description of the environment, an ocean acoustic propagation model and Monte Carlo simulation can be used to predict received signal statistics. Once received signal statistics are available for the different classes, a classifier can be designed. Here we examine a likelihood ratio binary source depth classifier for passive sonar. [Work sponsored by ONR Undersea Signal Processing.]

3:05


Passive source classification in the underwater environment is a challenging problem in part because propagation through the space- and time-varying media introduces variability and uncertainty in the signal. Acoustic
propagation models can predict received fields accurately, but they are sensitive to input environmental parameters which cannot be known exactly. This uncertainty in environmental knowledge used in signal predictions results in imperfect statistical class models. Classifiers that rely on simulations of the environment must therefore be robust to these imperfections. The minimum Hellinger distance classifier (MHDC) has been shown to be robust to such mismatches in situations with no noise. Low-noise situations allow demonstration of the properties of the classifier; however, real passive sonar tends to have significant noise levels. Therefore the MHDC’s performance is examined and compared to that of a log-likelihood ratio classifier when applied to narrowband passive underwater acoustic signals in the presence of noise. Both classifiers are applied to synthetic Gaussian data, synthetic acoustic data, and actual acoustic data, all with noise. Classifiers are evaluated using receiver operating characteristic curves, a traditional performance metric in signal processing. [Work supported by ONR Undersea Signal Processing.]

Automatic identification-of-the-source is an important problem in the field of speech forensics. That is, given a speech recording, the problem is to determine how to automatically determine the exact make and model of the acquisition device regardless of the speaker and speech content. Recent work developed by the authors have shown that statistical modeling of the device characteristics in terms of MFCCs provides excellent discriminative power when the recordings are obtained under controlled conditions (i.e., all recordings come from the same room and a unique device per class is used). In this study we propose to expand these results by analyzing the effects that room acoustics and intra-device variability have in the source identification accuracy. To perform this analysis, the NIST 2008 SRE dataset will be used. This corpus provides recordings acquired with multiple devices of the same make and model in different rooms.

3:20

5pSP9. Speech forensics: Automatic acquisition device identification. Daniel Garcia-Romero and Carol Espy-Wilson (Dept. of Elec. Eng., Univ. of Maryland, College Park, MD 20742, dgromero@umd.edu)

Automatic identification-of-the-source is an important problem in the field of speech forensics. That is, given a speech recording, the problem is to determine how to automatically determine the exact make and model of the acquisition device regardless of the speaker and speech content. Recent work developed by the authors have shown that statistical modeling of the device characteristics in terms of MFCCs provides excellent discriminative power when the recordings are obtained under controlled conditions (i.e., all recordings come from the same room and a unique device per class is used). In this study we propose to expand these results by analyzing the effects that room acoustics and intra-device variability have in the source identification accuracy. To perform this analysis, the NIST 2008 SRE dataset will be used. This corpus provides recordings acquired with multiple devices of the same make and model in different rooms.