Session 2aAA

Architectural Acoustics: Architectural Design for Acoustics—Issues and Examples

Philip W. Robinson, Chair

Graduate Program in Architectural Acoustics, Rensselaer Polytechnic Inst., 110 8th St., Troy, NY 12180

Chair’s Introduction—8:40

Contributed Papers

8:45

2aAA1. The voices of Antoni Gaudi’s architecture. Arturo Campos (Calle 25 # 419-A x 52 Jardines de Mirida, Merida Yucatán, Mexico)

Antoni Gaudi is one of the most important architects in the history of architecture. His architecture has increased the beauty of Barcelona. The civil and religious architecture of Gaudi involves mystic sounds, real or imaginary sounds, whispers almost imperceptible, and sonorous surprises in the journeys destined to peace and spiritual calm. In this research, which come from the Ph.D. thesis “The voices of Gaudi,” the sound of his buildings is explained. Their acoustical phenomena are produced by his architecture and the materials. The acoustics and sounds of three important buildings “La Sagrada Familia,” “La Pedrera,” and “Park Güell” will be explained in the presentation. “The Sagrada Familia” in Barcelona is a church where the sonorous itineraries exist and where the acoustic design of choirs, bells, and bell towers was an essential idea of Antoni Gaudi. “La Pedrera” is a living building with its voice singing with the people and “Park Güell” with the sounds of people, fountains, and rustic materials. Architecture and acoustic design transform Gaudi’s architecture in a daily murmur, highly spiritual.

9:00


The aim of this research is to evaluate sound control in mass transit stations, with a case study from Central Anatolia. For this aim Ankara’s central metro station is selected. Today, with the improvement of building technology, integration of different professions is increased. With this integration, special systems are used for some of the buildings, named “special buildings.” At these buildings, which have special character, sound is one of the professions and it needs acoustic criteria for their special characters. Nowadays authorities understand that one-value of dB is not enough for accepting noise criteria, and researched the value depending on frequency band and noise criteria curves. However, one should understand that these works are not enough for the architecture, up to now. Especially for the “intelligent buildings”, which is one specific part of “special buildings”, for the purpose of verifying the noise criteria, which depend on frequencies, analyzing the frequency bands of noise is very important. In the work, Kızılay station was selected, for example, as an intelligent building and the frequency of noise was analyzed and noise criteria for the interior were proposed. The comparison of criteria and tests results showed that, between 250 and 2000 Hz frequency at Saturday and Monday, there are big problems.

9:15

2aAA3. Acoustical drawbacks at one type project for education buildings. Filiz Bal Kocyigit (Dept. of Architecture, FT FAD Fac., Karabuk Univ., Safranbolu, Karabuk, Turkey)

The aim of this research is to evaluate acoustical quality drawbacks at one type projects for education buildings. For this aim Ankara (for Central Anatolia) and Karabuk (for Black Sea region) regions’ national primary schools are researched according to room acoustics condition and environment factors. These schools’ projects are compared with some private schools and other cities’ national school projects, and several sounds levels are measured at different critical zones in schools. Researches showed that there are several effectuated sound sources in schools which sometimes can be incessant, sometimes temporal, and directly affect the indoor noise level. The importance of these levels is, first, at the signal to noise ratio for communication between students, teachers, and other staff. Second, hearing and understanding the loudspeaker in daily period and especially during emergency time is vitally important. At this research, two different critical materials, conditions, and areas are determined; daily and constantly critical zones and critical areas at emergency conditions such as stairs, emergency doors, corridors, elevator indoor, etc. Not only emergency time but also daily period problems which are directly related to the noise level in schools need different solutions for the control of acceptable sound levels according to international standards.

9:30

2aAA4. The influence of background noise in student achievement. Ana M. Jaramillo and Michael Ermann (School of Architecture, Virginia Tech, Cowgill Hall, Blacksburg, VA 24060, anaja@vt.edu)

During our entire life span, we spend time in the activity of learning (anything from basic survival skills to highly intellectual processes), but this activity occupies most of the time during the first years of our lives. The classroom becomes then the official learning space that is designed to be conducive to this activity. The learning process, regardless of the task difficulty, is not a mechanical one. It requires a mental process, concentration, and attention in various degrees. Distractions can be detrimental to the learning. Most of the learning activities that occur at school settings require some kind of oral communication (teacher-student or student-student), and these activities call for appropriate room acoustics. In the presence of high levels of background noise, human beings have other resources to better understand the signal of interest. Those can be visual cues, previous knowledge of the topic, or mental ability to “fill in the blanks” in the received speech. The last two are abilities that come with age and experience. The purpose of this study is to better assert the importance of acoustics parameters in the design of classrooms and their relation to student achievement.

9:45

2aAA5. Corpus aspects to measure Spanish speech intelligibility: The methodology. Jorge Sommerhoff (Instituto de Acústica, Facultad de Ciencias de la Ingeniería, Campus Miraflores S/N, Valdivia, Chile) and Claudia Rosas (Universidad Austral de Chile, Valdivia, Chile)

This study presents the methodology employed to develop a subjective measurement tool for Spanish speech intelligibility. It is an experimental corpus, consisting of logatoms with a consonant + vowel + consonant structure, including all phonetically possible combinations in Spanish, with data from the first stage of the ongoing research project Fondecyt Regular No. 1090249 (CONICYT) in order to draw up lists of logatoms to measure the articulation percentage and their correlation with the speech transmission index. The information provided will facilitate the local improvement of the acoustic quality when building public educational facilities and will also
contribute to the national development of efficient habitability standards through a novel Spanish linguistic test aiming to serve as a reference for its application in the rest of the world’s Hispanic varieties.

10:00—10:15 Break

10:15
2aAA6. Computer analysis of speech intelligibility in classrooms. Sergio Feijoo (Dept. de Fisica Aplicada, Univ. de Santiago de Compostela, 15782 Santiago de Compostela, Spain, sergio.feijoo@usc.es)

The objective of this research is to study the relationship between classroom characteristics and speech intelligibility in order to optimize the amount of acoustic treatment and its position in the room. A rectangular shaped room was modeled with the ULYSSES acoustic analysis software. Total sound level, first reflections, the difference between direct and reflected sound, reverberation time, ALCONs, and STI were calculated. The amount of absorption treatment and its position in the room were much more important than the position of both the absorbent materials and the students to obtain good intelligibility scores. Nevertheless, for similar intelligibility scores, overall sound level is higher when the amount of absorbent material is controlled, allowing the listeners in the rear to benefit from higher speech levels. A reduction of 78% in the area of absorbent material can be achieved while maintaining the same STI score for all the positions in the room, with an increase in overall sound level of 4–5 dB.

10:30

Building indoor environmental quality (IEQ) surveys such as that developed by the Center for the Built Environment (CBE) at the University of California at Berkeley are currently collecting data from many buildings around the country and the world. Other entities with a vested interest in the operation of the United States building stock are also collecting acoustical data. While these data sets are publically available, they have not been analyzed by the acoustical community. As the concept of sustainability grows, the importance of “proving” that IEQ in buildings including acoustics performs as designed will also grow. This paper will review some of the available acoustical survey data sets including the Environmental Protection Agency’s Building Assessment Survey and Evaluation Survey (BASE) and National Center for Education Statistics’ Public School Principals Report.

10:45
2aAA8. Prospects of the treatment of acoustical insulation in building codes of México. Mario E. Vergara (Dept. of Architecture, Universidad de las Americas Puebla, Cholula Puebla 72810, Mexico)

The acoustical insulation of dwellings in order to protect them from environmental noise is an issue not yet addressed in building regulations in Mexico, but the Federal Government, through the National Housing Commission (CONAVI), has promoted the development of a Building Code for Dwellings, which in the future could include provisions about this subject. So far, authorities have focused their attention on the problems of energy efficiency and thermal insulation of public buildings. On the subject of housing, they have proposed levels of thermal insulation that could be adopted as standards and have also promoted the study of constructive solutions appropriate to those standards. The levels of acoustical and thermal insulation that are produced by various constructive solutions used by housing developers in the metropolitan area of the city of Puebla, Mexico, are analyzed in this work in order to compare their performance with international standards. A more comprehensive regulatory framework is needed in Mexico, and the results of this research will produce recommendations on acoustical insulation capacities that the National Housing Commission could include in its code. [Project supported by funds from CONACYT and CONAVI.]

11:00

In Latin America it is not common that the acoustic aspects in the construction of houses and residences were considered from the design. In this paper is presented a case of design of a residence, the property of an architect, in which from the initial stages the acoustic aspects were considered specifically for formal living room and piano area, TV room, home theater, bedrooms, dining room, kitchen, and bar room.

11:15

The acoustical comfort in restaurants can be perceived by the clients in terms of interpersonal communication, tranquility to eat, and also as environmental happiness at the place. That is a subjective factor that can make more popular a certain restaurant and can affect its business success. The reverberation time is a parameter related to the physical characteristics of a room and has influence on the acoustical sensation perceived by the costumers at a restaurant. In this paper the results of measurements of the reverberation time in restaurants at the city of Monterrey, Mexico, are compared with the opinion of its clients on the perception of their acoustical comfort.
Session 2aAB

Animal Bioacoustics: Software for Bioacoustic Analysis

David K. Mellinger, Chair
Oregon State Univ., 2030 S.E. Marine Science Dr., Newport, OR 97365

Invited Papers

8:30

2aAB1. Information technology challenges in passive acoustic monitoring of marine life. Michael Weise and James Eckman (Office of Naval Res., 875 N Randolph St., Arlington, VA 22203, michael.j.weise@navy.mil)

The explosion of passive acoustic monitoring (PAM) of marine life is generating massive amounts of data and creating information technology (IT) challenges in data collection, analysis, management, storage, and dissemination. We evaluate current levels of advancement for each IT challenge and identify areas that require further development. Current IT software is inadequate to identify all relevant calls and we do not adequately understand the complete range of sounds of interest, which necessitates the collection of high-bandwidth and long duration data sets. Large data sets generated by state-of-the-art PAM devices present several major challenges that require development of the computational capacity for timely data processing and display. Depending on the number of detectors and classifiers to be run, solutions may require anything from parallel computing to developing community-wide access to high throughput computing centers. New database approaches are needed that will enable users to query against known human or machine-generated annotations (detections, classifications) of existing data, add new annotation sets including detailed information on how they were created, export data products to external visualization services, and interface with existing external analytical tool sets. Databases or analytical services should be kept in repositories that are publicly accessible.

2aAB2. Software tools for marine mammals acoustic surveys and the implementation of mitigation procedures. Gianni Pavan, Claudio Fossati, Giovanni Caltavuturo, Cristina Francia (CIBRA, Dept. of Animal Biology, Univ. of Pavia, Via Taramelli 24, 27100 Pavia, Italy, gianni.pavan@unipv.it), Manuel Bassanini, and Nicola Modena (Novagem Solutions, 46020 Villa Poma, Mantova, Italy)

Several research groups are by now developing and working with software and hardware to implement PAM services worldwide. Despite considering standardization as a fundamental requirement, extensive experimentation still has to be done. In particular, the balance between software based and operator based classification, analysis, and interpretation is critical. A software suite has been developed and is continuously updated to assist researchers in conducting acoustic monitoring in marine mammals surveys; the aim is to keep improving the detection and classification of cetacean sounds, the source localization, and the real-time integration with visual detections into a GIS. The ultimate goal is to produce data for distribution and density estimation, and to provide a solid platform for mitigation. These tools have been developed throughout 10 years of SIRENA research cruises (NATO URC) and LDEO (Columbia Univ., NY) seismic cruises. The core of the system is the SEAPRO PAM software package that provides multisensor/multichannel wide-band recording and spectrographic display along with navigation data management and display, operator based sound classification, GIS plotting (integrated with sightings), and audio tools to make all received signals audible to the operator. Other modules provide spectro-statistical noise analysis, AIS data visualization, sound classification, and network transmission of acoustic data for distributed processing.

2aAB3. Evoked response study tool software for single and multiple evoked potential measurements in animals. James J. Finneran (U.S. Navy Marine Mammal Program, SSC Pacific Code 71510, San Diego, CA 92152, james.finneran@navy.mil)

The Evoked Response Study Tool (EVREST) is an integrated, portable hardware/software system designed for field measurement of evoked potentials. The software component of EVREST is a Windows-based application that is used for generating and calibrating sound stimuli and recording and analyzing evoked responses. The software has been used by a number of investigators to measure auditory evoked potentials in fish, marine mammals, and primates. The software features a graphical user interface and is compatible with a variety of commercial off-the-shelf data acquisition hardware. Options for sound stimuli include arbitrary waveforms loaded from disk or sums of up to 16 individual waveform components consisting of clicks or amplitude/frequency modulated tones. Evoked response data are averaged/displayed in real-time and also streamed to disk, allowing post-acquisition changes to analysis parameters. Objective response detection procedures include the F test, magnitude-squared coherence, and phase coherence. An adaptive staircase procedure is included for automated threshold measurements at multiple frequencies. In this talk, the operating principles and capabilities of the EVREST software are reviewed and example evoked potential data from dolphins and sea lions are presented. [Work supported by ONR.]
2aAB4. TRITON software package: Analyzing large passive acoustic monitoring data sets using MATLAB. Sean M. Wiggins (Scripps Inst. of Oceanogr., 9500 Gilman Dr., La Jolla, CA 92039-0205), Marie A. Roch (San Diego State Univ., San Diego, CA 92182-7720), and John A. Hildebrand (Scripps Inst. of Oceanogr., La Jolla, CA 92039-0205)

As consumer electronic technology continues to advance with lower power, higher speed, larger data capacity, and lower costs, the ability to collect large acoustic data sets becomes more practical. However, working with these large data sets can be challenging and requires different approaches to managing and analyzing the data than traditionally have been used with small data sets. TRITON is a software package developed for working with long-term, wide-band, multi-terabyte, acoustic data sets. One of TRITON's primary functions is to make long-term spectrograms from a large group of small (1 Gbyte) sequential data files. These long-term spectral averages provide a means of quickly evaluating long durations (hours) of wide-band data for sounds such as animal calling bouts and permit the user to zoom in to traditional spectral/time domain displays for finer scale analysis. TRITON has a graphical user interface and is based in MATLAB, which allows for easy modifications for specific data processing and analysis requirements. Examples of this are whistle and echolocation click detectors developed as TRITON extensions.


In the early 1990s web browsers were just emerging, leading us to the now obvious conclusion that the stand-alone computers in offices around the world could be woven into a distributed web of resources to form a globally shared library. With the support and encouragement of the U.S. Office of Naval Research, we established the Ocean Acoustics Library to serve as a means of publishing software of general use to the international ocean acoustics community. This paper will survey the available tools, with an aim also to explain the evolutionary process that has led to a set of models with quite different, but survivable characteristics.

10:10—10:30 Break


Automatic detection of animal vocalizations is now used widely for handling long-duration recording. Automatic detection methods inevitably make some errors—both false positive (wrong detection) and false negative (missed call) errors. Here a system is described for checking these errors. The MATLAB system “Osprey” allows viewing spectrograms, manipulating their parameters, and making various measurements of the displayed sounds. Another program, “checkDetections”, takes a log file that was output by an automatic detection software and systematically displays the detected sound in Osprey, allows a user to indicate whether the detection is correct, and then skips to the next detection in the log. This allows for rapid checking of detected sounds and calculation of the false-positive (wrong-detection) error rate. A second system, “checkMissedCalls”, displays random segments of sound in which no calls were found and allows the user to check whether there really were calls. This allows the user to estimate the false-negative (missed-call) error rate. Together these programs can be used to characterize a detector and arrive at a more accurate estimate of the number of calls present. Both of the programs are parametrized so that they can easily work with multiple data sets.

Contributed Papers

10:50

2aAB7. Rapid acquisition of marine mammal evoked potential audiograms by stranding networks. Dorian S. Houser (Natl. Marine Mammal Foundation, 2240 Shelter Island Dr. Ste. 200, San Diego, CA 92106, dorian.houser@nmnfoundation.org), Kathleen Moore, Sarah Sharp (Int. Fund for Animal Welfare, Washington, DC 20036), and James J. Finne-rain (U.S. Navy Marine Mammal Program, San Diego, CA 92152)

Two portable evoked potential measurement systems were constructed for use by stranding networks to determine the feasibility of non-expert collections of cetacean audiometric data. The systems were based on a customized version of the evoked response study tool with a simplified user interface designed for testing odontocetes. Stranding network personnel were provided one day of training on the system. Between January and April, 2010 the International Fund for Animal Welfare Marine Mammal Rescue and Research program attempted unsupervised AEP measurements on five live-stranded odontocetes: a harbor porpoise (Phocoena phocoena, n=1), common dolphins (Delphinus delphis, n=3), and an Atlantic white-sided dolphin (Lagenorhynchus acutus, n=1). Full or partial audiograms were obtained from the common and white-sided dolphins. The skin condition of the harbor porpoise was sufficiently poor as to prevent consistent electrode contact during testing. No audiogram was obtained from this animal. Audio- grams were typically dolphin in nature with an upper frequency limit approaching 160 kHz. Stranding network success at collecting odontocete AEP audiograms demonstrates that hearing tests can be conducted by non-expert operators with a streamlined and simplified testing system. This approach will increase the rate at which audiometric data are collected from marine mammals.

11:05

2aAB8. Integrating techniques for identifying individual sperm whales acoustically. Natalia Sidorovskaia (Dept. of Phys., UL Lafayette, UL Box 44210, Lafayette, LA 70504-4210, nas@louisiana.edu), Christopher O. Tie-mann (Univ. of Texas at Austin, Austin, TX 78712), George E. Ioup, and Juliette W. Ioup (Univ. of New Orleans, New Orleans, LA 70148)

Individual sperm whales may be identified acoustically from the properties of individual clicks or from the characteristics of their click trains. Thus far, our group has used cluster analysis (mainly self-organizing maps) and click change detection to identify whales by emphasizing the properties of individual clicks. Click change detection addresses the problem of click changes due to changing aspect (a turning whale) or other factors. Cadence analysis has been used to separate the clicks of individual whales based on click train properties. Localization has also been used to separate the clicks of individual whales. Manual click train identification employs both individual click properties and sequence analysis to identify individual whales. Each of these techniques has shown promise. Some have been compared on
the same data set. In this work, multiple methods are applied to the same data sets in order to verify the methods and to develop a multi-tiered approach to identifying individual sperm whales. [Research supported by SPAWAR and ONR.]

11:20

2aAB9. A system for tracking frequency-modulated bowhead whale calls on vector sensors in the presence of seismic exploration activity. Aaron Thode (Marine Physical Lab., Scripps Inst. of Oceanog., UCSD, 9500 Gilman Dr., La Jolla, CA 92110-0238, athode@ucsd.edu), Katherine H. Kim, Susanna B. Blackwell, Charles R. Greene (Greeneridge Sci. Inc., Santa Barbara, CA 93117), and Michael Macrander (Shell Exploration and Production Co., Anchorage, AK 99503)

A group of software tools has been developed for automated detection and localization of bowhead whale sounds with arbitrary frequency-modulated tones. The software is a mixture of JAVA, UNIX C-shell scripts, and MATLAB components, and consists of six sequential components: an incoherent spectral band event detector that yields time and bearings of transient events; an interval estimator to remove weak airgun signals; a feature extractor that processes an input spectrogram to yield a 23-element feature vector; two feed-forward neural network classifiers that sequentially winnow non-biologic signals and pinnipeds; a linking procedure to combine signals detected on sensors several kilometers apart; and a localization method that uses a maximum-likelihood procedure for estimating a position from a series of bearings. The software has been applied to data collected between 2007 and 2009 from at least 35 vector sensor autonomous recorders deployed over a 280-km swath in the Beaufort Sea. The largest dataset (2008) logged 1.35 × 10^6 calls at 386,000 locations over 54 days across 40 recorders and took 24 days to compute using a single 2.66-GHz processor. Seven recorders consumed over 50% of the processing time. [Work supported by Shell Exploration and Production Company.]

TUESDAY MORNING, 16 NOVEMBER 2010

Session 2aAO


Peter Gerstoft, Cochair

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Martin Siderius, Cochair

Dept. of Electrical and Computer Engineering, Portland State Univ., 1900 S.W. 4th Ave., Portland, OR 97201

Chair’s Introduction—8:00

Invited Papers

8:05

2aAO1. Equipartition and retrieval of Green’s function. Richard L. Weaver (Dept. of Phys., Univ. of Illinois, 1110 W Green St., Urbana, IL 61801, r-weaver@uiuc.edu)

Research on retrieval of Green’s function by correlation of ambient noise has seen enormous growth in recent years, in particular in seismology, exploration geophysics, and ocean acoustics, and in general whenever receivers are convenient while active sources are not. As such it resembles various other methodologies such as daylight imaging, sound shadows, and correlations with noisy sources which are, in fact, different in principle. Retrieval has its theoretical origin in the concept of wave equipartition and the fluctuation dissipation theorem of statistical thermal mechanics. Proof that the Greens function is retrieved from the correlations of noise is the simplest and most elegant in the thermal case, but acoustic noise is rarely thermal and those proofs do not always apply. Nevertheless, diffuse acoustic wave fields will sometimes approximate equipartition (it is routinely assumed in room acoustics and statistical energy analysis), and retrieval is often strikingly feasible even where that approximation is weak. This talk will review some of the history of retrieval in acoustics, particularly its analytic proofs and its successful applications. Rates of convergence, the effect of non-stationarity of noise, and the practical effect of imperfect equipartition, will also be discussed.

8:55

2aAO2. Ambient noise in the Mariana Trench. David R. Barclay and Michael J. Buckingham (Scripps Inst. of Oceanogr., Univ. of California, San Diego, 9500 Gilman Dr., La Jolla, CA 92093-0238, dbarclay@mpl.ucsd.edu)

In November 2009, ambient noise measurements were made in the Mariana Trench from the surface to a depth of 9000 m using the instrument platform Deep Sound. Deep Sound is a free-falling acoustic recorder designed to descend from the ocean’s surface to a pre-assigned depth where it drops an iron weight and returns to the surface under its own buoyancy. The ascent and descent rate is 0.6 m/s, resulting in an 8 deployment time to 9 km. The instrument recorded the continuous ambient noise time series over the bandwidth 5 Hz–30 kHz on four hydrophones mounted with vertical and horizontal spacings. Environmental data were recorded on a CTD and used to calculate the local sound speed. Power spectra of the ambient noise were calculated as a function of depth, while vertical and horizontal coherence were calculated and used to infer information on the directionality of the noise field. The spectral levels of ambient noise over the measured range of frequency were found to increase with depth. [Work supported by ONR and the UC Ship fund.]
2aAO3. Noise inversion: Making use of the signed impulse response. Chris Harrison (NATO Undersea Res. Ctr., Viale San Bartolomeo 400, 19126 La Spezia, Italy)

The impulse response of sediment layering can be obtained by cross correlating ocean ambient noise from the steered beams of a vertical array. Given a moving array the envelope of this signal (i.e., the Hilbert transform) maps out a sub-bottom profile. However, there is still additional information in the signed noise amplitude, and this will be demonstrated with experimental data from a number of sites at which a vertical array was moored. Essentially the impulse response is equivalent to that of any active system (the effective transmit pulse being determined by the user-defined processing band) and, consequently, this process is sensitive to changes in acoustic impedance. The quality of the signal can be improved by adaptive beam forming but although it has been shown that, strangely, this reverses the sign [Harrison, J. Acoust. Soc. Am. 125, 3511–3513 (2009)] here we show it does not alter its fidelity. In fact, if the number of layers is constrained as often done in inversion, it is possible to resolve their separations too even if smaller than the effective transmitted pulse length. More information is also available in the comparison of up/down beam powers at angles other than vertical.

2aAO4. Ocean bottom profiling with ambient noise: A model for the passive fathometer. James Traer, Peter Gerstoft (Scripps Inst. of Oceanogr., Univ. California San Diego, 9500 Gilman Dr., La Jolla, CA 92093, jtraer@ucsd.edu), and Martin Siderius (Portland State Univ., Portland, OR 97201)

An analytical model is presented for the passive fathometer response to ocean surface noise, interfering discrete noise sources, and locally uncorrelated noise in an ideal waveguide. The leading order term from the ocean surface noise produces the cross-correlation of vertical multipaths, yielding the depth of sub-bottom reflectors. Discrete noise incident on the array produces peaks that may obscure the sub-bottom reflections but can be attenuated with the use of adaptive processing. Such adaptive methods are described and applied to several data sets to demonstrate improvements possible as compared to conventional processing. Motion of the array, due to surface waves, is inferred from the passive fathometer response and, applying a correction for this motion, the seabed reflection is averaged coherently to yield an estimate of the seabed Green’s function.

9:55—10:20 Break

10:20

2aAO5. Ocean acoustic thermometry with ambient noise. Oleg A. Godin (CIRES, Univ. of Colorado and NOAA/Earth System Res. Lab., Mail Code R/PSD99, Boulder, CO 80305-3238, oleg.godin@noaa.gov), Nikolay A. Zabotin (CIRES and Dept. of Elec. and Comput. Eng., Univ. of Colorado, Boulder, CO 80309-0425), and Valery V. Goncharov (P. P. Shirshov Oceanology Inst. of the Russian Acad. of Sci., Moscow 117997, Russia)

Diffuse acoustic illumination, provided by ambient and shipping noise in the ocean, can be used as a probing signal to characterize the environment in a cost-effective and non-invasive manner. Valuable quantitative information about sound speed and, hence, temperature fields in water is contained in acoustic travel times between spatially separated receivers. The travel times can be retrieved from cross-correlations of noise recorded by the receivers. Practical applications of noise interferometry to ocean thermometry face a number of obstacles, the main challenge being the need to achieve very high accuracy of passive sound-speed measurements in a dynamic environment with a complex noise field. This paper reviews the theoretical background of wave interferometry in inhomogeneous media with non-perfectly diffuse noise from the viewpoint of ocean remote sensing. Feasibility of ocean acoustic thermometry is illustrated using noise recordings of opportunity, which were obtained in the North Pacific Ocean as a by-product of a long-range sound propagation experiment. Requirements to future dedicated passive acoustic systems for long-term observations of variations in the ocean heat content are discussed. [Work supported by ONR.]

10:40

2aAO6. Cross-correlation of diffuse surface noise between horizontally separated sensors, Stephanie E. Fried, Shane C. Walker, and William A. Kuperman (Scripps Inst. of Oceanogr., Univ. of California San Diego, 9500 Gilman Dr., La Jolla, CA 92093-0238, sfried@ucsd.edu)

The use of ambient noise in the ocean to extract the time of arrival structure of the time domain Green’s function (TDGF) between horizontally separated sensors was first demonstrated in deep water using shipping noise [Roux et al., J. Acoust. Soc. Am. 116, 1995 (2004)]. Subsequently, surface noise recorded along a vertical array was used in the passive fathometer application [Siderius et al., J. Acoust. Soc. Am. 120, 1315 (2006)]. A noise field from primarily volumetrically distributed biological noise sources recorded along a horizontal array was also shown to extract a shaded TDGF in very shallow water [Fried et al., J. Acoust. Soc. Am. 124, EL183 (2008)]. This same noise correlation function (NCF) processing is now demonstrated along horizontally separated hydrophones using diffuse, low sea-state surface noise, where the spectrogram of the noise reveals no significant shipping component. These results are presented along with a discussion of the relevant signal processing as well as the theory and simulation used to understand them. Of particular interest is the build-up time of the NCF under these relatively homogeneous environmental conditions as opposed to what other less homogeneous situations might produce.
Remote sensing in the ocean using ambient noise techniques is starting to be considered a viable option for surveying seabed properties. Historically, active techniques have been used to measure seabed layering and reflectivity. Part of the appeal of ambient noise remote sensing is that it allows for surveying in areas where active sound transmissions are not allowed (e.g., in marine protected areas). Although it may seem far-fetched to expect passive noise remote sensing methods to extract similar information about the seabed as active methods, in recent years results have been promising. However, direct comparisons between active and passive methods have been limited. One of the best ways to measure seabed properties is using what is often referred to as wide-angle reflection measurements. One version of the technique uses a single hydrophone at a fixed location with a sound projector transmitting from close range out to about a kilometer. The analysis of the reflection measurements are very similar to the ambient noise technique in that the seabed layering is extracted from the time-domain data and the frequency domain reflectivity is used to determine seabed properties. In this presentation, we will compare results from active reflectivity measurements with ambient noise remote sensing.

**11:15**

2aAO8. Retrieval of deterministic ray travel times from noise cross-correlations. Oleg A. Godin (CIRES, Univ. of Colorado and NOAA /Earth System Res. Lab., Mail Code R/FS039, Boulder, CO 80305-3328, oleg.godin@noaa.gov) and Nikolay A. Zabotin (Univ. of Colorado, Boulder, CO 80309-0425)

Two-point correlation functions of diffuse wave fields are known to approximate the deterministic Green’s functions that describe wave propagation between the two points. The deterministic ray travel times retrieved from cross-correlations of ambient noise recorded on vertical line arrays of hydrophones serve as the input data for passive acoustic thermometry and tomography of the water column. Accuracy of the sound-speed and water temperature inversions is largely controlled by the accuracy of the ray travel time measurements. Achieving oceanographically relevant results requires the relative error of the passive travel time measurements to be less than 0.1%. Often, the absolute error of the travel time measurement needs to be small compared to the acoustic wave period. In this paper, stationary-point analysis of an asymptotic, high-frequency integral representation of the noise cross-correlation function is used to determine ways for accurate retrieval of ray travel times. Limitations on the accuracy of the travel time retrieval due to noise anisotropy, medium non-stationarity, source and receiver motion, and finite noise averaging times are discussed. [Work supported by ONR.]

**11:30**


It has been demonstrated that the empirical Green’s function can be retrieved by cross correlation of ambient noise. When working with ambient noise in the ocean acoustic environment, long periods of averaging are necessary to achieve a good SNR of the extracted Green’s functions. For an array of sensors, the cross-spectral density matrix (CSDM) contains the equivalent information in the frequency domain. In this paper, we use statistical methods using random matrix theory (RMT) to improve the estimate of the CSDM and hence increase the convergence of the empirical Green’s function to the true Green’s function. In RMT, each element is considered a random variable and from this, features of the eigenvalue spectrum can be extracted. In this paper, we make use of the Marchenko–Pastur law of sample covariance matrices to analyze the eigenvalue spectrum of the CSDM and discard the part deemed to be just noise, based on the test hypothesis. Results are presented from the SW06 data.

**11:45**


Air-sea interaction is driven mainly by the wind blowing over the ocean surface, and its monitoring is an important task. Using ambient noise for the wind speed retrieval was suggested long ago; however, the proposed approaches rely usually on the local measurements with bottom-mounted hydrophones. We present another approach based on measurements of directivity of the ambient noise in the horizontal plane at low frequencies of the order of a few tens of hertz. Such measurements would be done with the help of ocean interferometers, which represent a couple of long vertical line arrays separated by a distance of the order of a few tens of kilometers and coherently measuring the ambient noise. In contrast to the methods of wind measurement based on local measurements, our approach relies on measurements of distant noise sources. The method principally allows mapping ocean wind over vast areas of the order of 1000 km in horizontal scale with resolution of the order of 10 km. A time of averaging required to overcome statistical variability of the noise field was estimated to be of the order of 3 h. The results of numerical simulations will be presented for propagation conditions typical for North Atlantic.
Biomedical Ultrasound/Bioresponse to Vibration: Doppler Methods and Blood-Flow Imaging in Medical Ultrasound

Jonathan Mamou, Cochair
Riverside Research Inst., 156 William St., New York, NY 10038

Michael L. Oelze, Cochair
Beckman Inst., Univ. of Illinois, 405 N. Mathews, Urbana, IL 61801

Invited Papers

8:00


A number of cancer therapies including radiation, chemotherapy, and anti-vascular treatments act by causing endothelial cell death, leading to microvascular destruction within tumours. We have been using high-frequency (20–40 MHz) power Doppler micro-ultrasound to study the effects of a variety of anti-cancer therapies on tumor blood micro-vessels. Using breast and prostate xenograft tumors in mice with radiation alone, large doses above 8 Gy appeared to diminish the vascular by up to 40%–60%. Radiation given in the presence of a basic fibroblast growth factor did not have this effect. The anti-angiogenic Sutent suppressed vascular growth and enhanced radiation effect, but tumor vasculature exhibited a rebound effect when therapy was stopped. In contrast, novel anti-vascular microbubble based therapies induced a greater synergistic tumor cell kill due to vascular disruption. Tumor cell kill increased when microbubbles were used in conjunction with radiation, with increases of cell kill from 5% to over 50% with combined single treatments linked to destruction of the micro-vasculature. In summary, power Doppler micro-ultrasound can be used quantitatively to assess the effects of anti-cancer therapies on tumor micro-vasculature. Effects of anti-angiogenic agents and newer anti-vascular therapies and their rebound effects can be monitored non-invasively and longitudinally in preclinical models.

8:20

2aBB2. On the use of three-dimensional Doppler acquisition for real-time volume flow estimation. Tej Desai, Stephen Z. Pinter, J. Brian Fowlkes (Dept. of Radiology, Univ. of Michigan, 1301 Catherine St., Ann Arbor, MI 48109-5667), Jonathan M. Rubin (Dept. of Radiology, Univ. of Michigan, Univ. Hospital, Ann Arbor, MI 48109), Man Zhang (Dept. of Radiology, Univ. of Michigan, Ann Arbor, MI 48109-5667), Anne L. Hall (GE Healthcare, Milwaukee, WI 53219), and Oliver D. Kripfgans (Univ. of Michigan, Ann Arbor, MI 48109-5667)

Clinical volumetric blood flow estimation relies on several assumptions. Among them are cylindrical vessel geometry, symmetric flow profile, and Doppler angle. None of them are known well enough to obtain clinically relevant estimates. 3-D color flow acquisition circumvents these assumptions, posing a viable tool for in vivo blood volume flow analysis. A 4-D cardiac scanner operating a 2-D array for real-time 3-D color flow imaging [GE Healthcare, Milwaukee, WI] was used. The array was positioned to fully intersect a 2-cm-diameter flow tube with the constant-depth plane (CPlane). Blood mimicking fluid was circulated at up to 6 l/min using a cardiac bypass pump (60 and 80 beats/min). A trigger source synchronized the pump and scanner. Data volumes were acquired equally spaced throughout the cardiac cycle. Temporally resolved volume flow was derived from CPlane data integration using Doppler partial volume correction. Results show less than 7% mean flow error for temporally resolved volume flow (100 points per cardiac cycle and 50 averages). Single points in the cardiac cycle (10 max total), including systolic and diastolic flow, can be acquired within 50 heartbeats. Average, nontime-resolved flow can be estimated within 5 s.

8:40


Medical ultrasound systems measure the blood velocity by tracking the blood cells motion along the ultrasound field. This is done by pulsing in the same direction a number of times and then finding e.g., the shift in phase between consecutive pulses. Properly normalized, this is directly proportional to the axial blood velocity. A major drawback is that only the axial velocity component is found. Often the lateral component is most important as blood vessels run parallel to the skin surface. The talk presents the transverse oscillation approach, which also can find the lateral velocity component by using a double oscillating field. A special estimator is then used for finding both the axial and lateral velocity components, so that both magnitude and phase can be calculated. The method for generating double oscillating ultrasound fields and the special estimator are described, and its performance revealed for a flow rig setup. Several examples from the clinical use of the approach are shown. From these it is seen that both velocity magnitude and angle varies temporally and spatially across the cardiac cycle, and it is, thus, important to estimate both continuously over the image region and time.
9:00
2aBB4. A flexible implantable sensor for post-operative monitoring of blood flow. Jonathan M. Cannata (Dept. Biomed. Eng., Univ. of Southern California, Los Angeles, CA 90089), David Vilkomerson, Tom Chilipka (DVX LLC, Princeton, NJ 08540), Hao-Chung Yang (Univ. of Southern California, Los Angeles, CA 90089), Sukgu Han, Vincent L. Rowe, and Fred A. Weaver (Univ. of Southern California, Los Angeles, CA 90033)

A blood flow measurement system utilizing flexible ultrasonic Doppler patch-sensors and battery operated electronics was developed. These sensors can be wrapped around a blood vessel to safely and accurately measure flow. Prototype sensors were fabricated by spin-coating 12-μm-thick P(VDF-TrFE) films onto 25-μm-thick flexible polyimide substrates with patterned gold electrodes.

The transmitting electrodes consisted of an array of equally spaced traces. Oppositely oriented electric fields were applied to alternate traces during polarizing so that when the traces are excited by a 40-MHz sinusoidal signal, the P(VDF-TrFE) film produces successive positive and negative pressure waves that sum to form two diffraction-grating beams at 39 deg with respect to blood flow. A uniform electrode for a receiving transducer was positioned to maximize the reception of the backscattered Doppler signal from the center of a 3-mm-diameter blood vessel. Patch-sensors were placed around the abdominal aorta of a New Zealand white rabbit. The mean peak blood velocity measured by the sensor during a 5-h period was 122 cm/s, which compared favorably to the peak velocity of 120 cm/s detected by a Duplex ultrasound system. [Work funded by NIH/NHLBI Grant no. R41-HL87467-01A2.]

9:20
2aBB5. A semi-quantitative method in microbubble-based ultrasound contrast imaging. Naohisa Kamiyama, Hiroki Yoshiara, and Tetsuya Yoshida (Ultrasound Dept., Toshiba Medical Systems Corp., 1385 Otawara, 324-8550, Japan, naohisa.kamiyama@toshiba.co.jp)

The aim of our study is to construct a semi-quantitative tool which can simply represent the measurable blood flow with a microbubble-based contrast agent. The possibilities of quantifying the blood flow are explored by many researchers to classify the tumors or to evaluate the response to anti-angiogenic therapy. Flash-replenishment (FR) sequence, which utilizes destruction of the bubbles by ultrasound exposure and observes reperfusion into the scan volume, has become a more applicable technique in contrast echography. The theoretical equations in FR sequence have been described to quantify hepatic parenchymal blood flow. The arrival time 2-D mapping was developed to acquire such diagnostic information easily in the routine examinations. This method watches the inflow of the microbubble in the scan image and paints colors pixel-by-pixel corresponding to the time of arrival. Results of the efficiency of this method will be shown, not only for tumor characterization but also for quantifying the fibrosis stage for diffuse liver disease.

9:40
2aBB6. The wall-filter selection curve method for objective tuning of power Doppler clutter filter cutoff velocity. James C. Lacefield and Stephen Z. Pinter (Biomedical Eng. Grad. Prog. and Robarts Res. Inst., Univ. of Western Ontario, London, ON N6A 5B9, Canada, jlacefie@uwo.ca)

High-frequency power Doppler ultrasound is commonly used to assess vascularity in small-animal cancer models, but quantitative images can be difficult to obtain in the presence of clutter artifacts. To improve vascular quantification, the color pixel density (CPD) in a region of interest can be plotted as a function of wall-filter cutoff velocity to produce the wall-filter selection curve. A mathematical model based on receiver operating characteristic statistics was developed to guide the interpretation of wall-filter selection curves. Mathematical predictions were tested using a VisualSonics Vevo 770 system with a 30-MHz transducer and a flow phantom containing four 200–300-μm-diameter vessels. The phantom mimicked vessel configurations observed in micro-CT images of a transgenic mouse prostate cancer model. Selection curves characteristically include a plateau whose CPD may correspond to either the total vascular volume fraction or to the volume fraction of a subset of vessels in the region. The flow-phantom data indicate that the plateau provides a reliable estimate of total vascularity if the plateau begins at a cutoff velocity <2 mm/s and is longer than 0.5 mm/s. The wall-filter selection curve may enable adaptation of scanner settings to changing flow conditions as a tumor progresses during a longitudinal study.

10:00—10:15 Break

Contributed Papers

10:15
2aBB7. Improved method for objective selection of power Doppler wall filter cut-off velocity for microvascular ultrasound imaging. Mai Elfarnawany, Stephen Z. Pinter, and James C. Lacefield (Biomedical Eng. Grad. Prog. and Robarts Res. Inst., Univ. of Western Ontario, London, ON N6A 5B9, Canada, melfar@imaging.roberts.ca)

The wall-filter selection curve method has been enhanced to improve detection and interpretation of color pixel density (CPD) plateaus. The improved algorithm was developed by analyzing data acquired from three fields of view in a four-vessel flow phantom using a 30-MHz swept-scan transducer. An N-point maximum envelope peak search applied to the first difference of CPD detects selection curve plateaus by incorporating criteria that identify intervals of minimum variation in CPD. Selection curves for regions of interest (ROIs) containing multiple vessels can be difficult to interpret, so the algorithm subdivides the image into small ROIs, constructs selection curves for each ROI, and sums the resulting vascularity estimates. The lower limit on ROI size is constrained by a need to avoid ROIs that are completely filled by blood. A multiple-step decision algorithm was designed that considers the number, length, and slope of each plateau to identify the cutoff velocity that yields the best vascularity estimate. At high (> 5 mm/s) flow velocities, the decision algorithm yielded a summed CPD that was within 5% of the vascular volume fraction in each field of view. These improvements are an initial step toward automating wall-filter cutoff settings in a power Doppler system.

10:30
2aBB8. Automatic detection of micro-emboli by means of a generalized autoregressive conditional heteroskedasticity model. J.-M. Girault, S. Menigot, and L. Deibine (UMRS Inserm U930, CNRS ERL 3106, Universite Francois Rabelais, Faculte de Medecine, 10 boulevard Tonnelle, BP3220, 37032 Tours, France)

Detection of micro-emboli is of great clinical importance to prevent cerebrovascular events and to identify the causes of such events. Standard detection techniques implemented in the most commonly used systems are
generally based on the passing of an energy threshold. The value of this threshold can be set just above the statistically highest detected energy of a Doppler signal related to circulating blood, i.e., during the systolic phase. This choice of threshold consequently prevents all detection of microemboli events whose energy might be lower than the systolic energy. The main idea of our technique was to test adaptively the whiteness of the prediction error between the Doppler signal and a synthetic signal obtained by a GARCH model. By assuming that micro-emboli are unforecasted, we hypothesize that embolic signals are unpredictable. We hypothesize that our new micro-emboli detector is a candidate technique to detect very small micro-emboli. We have tested and compared our new technique to the standard Fourier technique (Fourier). For a probability of detection fixed to 100%, the probability of false alarm of the standard automatic technique is 33% whereas the detector based on the GARCH model is less than 3%. This study demonstrates that our new technique detects micro-emboli hitherto not identified by classical methods.

10:45
2aBB9. Investigating vascular-targeting strategies with three-dimensional power Doppler ultrasound. Ahmed El Kaffas, Anoja Giles, and Gregory J. Czarnota (Dept. of Medical Biophys./Dept. of Radiation Oncology, Univ. of Toronto/Sunnybrook Health Sci. Ctr., 2075 Bayview Ave., Toronto, ON M4N 3M5, Canada, aelkaffas@gmail.com)

Vascular targeting agents have been recently combined strategically with existing cancer therapies to potentially enhance tumor response. Our aim was to investigate the role of blood vessels in radiation response and how blood vessels can be targeted to enhance treatments. Breast cancer MDA-MB-231 xenographs were treated with single radiation doses of 0–16 Gy alone, or in combination with Sutent, an antiangiogenic agent. 3-D ultrasound tumor data were acquired before and 24 h after treatment using a 25-MHz transducer and a VEVO770. The vascularity index (VI) was used to quantify blood from power Doppler data, while quantitative ultrasound spectroscopy (QUS) was used to monitor tumor cell death. Staining using TUNEL and CD31 of tumor sections was used to measure cell death and tumor vasculature distributions. Preliminary results indicated a VI decrease of up to 50% when tumors were irradiated with 16 Gy. Sutent-radiation combination treatments showed an increase in the VI, which may be associated with a vascular normalization effect. Analyses with QUS and TUNEL staining indicate an enhanced dose-dependent increase in tumor cell death when radiation was combined with Sutent. These results indicate that Sutent treatment may radiosensitize tumors by altering the tumor microenvironment. [Work funded by the Canadian Breast Cancer Foundation.]

11:00
2aBB10. High-frequency ultrasound Doppler velocimetry measurements of intra-cochlear structures in human temporal bones. Jeremy Brown, Zahra Torbatian, Phil Garland, Rob Adamson (Biomedical Eng., Dalhousie Univ., 5981 University Ave., Halifax, NS B3H 1W2, Canada, j.brown@dal.ca), Rene Van Wijhe, Julian Savage, and Manohar Bance (Dalhousie Univ., Halifax, NS B3H 2Y9, Canada)

Hearing loss is one of the most prevalent, chronic, and fast growing disorders, affecting approximately one-tenth of the population. Diagnostic imaging tools currently used in this field such as CT and MRI do not have sufficient spatial/temporal resolution to properly diagnose most of the underlying causes of hearing loss. In this work, we present the first high-resolution velocimetry measurements of intra-cochlear structures using high-frequency (40 MHz) pulsed-wave Doppler ultrasound. A 40-MHz single-element transducer based on PMN-PT single-crystal piezoelectric was fabricated in-house and mounted onto the tip of a needle. The transducer was a side-looking circular disk with a 1 mm outer diameter. A custom data acquisition system was developed in order to perform pulsed-wave Doppler and was synchronized with an acoustic stimulus. Velocimetry measurements were performed on the basilar membrane located inside the cochlea of fresh temporal bones by imaging across the round window membrane and stimulating the ear drum with pure-tone sounds. For the first time, the vibration of the basilar membrane was measured on a completely intact cochlea using ultrasound velocimetry. The middle ear resonance could be detected by measuring the frequency-dependent basilar membrane velocities between 100 and 2500 Hz. This resonance was verified using a laser vibrometer.

11:15
2aBB11. Investigating the effects of intensity threshold on high-frequency three-dimensional power Doppler ultrasound. Ahmed El Kaffas, Sheldon Kwok, Anoja Giles, and Gregory J. Czarnota (Dept. of Medical Biophys./Dept. of Radiation Oncology, Univ. of Toronto /Sunnybrook Health Sci. Ctr., 2075 Bayview Ave., Toronto, ON M4N 3M5, Canada, aelkaffas@gmail.com)

High-frequency power Doppler ultrasound is a non-invasive vascular imaging modality ideal for a quantitative assessment of vascularity in preclinical models. Although it permits detecting slower blood flows than other ultrasound Doppler methods, it is limited by flow artifacts, complicating the interpretation of tumor vascularity metrics such as the vascularity index (VI). Our aim was to examine how parameters such as the intensity threshold affect the VI in in vivo xenograft models. Breast MDA-MB-231 tumors in the hind leg of SCID mice were treated with 0 or 8 Gy radiation, and imaged before and 24 h after treatment. A Vevo770 was used along with a 25-MHz center frequency transducer to obtain 3-D power Doppler images of tumors. Mice were again imaged while in the same position 1 min after sacrifice at 24 h. This was carried out for direct comparison of true blood flow signal to system noise level. The VI was computed for a range of intensity thresholds. The VI-threshold curves varied differently as a function of treatment, and preliminary data indicate an optimal plateau where the intensity threshold should be set, yielding a nearly true blood flow signal and minimizing noise signal. [Work funded by the Canadian Breast Cancer Foundation.]

11:30
2aBB12. Estimating fetal heart rate from multiple Doppler ultrasound signals. I. Voicu, J.-M. Girault, D. Kouame, M. Fournier-Massignan, and F. Perrotin (UMRS Inserm U930, CNRS ERL 3106, Universite Francois Rabelais, Faculte de Medecine, 10 boulevard Tonnele, BP3220, 37032 Tours, France)

In utero, monitoring of fetal well being or suffering is today an open challenge due to the high number of clinical parameters to be considered. An automatic monitoring of fetal activity, dedicated for quantifying fetal well being, becomes necessary. For this purpose and in a view to supply an alternative for the Manning test, we used an ultrasound multitransducer multigate Doppler system. One important issue is the accurate estimation of fetal heart rate (FHR). An estimation of the FHR is obtained by evaluating the autocorrelation function of the Doppler signals for ills and healthiness of the fetus. However, this estimator is not robust enough since about 20% of the FHR is not detected in comparison to a reference system. These non-detections are principally due to the fact that the Doppler signal generated by the fetal movement is strongly disturbed by the presence of several other Doppler sources (mother’s movement, pseudo breathing, etc.). By modifying the existing method (autocorrelation method) and by proposing new estimators used in the audio’s domain, we reduce to 5% the probability of non-detection of the fetal heart rate. These results are really encouraging, and they enable us to plan the use of automatic classification techniques in order to discriminate between healthy and suffering fetus.

11:45
2aBB13. Measuring tendon velocities using vector tissue Doppler imaging. Avinash Erunkar (Dept. of Elec. & Comput. Eng., George Mason Univ., Fairfax, VA 22031, aerunkar@gmu.edu), Siddhartha Sikdar (George Mason Univ., Fairfax, VA 22031), Christopher Stanley, Laura Prosser, Lindsey Bellini, Daniel Bland, Katharine Alter, and Diane Damian (NIH, Bethesda, MD 20892)

We have developed a vector tissue Doppler imaging (vTDI) method to quantify the magnitude and direction of tissue motion. The goal of this study was to quantify the repeatability of vTDI in measuring the contraction velocity of the tibialis anterior (TA) tendon in patients with cerebral palsy and
foot drop (impaired dorsiflexion). vTDI was implemented on Ultrasonix Sonix Touch ultrasound system with a 5-14-MHz linear array transducer. The array was electronically split into two transmit and two receive apertures to estimate velocity vectors. Transmit and receive beams were steered by ±15 deg. We conducted 42 trials on 7 subjects. Our preliminary results show that TA tendon velocities measured using vTDI have a strong linear correlation with the joint angular velocity estimated using a conventional 3-D motion capture system. We observed a peak velocity of 5.20±1.58 cm/s during dorsiflexion and 8.45±2.06 cm/s during the gravity-aided passive relaxation phase. The R² values for all 42 trials were 0.77±0.10. A second velocity measurement was made on three subjects after an interval of 4 weeks. We obtained repeatable velocity estimates with the standard deviation of the radius of action less than 0.13 cm. This demonstrates that vTDI is a feasible and reproducible method for measuring tendon velocities.
2aED4. Teaching acoustics in Argentina. Daniel Gavinoich and Nilda Vechiatti (Laboratorio de Acustica y Electroacustica, Facultad de Ingenieria de la Universidad de Buenos Aires, Paseo Colon 850, primer piso, 1063 Buenos Aires, Argentina)

In Argentina, the teaching of acoustics is included in the curricula of several academic institutions, being a part of the technical instruction received both in undergraduate and graduate majors. Acoustics is part of numerous college and technical careers, several of which will be detailed during this presentation. At the same time, a large number of post graduate and extension training classes are offered by the same and other institutions. It is only fair to mention that the Engineering School of the Buenos Aires University was the pioneer in teaching this discipline in Argentina, when Electroacoustics started to be taught in 1956 as a mandatory subject in the formerly joint Electronics Engineering plan, thus enabling the formation of the Acoustic and Electroacoustic Laboratory of the Faculty, a laboratory that still exists. Shortly after, in October 1961, the Acoustic and Lighting Research Center was founded in the Architecture and Urbanism School at the Cordoba National University. The extensive growth and ramifications we have seen so far make the list unavoidably incomplete, since the present possibilities of learning this science in the country is too large to be exposed, allowing us a reduced mention and a brief exposure.

2aED5. History and educative programs of the Acoustics Laboratory of the Nuevo Leon State University in Monterrey, Mexico. Fernando J. Elizondo-Garza, Adrián García Mederez, and Carlos Armando Lara Ochoa (Acoust. Lab., Mech. and Elec. Eng. School, Univ. Autónoma de Nuevo León, P.O. Box 28, sucursal F, Cd. Universitaria, San Nicolás 66450, NL, Mexico, fjelizon@hotmail.com)

The Acoustics Laboratory of the Mechanical and Electrical Engineering School of the Nuevo Leon State University in Monterrey, Mexico, founded by Eng. Miguel Medina, has been the axis of the education, research, and consulting to companies and government in this field of science in the northeast of Mexico. In this paper the history of this department, its vision, the services offered, and the approach of the courses on acoustics that are offered to the students of the university as well as through continuing education and by other courses are presented.


The acoustics area has a very secondary place in the engineering degree studies in Uruguay. Only one early course on the physics of waves provides some idea of the broader aspects. Furthermore, some courses that used to provide the students a partial idea on some aspects of noise are no longer issued. Nowadays, most of the working market does not require high training on acoustics, but gradually noise issues are gaining importance and they are often addressed by professionals with no training in the subject. This fact has created in the collective imagination the idea that noise issues should be managed from an intuitive and empirical sense, rather than from a rigorous and formal approach, as would be desirable. Recently, environmental awareness has led people to include some basic aspects of environmental noise in the training of future civil engineers. Students who choose Civil Engineering on Hydraulics and Environmental will achieve further knowledge in their formation on environmental noise, its consequences, and management. On the other hand, the awakening of a “culture of the lawsuit” has generated the need of postgraduate activities to update and improve the skills of professionals interested in getting involved in this field.

2aED7. Acoustics at the Georgia Institute of Technology. Erica E. Ryherd, Mardi Hastings, and John Doane (Woodruff School of Mech. Eng., Georgia Inst. of Technol., Atlanta, GA 30332-0405, erica.ryherd@me.gatech.edu)

Acoustics at Georgia Tech spans multiple schools, including Mechanical Engineering, Electrical Engineering, Aerospace Engineering, Biomedical Engineering, Psychology, Music, Physics, Mathematics, and Architecture. The program began over 50 years ago and strengthened considerably in the 1960s and 1970s after Eugene Patronis, Ben Zinn, and Allan Pierce joined the faculty in the Schools of Physics, Aerospace Engineering, and Mechanical Engineering, respectively. Since then hundreds of students in acoustics have graduated and hold positions in academia and industry around the world. Currently the School of Mechanical Engineering has 11 academic faculty members with primary interest in acoustics and dynamics. Areas of research include architectural acoustics, psychoacoustics, noise control, environmental acoustics, sustainable systems design, underwater acoustics, bioacoustics, ultrasounds, active/passive control, fluid-structure interaction, nonlinear acoustics, acousto-optics, micromachined sensors and actuators, vibration of nonlinear and frictional systems, shock and vibration isolation, structural acoustics, wave propagation, and structural health monitoring. Masters and Ph.D. level programs are offered in addition to various undergraduate courses. The depth of knowledge at Georgia Tech facilitates a variety of collaborations, allowing students a multi-disciplinary education in the science and application of acoustics. Student interactions are further facilitated by a number of organizations on campus, including a recently established student chapter of the ASA.

2aED8. Description of the National Polytechnic Institute acoustics program. Sergio Beristain (ESIME, IPN, IMA, Lab Acoust., P. O. Box 12–1022, Narvarte, Mexico City, 03001, Mexico, sberista@hotmail.com)

At the National Polytechnic Institute in Mexico City there has been, in the last 35 years, the teaching of the major in acoustics within the Communications and Electronics Engineering BSc studies. Thanks to the vision of Eng. Tomas Guzman Cantu, and the support of all the lecturers and the students, there are now over 1000 students graduated in this area. In the last promotion, there are over 90
students about to graduate, most of them have gone to the audio business in recording studios and broadcast centers. However, there are also graduated people in the areas of noise, hospitals and clinics, architectural acoustics, machinery vibration and structures, animal bioacoustics, speech and audiolog, designing and evaluation of musical instruments, etc. Details of the studies are presented.

9:45
2aED9. Acoustics at University of Texas: History and current introductory course in physical acoustics. David T. Blackstock (M.E. Dept., UT Austin, 1 University Station C2200, Austin, TX 78712-0292)

As an academic discipline at the University of Texas, acoustics began in the 1930s under C. Paul Boner in Physics and Lloyd A. Jeffress in Psychology. World War II saw Boner and many physics graduate students go to Harvard for “war work,” largely in underwater acoustics. When the war ended, Boner returned and founded the Defense Research Laboratory, later named Applied Research Laboratories. Interest in acoustics grew in Physics during the postwar years but eventually waned in the 1950s and 1960s. Acoustical activity developed in the 1960s in the College of Engineering, chiefly Electrical Engineering and Mechanical Engineering. Today physical and engineering acoustics is a strong interdisciplinary program at Texas, with faculty in several departments in Engineering and still a vestige in Physics. In addition, much work on speech, hearing, and music is done in other parts of the University. Engineering features two basic courses in physical acoustics, Acoustics I and II, and five specialty courses, which are described in an accompanying paper. Here we concentrate on Acoustics I and II, which provide an introduction to propagation, reflection and transmission, refraction, normal modes, horn theory, propagation in stratified fluids, absorption and dispersion, waveguides, directional radiation, diffraction, and arrays.

10:00—10:15 Break

10:15
2aED10. Graduate acoustics at University of Texas at Austin. David T. Blackstock, Neal A. Hall, Mark F. Hamilton, Elmer L. Hixson, Preston S. Wilson (Cockrell School of Eng., The Univ. of Texas at Austin, 1 University Station C2100, Austin, TX 78712), Michael R. Haberman, Marcia J. Isakson, David P. Knobles, and Clark S. Penrod (The Univ. of Texas at Austin, Austin, TX 78713-8029)

While graduate study in acoustics takes place in several colleges and schools at the University of Texas at Austin (UT Austin), including Communication, Fine Arts, Geosciences, and Natural Sciences, this presentation focuses on the acoustics program in Engineering. The core of this program resides in the Departments of Mechanical Engineering (ME) and Electrical and Computer Engineering (ECE). Acoustics faculty in each department supervise graduate students in both departments. One undergraduate and seven graduate acoustics courses are cross-listed between ME and ECE. Instructors for these courses include staff at Applied Research Laboratories at UT Austin, where many of the graduate students in acoustics hold research assistantships. The undergraduate course, offered every fall, begins with basic physical acoustics and proceeds to draw examples from different areas of engineering acoustics. Three of the graduate courses are offered every year, a two-course sequence on physical acoustics and a course on acoustic transducers. The other four graduate courses, offered in alternate years, are on nonlinear acoustics, underwater acoustics, architectural acoustics, and ultrasonics. An acoustics seminar with invited speakers is held most weeks during the academic year, averaging over 10 per semester. The ME and ECE departments both offer a Ph.D. qualifying exam in acoustics.

Contributed Papers

10:30
2aED11. The use of information technology and communication in the teaching of acoustics in the National Polytechnic Institute. Jorge Becerra García, Patricia Lorens Ramírez Rangel, and Dagoberto García Alvarado (IPN-ESIME Zacatenco, Lab. of Psychoacoustics, Unidad Profesional “Adolfo Lopez Mateos”, Col. Lindavista, CP 07738, DF, Mexico)

The IPN-ESIME Zacatenco has been making major changes in the educational field, since the goal is that technological education is kept at the forefront of the learning processes of their graduates in the various schools of IPN. The ESIME with its acoustics specialty do not remain outside this fact and promotes the redesign of its processes of teaching each of the subjects to students in an era of a more competitive knowledge society. To achieve this, graduates of the specialty must have more elements of judgment in carrying out their professional activities. The institute promotes further that students graduate and leave with more modern learning tools.

10:45
2aED12. New methods for acquiring a Bachelor of Science level in the National Polytechnic Institute careers. Pablo Lizana and Itzala Rabadan (Acoust. Lab., ESIME, IPN, Antonia 15 15 San Jeronimo Lidice, DF, Mexico, plizana@ipn.mx)

The main problem for the Bachelor of Science (BSc) students at the National Polytechnic Institute (IPN) is the lack of the BSc certificate, which sets them at a disadvantage compared with students of other institutions of the same level, locally and abroad. After several years of discussion, the Ad- visory Board of the IPN Mexico was able to incorporate a new way for get- ting the certificate, in order to reduce that problem, but the experience has shown that even having the same quality as the traditional way, some stu- dents and lecturers show resistance to the change to the establishment. The main problem is the lack of time the students have in order to develop their thesis within the allocated time frame.

11:00
2aED13. A theoretical course in underwater acoustics based on the sonar equation. Marcia J. Isakson (Appl. Res. Labs., The Univ. of Texas, 10000 Burnet Rd., Austin, TX 78713)

Current underwater acoustics students require a theoretical knowledge of propagation, scattering, and target strength as well as practical knowledge of how these concepts are applied in the modern sonar system. A curriculum based on the sonar equation was developed at the University of Texas, which includes theoretical propagation methods such as ray theory and normal modes, important oceanographic phenomena, target and interface scat- tering, sonar signal processing, and basic sonar design. This curriculum is described in order to spur a discussion of teaching for the modern underwa- ter acoustics student.

11:15

The fact that the research on acoustics represents an important segment of science and a significant necessity of society is certainly commonplace among acousticians. In many situations, however, it is only marginally in- corporated into the curricula of universities. This is particularly the case in
the Brazilian educational and research institutions, mainly due to the lack of a comprehensive curriculum that accounts for basic education in acoustics, despite the enormous demand for acousticians in Brazil. To meet this demand, the Federal University of Santa Maria has begun, in the end of 2008, a dedicated 5-year undergraduate course in Acoustical Engineering to provide students with more than basic knowledge in acoustics, vibrations, and audio. Within the first four semesters the curricula are structured to provide fundamental knowledge for engineers with traditional engineering subjects such as math, physics, mechanics, electrical circuits, and the like, accompanied by disciplines concerning musical education for acoustical engineers (first to fourth semester) and issues related to human exposure to noise. Starting at the fifth semester education in acoustics, vibration, and audio is broadened each semester with disciplines like room acoustics, noise control, environmental acoustics, electroacoustics, numerical methods in acoustics, psychoacoustics, and so on. The talk will present the structure of the course as well as discuss challenges and difficulties.

Session 2aMU

TUESDAY MORNING, 16 NOVEMBER 2010


James W. Beauchamp, Chair

School of Music, Univ. of Illinois at Urbana-Champaign, Urbana, IL 61801

Contributed Papers

8:40

2aMU1. Modeling the piano string as a mass-spring chain. Junji Kuroda (Corporate R&D Ctr., Yamaha Corp., 203 Matsunokijima, Iwata-shi, Shizuoka-ken 438-0192, Japan; junji_kuroda@gmx.yamaha.com), Julius Smith, and Jack Perng (Stanford Univ., Stanford, CA 94305)

The mass-spring chain allows accurate modeling of vibrating strings in all three spatial dimensions. [Rowland and Pask, Am. J. Phys. (1999)]. This presentation describes striking one or more adjacent masses of such a string model with a forcing function corresponding to the piano hammer [http://www.ioc.ee/stulov/PUB.htm]. An advantage of striking a mass-string model in which each mass may move in three dimensions, is that coupling from transverse to longitudinal modes of vibration are naturally provided.

8:55

2aMU2. A stiff mass-spring-chain model for piano strings. Junji Kuroda (Corporate R&D Ctr., Yamaha Corp., 203 Matsunokijima, Iwata-shi, Shizuoka-ken 438-0192, Japan; junji_kuroda@gmx.yamaha.com), Julius Smith, Katarina Van Heusen, and Jack Perng (CCRMA, Stanford Univ., Stanford, CA 94305)

The mass-spring chain allows accurate modeling of vibrating strings in all three spatial dimensions. [Rowland and Pask, Am. J. Phys. (1999)]. However, such a model is for non-stiff strings. A stiff mass-spring string model is proposed, consisting of three or more cross-coupled parallel mass-spring chains.

9:10

2aMU3. Sound synthesis of the harpsichord pluck using a physical plectrum-string interaction model. Chao-Yu J. Perg (Dept. of Phys., Stanford Univ., 382 Via Pueblo Mall, Stanford, CA 94305; perng@stanford.edu), Julius O. Smith, and Thomas D. Rossing (Ctr. for Comput. Res. in Music and Acoust., Stanford, CA 94305)

The harpsichord, in a broad sense, refers to the family of plucked keyboard instruments. The sound of the harpsichord tone is unique and is not mistaken for other plucked string instruments, such as the guitar. The synthesis of harpsichord sound has been done previously [Valimaki et al., EURASIP J. Appl. Signal Process. 7, 934–948 (2004)], where excitation signals are extracted through recorded tones. In this work, we present a revision to the harpsichord plucked model that we proposed earlier [Chao-Yu J. Perg et al., J. Acoust. Soc. Am. 127, 1733(A)–(2010)], one which incorporates basic friction models. The harpsichord plectrum-string interaction model is used for the sound excitation, which is then implemented with a digital waveguide to produce the synthesized plucked string tone. Our harpsichord plectrum-string interaction model allows for controllability of physical parameters that allow for a range of expressiveness. The effects in which changing the physical parameters have on the synthesized pluck sound are compared and discussed.

9:25

2aMU4. Practical determination of acoustic parameters of the singing voice implemented in the interactive analysis software EVOCANTO. Felipe Orduña-Bustamante and Gisela Gracida Olvera (Grupo de Acústica y Vibraciones, Centro de Ciencias Aplicadas y Desarrollo Tecnológico, Universidad Nacional Autónoma de México, Mexico)

Practical methods of analysis with low computational complexity are presented for the determination of acoustic parameters from radiated sound pressure spectra of sustained sung vowels. Sound spectra are analyzed in order to obtain the fundamental vocal source frequency F0 and its harmonics, the corresponding musical pitch, the first few “formant” frequencies F1, F2, possibly up to F5, that are apparent from the harmonic contour, a nominal vocal tract shape derived from (F1, F2), a measure of formant-to-harmonic tuning of F1 and F2, indication of the presence of a singer’s formant, and a measure of the rate of vibrato. Real-time determination and display of these parameters are integrated into the singing voice analysis software EVOCANTO, a graphical interactive didactic computer program which is intended as a teaching aid for singing tutors and students. Examples of the use of EVOCANTO are presented for typical male and female singing voice registers.

9:40

2aMU5. Reconstructing individual monophonic instruments from musical mixtures using scene completion. Jinyu Han and Bryan Pardo (EECS Dept., Northwestern Univ., 2133 Sheridan Rd., Rm. 3-323, Evanston, IL 60208, pardo@northwestern.edu)

Monaural sound source separation is the process of separating sound sources from a single channel mixture. In mixtures of pitched musical instruments, the problem of overlapping harmonics poses a significant challenge to source separation and reconstruction. One standard method to resolve overlapped harmonics is based on the assumption that harmonics of the same source have correlated amplitude envelopes: common amplitude modulation (CAM). Based on CAM, overlapped harmonics are approximated using the amplitude envelope from the non-overlapped harmonics of the same note. CAM assumes non-overlapped harmonics from the same note...
are available and have similar amplitude envelopes to the overlapped harmonics. This is not always the case. A technique is proposed for harmonic temporal envelope estimation based on the idea of scene completion. The system learns the harmonic envelope for each instrument’s notes from the non-overlapped harmonics of other notes played by that instrument, wherever they occur in the recording. This model is used to reconstruct the overlapped harmonic envelopes for obstructed harmonics. This allows reconstruction of completely overlapped notes, yet does not require predetermined instrument models. Experiments show the proposed algorithm performs better than an existing system based on CAM when the harmonics of pitched instrument are strongly overlapped.

9:55
2aMU6. Periodicity series of initial transients of musical instruments for real-time implementation. Rolf Bader (Inst. of Musicology, Univ. of Hamburg, Neue Rabenstr. 13, 20354 Hamburg, Germany, R_Bader@t-online.de)

A real-time algorithm for initial transients of musical instruments displaying the periodicity changes of the system during the transient is presented as shown with blown instruments. When the system consists of a reed and an air column, the stand point of the reed is taken and only one equation formulates the action of the reed upon itself as delayed and damped via the air column. A general system variable is used which includes both the periodicity and the amplitude of the system. This is possible as a back impulse acting in-phase with the reed will not only cause an enlargement of the reed’s amplitude but will in accordance with this enlargement also lead to an enlargement of the reed’s periodicity. When using an exponential damping and solving the equation for the system parameter of the next time step, an equation similar to that of the logarithmic map appears. According to the system parameters, for the steady state, a whole bifurcation scenario occurs with stable, bi-stable, and chaotic motion representing a stable tone, a multithonic, or a blowing noise. When formulated for instruments with multiple reflection points, such as violins or guitars, depending on the delays and dampings, the system shows an astonishing stability over the whole playing range because of these multiple reflections.

10:10—10:20 Break

Invited Papers

10:20
2aMU7. Analysis and digital enhancement of "brassiness". Charles M. Cooper, Jonathan S. Abel (CCRMA, Dept. of Music, Stanford Univ., 541 Lasuen Mall, Stanford, CA 94305), and Nurgun Erdol (Florida Atlantic Univ., Boca Raton, FL 33431)

The distinctive “brassy” timbre of instruments such as the trumpet and trombone has been variously attributed to spectral brightening of the mouthpiece excitation and to nonlinear wave-propagation effects, primarily shock waves, whose existence and importance are the subject of ongoing investigation. In the research presented here, spectral brightening at higher amplitudes—even at levels insufficient to produce shock waves—is explained analytically in terms of the known amplitude-dependent propagation speed of sound. This spectral-enrichment effect is experimentally demonstrated by acoustic measurements, quantitatively analyzed in terms of phase modulation and the generation of harmonically related sidebands, and perceptually confirmed by digital simulation. Efficient discrete-time methods for implementing the desired signal-dependent delay or phase modulation, as well as shock formation, are described. This “brassification” effect can be used for synthesis of brass-like sounds or to increase the perceived brassiness of live or recorded instrument sounds.

10:40
2aMU8. Real-time computational audio scene analysis, transformation, and synthesis for active music listening. Sylvain Marchand (LaBRI - CNRS, Univ. of Bordeaux, 351 cours de la Liberation, 33405 Talence Cedex, France, sylvain.marchand@labri.fr)

Spectral models attempt to parametrize sound at the basilar membrane of the ear, thus permitting transformations closely linked to the perception. However, for high-quality real-time applications, these models require methods for a precise analysis and an efficient synthesis. When dealing with musical sound, that is, a polyphonic mix of non-stationary complex sounds, the main challenge is to extract these different sounds present in the musical mix. This can be done using (psycho)acoustical knowledge about the sound sources (computational auditory scene analysis approach). However, the quality is often not sufficient. When access to the compositional process is given, another option is to use some bits of this ground truth as additional information (informed analysis approach). This more precise analysis allows deeper transformations, and using psycho-acoustical considerations, efficient data structures, and algorithms, it is possible to re-synthesize the sounds from the model parameters in real time and with a high quality. This opens up new impressive applications, such as “active listening”, enabling the listener to interact with the sound while it is played. The musical parameters (e.g., loudness or spatial location) of the sound sources present in the musical mix can now be changed interactively.

11:00

Regarding the decomposition of an audio stream in elementary sound objects, the non-negative matrix factorization (NMF) is now one of the prominent tools that can be employed. For instance, in the context of an automatic music transcription application, the interest of the technique relies on its ability to use redundancies over a whole piece to hopefully be able to identify separated sound objects as musical notes. In this ideal case, the spectrogram factorizes into a matrix of inferred spectral atoms and a matrix of temporal activations associated to these atoms. However, the unconstrained version of the NMF does not guarantee that the resulting atoms are actually musical notes: the spectrum of a single note can be split into several atoms or, conversely, a single atom could represent more than one note. The work that will be presented here deals with parametric models designed to describe the spectral modifications that occur along the progress of a single audio event (e.g., a musical note) in order to obtain a more meaningful decomposition from a musical point of view.
We present a novel interface for selecting and isolating sounds from complex audio mixtures. Traditional interfaces in audio editors provide graphical representations of sounds through which a user might be able to visually identify elements of sounds in a mixture; they do not, however, facilitate object-specific editing (e.g., selecting only the voice of a singer in a song). The proposed interface uses audio guidance from a user in order to specify a target sound within a mixture. The user is asked to vocalize the desired target sound, and an automated process identifies and isolates the elements of the mixture that best relate to the user’s input. Such a tool makes the extraction of complex sources an easier process than using graphical tools or statistical source separation models.

This paper describes recent research on singing information processing [Goto et al., Proc. of IEEE ICASSP (2010)], which is music information research for singing voices. Singing is one of the most important elements of music since a great number of people listen to music with a focus on singing, especially in the case of popular music. The concept of singing information processing systems is broad and still emerging, but three important categories are singing understanding systems, music information retrieval systems based on singing voices, and singing synthesis systems. Singing understanding systems have been developed for various tasks such as synchronizing lyrics to vocal music, singer name identification, singing skill evaluation, creating hyperlinks between phrases in the lyrics of songs, and detecting breath sounds. Music information retrieval systems based on similarity of vocal melody timbre and vocal percussion, as well as singing synthesis systems for speech-to-singing synthesis and singing-to-singing synthesis, have also been developed. Common signal processing techniques, such as techniques for extracting vocal melody from polyphonic music recordings and modeling the lyrics by using phoneme hidden Markov models for singing voices, are used in these systems. [This research was supported in part by CrestMuse, CREST, JST.]

Noise: Environmental Noise in Cities I: Noise Surveys

Chair’s Introduction—8:30

Invited Papers

8:35

2aNS1. Noise impact in a residential development in Zapopan Jalisco, Mexico. Cesar Muñoz Aceves, Juan Jose Ortiz Garcia, Martha Esparza Osuna, and Viridiana Mendez Rodriguez (CUCEI, Universidad de Guadalajara, Blvd. Marcelino Garcia Barragan No. 1421, Guadalajara, Jal, C.P. 44430, Mexico)

Altagracia is a residential development in Zapopan Jalisco. Here noise due to stationary and mobile sources is an important environmental stressor, since the settlement is close to industry complexes and the major periphery road of the metropolis. As a way of getting knowledge about what noise sources contribute the most to the noise impact within the development, a noise environmental assessment was performed. 14 samples were taken to evaluate the contribution of mobile and stationary sources. Sampling sites were established as a result of a pilot experiment at which the highest noise levels were recorded. Data processing and analysis were performed taking into account national standards for stationary sources (NOM 081-ECOL-94), and mean values for the mobile sources, since there is not specific regulation for them. Results of the study, highlights mobile sources as the major contributor of noise impact within the residential area.
2aNS2. Study of noise, environmental quality, and health risks in an avenue of the city of Guadalajara, Jalisco, Mexico. G. Galindo Barragan, M. Orozco Medina, J. Garcia Velasco, and A. Gonzalez Hernandez (IMACH, Universidad de Guadalajara, Km. 15.5, Carretera Nogales, Las Agujas, Zapopan Jal., C.P. 45110, Mexico, morozco@cucba.udg.mx)

Unsuitable urban development plans prompt environmental problems that might seriously affect health such as stress, hearing impairment, and respiratory problems. According to the World Health Organization [WHO (2001)], air pollution is one of the leading causes of illness and reduced quality of life in general. This research examines the environmental quality in a critical area of Guadalajara City, taking into account physical and chemical parameters, as well as air microbiology, as an integrated approach to identify health problems in exposed population due to pollution. On this basis, the study suggests an attention proposal.

2aNS3. Environmental noise attached to an integral quality diagnostic on recreational public areas. Loyda Chavez Rubio, Martha Orozco Medina, Gabriela Hernandez Ramirez, David Hernandez Arellano, and Gerardo Ramirez Trejo (IMACH, Universidad de Guadalajara, Km. 15.5, Carretera a Nogales, Las Agujas, Zapopan, Jal., C.P. 45110, Mexico, morozco@cucba.udg.mx)

Environmental quality is one of the most important topics for the social well being and must be considered in order to have a healthy social system. Shopping malls have become an essential part of the daily life of Mexican society as recreational and leisure public areas as well as a strong source of employment and economic incomes. However, the shopping malls can also be a huge source of noise, urban waste, and traffic, among other impacts it can cause in the environment. The objective of this study is to recognize and analyze the environmental quality on recreational public areas, including the way these places are managed, and now they manage solid waste, water savings, energy consumption, and measuring the noise levels inside and outside of the building; the registered range was between 60 and 90 dB(A). Another analyzed aspect is the existence or absence of educational programs oriented to visitors. It is important to find applicable strategies to make these recreational areas a place with improved environmental quality.

2aNS4. Urban noise and transport as a base of environmental quality strategy. Martha Orozco Medina and Figueroa Montano Arturo (IMACH, Universidad de Guadalajara, Km. 15.5 Carretera Nogales, Las Agujas, Zapopan, Jal., C.P. 45110, Mexico, morozco@cucba.udg.mx)

In recent years, urban noise has become an essential element in the environmental analysis since the study allows for a support tool for the analysis of environmental quality in the cities. Additionally, residents of large cities are demanding more options for transportation improvements from the standpoint of energy efficiency, low emissions of pollutants, and strategic routes, among others. This proposal presents the results of a diagnosis of urban noise on an avenue with high traffic flow in the city of Guadalajara, and analyzed during the construction of a rapid rail transit preferential and once it is running. The results of the noise levels actually indicate that the entry of a fast and environmental friendly bus significantly reduces noise emissions and contributes to the acoustic quality of the avenue with the benefits it represents for the people.

10:15

2aNS5. Noise level over 40 dB. Carlos Barroeta, Floriberto R. Ortiz, Juan Francisco Novoa Colin, and Marisol Salinas Salinas (ESIME, IPN, Mexico City FDM 07038, Mexico, charroet@hotmail.com)

Noise in cities is one of the most important problems for the population of the world. Traffic and industry noise is possibly producing new illness; 40–90 dB is the range of tolerance value for the human ear; depending on the application, some noise was analyzed in the real world. These parameters should be observed in design tactics for getting a better environment policy. Corporations and citizens must work for less pollution and more opportunities for development.

10:30

2aNS6. Measurement time of urban noise in Montevideo (Uruguay) and Medellin (Colombia). Alice Elizabeth Gonzalez (Dpto. Ingenieria Ambiental, F. de Ingenieria / Udelar, J. Herrera y Reissig 565, CP 11300, Montevideo, Uruguay, aliceelizabetghonzalez@gmail.com) and William Giraldo (Politecnico Colombiano Jaime Isaza Cadavid, Medellin, Colombia)

The duration of ambient noise measurements is one of the susceptible aspects regarding the correct design of a large measurement campaign, for example, to build an urban acoustic map. Previous work in Montevideo and Medellin lead to determine the stabilization time of the measurements with the aim of achieving accuracies from ± 1 to ± 3 dB in relation to the value of Leq,1h and Leq,d (Leq of the daytime). Montevideo data were taken at five points of the city, with traffic densities (IMH, time average intensity) from 200 to 1700 vph. Medellin data were taken in two monitoring stations, each equipped with a fixed sound level meter. In one of the stations, the IMH value is 1880 vph, while in the other one is 2120 vph except Sundays, when it is reduced to approximately 950 vph. Measured LA,eq,1h and Leq,d values are compared with those obtained for the same period with data corresponding to samples of 10, 15, 20, and 30 min. In both cases, the representativeness of the measurements increases with the enlargement of measurement time. Despite their great differences, both cities admit similar durations for measurements of urban noise to obtain representative data.

10:45


To elaborate noise maps, national and international directives indicate that urban noise measurements must be carried out 4 m high, so as to obtain the map by the method of direct measurements, or for the calibration of mathematical models that predict sound levels. As a continuation of research plans to be carried out in the Acoustic and Lighting Laboratory of the Commission of Scientific Research, Province of Buenos Aires (LAL—CIC), and in the Acoustic and Electroacoustic Laboratory, Engineering Faculty of the University of Buenos Aires (LACEAC—FI UBA), and in order to estimate
the error that will be committed using the prediction method rather than the direct measuring method, urban noise measurements were taken in surrounding areas in a quick access to La Plata City, at different heights and different distances from the sound source.

11:00  
2aNS8. Noise propagation through open windows of finite depth into an enclosure. Caleb F. Sieck (Architectural Eng. Prog., Univ. of Nebraska-Lincoln, 243 Peter Kiewit Inst., 1110 S. 67th St., Omaha, NE 68182-0681) and Siu-Kit Lau (Univ. of Nebraska-Lincoln, Omaha, NE 68182-0681)

Predicting the insertion loss of an opening backed with an enclosed space is important for building noise control. Recent research in sound transmission through apertures of finite depth in infinite rigid baffles has included the effects of propagating and evanescent modes within the aperture in order to extend models to higher frequencies. The present study extends the model to the case of the aperture backed by a cavity as opposed to sound radiating into half-space. The role of coupling between the aperture modes, radiation modes, and cavity modes in the transmission was investigated. The results were compared to those of previous models which neglected the depth of the aperture and finite element modeling using COMSOL. Multiphysics. Comparisons show that the current model is effective at predicting the sound transmission loss through the aperture and the acoustic field within the cavity for an obliquely incident plane wave. By changing impedance conditions on the half-space side of the aperture and within the aperture, the model has been used to evaluate passive noise control techniques.

11:15  
2aNS9. Revaluation of urban sounds in Barcelona. De Gortari Jimena (Arquitectural School, Univ. Pol. Catalunya, Barcelona, Spain, jimena.de.gortari@gmail.com)

This paper summarizes the results of the ongoing Ph.D. thesis “The revaluation of urban sounds in the Gothic Neighborhood, Barcelona, Spain,” which will be presented by September 2010. Acoustic urban space quality can be read from multiple indicators such as the configuration of the built environment, the residents’ perception, and perceived sounds, among others. In order to find those indicators, several elements were put together on a single methodology, so they could be tested at certain public spaces within Barcelona’s gothic neighborhood. One of the main results is the importance given by the people to some of the analyzed areas as places to rest from the noise of the city, although these places exceed the recommended noise values. Furthermore, the variety of nice sounds that people perceive, despite the prevailing background noise during most part of the day, should be highlighted.

TUESDAY MORNING, 16 NOVEMBER 2010  
CORAL GALLERY 1A/2A, 8:30 TO 9:45 A.M.

Session 2aPAA

Physical Acoustics: Resonators and Cavitation

Thomas J. Matula, Chair

Applied Physics Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105-6698

Contributed Papers

8:30  
2aPAA1. Internal mapping of the normal modes of a liquid filled spherical resonator. Joel Mobley (Dept. of Phys., Natl. Ctr. for Physical Acoust., Univ. of Mississippi, University, MS 38677, jmobley@olemiss.edu), Jason L. Raymond, Wayne E. Prather (Univ. of Mississippi, University, MS 38677), and D. Felipe Gaitan (Impulse Devices, Inc., Grass Valley, CA 95945)

We report on the measurements of the pressure fields set up inside a water-filled steel shell over a spectral range (10-50 kHz) that includes 75 modes of the composite system. The 9.5-in.-diameter shell has a wall thickness to radius ratio of h/R = 0.158, which is beyond the thin shell limit. Hydrophone scans of the acoustic pressure fields in the fluid were performed, from which 46 modes of the composite resonator were identified. The modal frequencies are in good agreement with those predicted by Mehl for a fluid-filled shell of arbitrary wall thickness. The radial pressure profiles of the modes are shown to fit the expected spherical Bessel profiles for solutions of the Helmholtz equation under spherical symmetry. The results are interpreted in terms of the idealized rigid shell and empty shell limits.

8:45  
2aPAA2. Enhancement and suppression of the response of a water-filled spherical shell resonator due to the interaction of modes. Joel B. Lonzaga and Joel Mobley (Natl. Ctr. for Physical Acoust., The Univ. of Mississippi, 1 Coliseum Dr., University, MS 38677, jblonzag@olemiss.edu)

In this paper we report on the suppression and enhancement of modal amplitudes in a water-filled, spherical shell resonator as a function of sound speed in water. The resonator, which has a 1-in.-thick wall and a 9.5-in.-diameter, was driven by a transducer bolted to the external wall. The drive frequency was swept over specific resonant modes whose frequencies have different dependencies of the sound speed in the water. Theoretical analysis of the resonant modes revealed that the suppression or enhancement of the response occurs for specific values of the sound speed when the initially distinct resonant frequencies of two modes meet at a crossover point. Additional theoretical modeling shows that the suppression or enhancement is caused by the interaction between the two modes when their resonant frequencies intersect.

9:00  
2aPAA3. Inertial cavitation threshold dependence on static pressure. Kenneth B. Bader, Jason L. Raymond, Joel Mobley, Charles C. Church (Natl. Ctr. for Physical Acoust., Univ. of Mississippi, University, MS 38677, kbader@olemiss.edu), and D. Felipe Gaitan (Impulse Devices, Inc., Grass Valley, CA 95945)

It is well known from observations in the ocean, as well as limited experimental measurements and theoretical predictions for spherical bubbles, that the threshold acoustic pressure for generation of inertial cavitation increases with hydrostatic pressure. However, the precise nature of the relationship between hydrostatic and acoustic pressures at threshold has not been well studied. We report here the results of a series of rigorous experiments designed to better define this relation and investigate the nature of the nuclei responsible for the cavitation events observed. The hydrostatic pres-
sure dependence (up to 300 bars) of the inertial cavitation threshold in ultrapure water was measured in a radially symmetric standing wave field in a spherical resonator driven at a resonant frequency of 25.6 kHz. The results will be compared to an independent analysis of the particulate content of the liquid. [Work supported by Impulse Devices, Inc.]

9:15
2aPAa4. Bubble size distribution and wave attenuation in an ultrasonic cavitation field. Stéphane Labouret and Jacques Frohly (Institut d’électronique et Micro-electronique et Nanotechnologies, Département d’Opto-Acousto-Electronique, Université de Valenciennes, le mont Houy, F-59 313 Valenciennes Cedex 9, France)

The bubble size distributions of a cavitation bubble field in a standing wave have been determined with an electromagnetic method, for bubble radii from 1.2 µm to about 10 µm. Results will be displayed for 349 kHz and 1.1 MHz ultrasound frequencies in continuous mode. A stabilization of the size distribution by the wave attenuation in the bubble field is proposed for a cavitation regime of low activity at 349 kHz. Results in chopped irradiation mode are presented at both frequencies. Measurements in continuous irradiation in the presence of several concentration of sodium dodecyle sulfate will also be displayed.

TUESDAY MORNING, 16 NOVEMBER 2010
CORAL GALLERY 1A/2A, 10:15 TO 11:45 A.M.

Session 2aPab

Physical Acoustics: General Physical Acoustics I

Bart Lipkins, Chair
Mechanical Engineering, Western New England College, 1215 Wilbraham Rd., Springfield, MA 01119

Contributed Papers

10:15
2aPAb1. Coherent backscattering enhancement in cavities: Simple shape cavity revisited. Catheline Stefan, Galloï Thomas, Roux Philippe (LGIT, Grenoble Univ., 38000 Grenoble, France), Ribay Guillemette (CEA, LIST, Gif-sur-Yvette, France), and de Rosny Julien (Institut Langevin, Paris Univ., Paris, France)

Coherent backscattering effect (CBE) is classically introduced in disordered, random, or chaotic media. In this work, the attention is focused on simple parallelepiedic cavities since, contrary to a widespread idea, CBE can also be observed for a pure-tone source in a 1-D cavity. This approach is of two-fold interest. First, analytical computations predict a dimensional dependence of the coherent backscattering enhancement according to a (3/2)d law, d being the dimensionality of the cavity, that have not yet been compared to experiments. Second, it opens new ballistic interpretations for which each multiply reverberated path is associated with more (rectangle and parallelepiedic cavities) or less (1-D cavity) than one single reciprocal counterpart. This paper is the first of two; the second paper dealing with some impacts of symmetry on CBE.

10:30

Through experiments and simulations, some consequences of symmetry on coherent backscattering enhancement (CBE) in cavities are reported here.

First, CBE outside the source is observed (a) on a single symmetric point in a 1-D cavity, in a 2-D circular membrane, and in a 2-D symmetric chaotic cavity; (b) on three points in a rectangle; and (c) seven in a parallelepiped. Second, the existence of enhanced intensity lines and planes in 2-D or 3-D cavities is demonstrated. Third, we show how antisymmetry is responsible for the existence of a coherent backscattering decrement with a dimensional dependence of (12)d law, d being the dimensionality of the cavity.

10:45

The acoustical model of T. von Kármán and N. Moore is used to predict ballistic shocks produced by slender like projectiles traveling at supersonic speeds. This is a linear acoustics model which assumes the projectile as a linear distribution of point sources whose diameters vary according to the projectile shape profile. In this paper, an updated version of the T. von Kármán and N. Moore model which introduces nonlinear distortion of the N waves for describing shock overpressure and N wave duration is presented. Nonlinear distortions are introduced in the script in the form of the well known Whitham expression for N waves in the far field. Furthermore, the results of these computer simulations are validated using previously recorded waveforms obtained from real sniper weapons of different calibers.
A closed-form analytical solution for the acoustic scattering of a new class of Bessel beams, termed non-diffracting Bessel vortex beams of fractional type $\alpha$, is derived. This new class of Bessel beams preserves the same non-diffracting feature of conventional high-order Bessel vortex beams of integer order. The far-field acoustic scattering field by a rigid sphere centered on the beam axis is expressed as a partial wave series involving the real number $\alpha$, the scattering angle relative to the beam axis $\theta$, and the half-conical angle $\beta$ of the wave number components of the beam. Unlike the acoustic scattering properties of conventional high-order Bessel vortex beams of integer order [F. G. Mitri, Ann. Phys. NY, 333, 2840–2850 (2008); P. L. Marston, J. Acoust. Soc. Am. 124, 2905–2910 (2008)], both the acoustic forward and backward scattering of Bessel vortex beams of fractional type $\alpha$ by a rigid sphere do not vanish unless $\alpha$ becomes an integer number. The novel properties of non-diffracting Bessel vortex beams of fractional type $\alpha$ may lead to the development of an “acoustic blender” with possible applications in particle rotation, mixing, and manipulation. [Work supported by LANL-LDRD# X9N9.]

The shaping mechanism for anisotropic wind velocity and temperature fluctuations of the atmosphere and their effect on the long-range infrasound propagation are studied. Based on the model developed for the wind velocity and temperature fluctuations induced by atmospheric gravity waves in the middle and upper atmosphere, the acoustic fields from pulsed infrasound sources (such as ground-surface explosions and volcanoes) are calculated by a parabolic equation method at different ranges from the sources (up to 300 km). The scattering of infrasound signals by highly anisotropic wind velocity and temperature inhomogeneities existing in the stratosphere and lower thermosphere is shown to significantly contribute to the stratospheric and thermospheric arrivals of the signal.
2aSA2. Efficient prediction of broadband radiation from vibrating panels in the high-frequency range. Donald B. Bliss and Linda P. Franzoni (Mech. Eng., Duke Univ., Durham, NC 27708)

Broadband high-frequency panel radiation is shown to have simple directivity characteristics expressed analytically by a limited set of parameters. For subsonic structural waves, namely below coincidence, the far-field mean-square pressure is associated with radiation from the panel perimeter. For low structural damping, this edge radiation depends on the mean-square structural response of the panel and the reflection coefficients of structural waves at panel boundaries. The spatial transform of surface velocity can be expanded and inverted term-by-term in the radiating wavenumber region, giving an asymptotic result in physical space with singularity functions along the edges. The far-field is produced by monopole, dipole, quadrupole, etc., edge radiators with relative strengths that depend on surface-wave Mach number and wavefront orientation. Edge radiation cannot be explained simply in terms of uncanceled volumetric sources, as often stated incorrectly. The correct interpretation of the sub-coincidence radiation is provided in both physical space and transform space. High-frequency broadband radiation above coincidence is also examined. Then the panel behaves as a surface radiator and directional beaming is observed. The work gives insight into the role of uncertainty in boundary conditions, allowing the establishment of bounds on the radiated power and variation in directivity.

8:55

2aSA3. Underwater sound radiation by a layered infinite plate with discontinuity introduced by a signal conditioning plate. Jie Pan and Yanni Zhang (School of Mech. Eng., Univ. of Western Australia, Crawley, WA 6009, Australia, pan@mech.uwa.edu.au)

Adding a layer of elastic foam to a baseplate reduces underwater sound radiation from the vibrating baseplate and sound reflection of incoming sound. To increase the local signal to noise ratio of a detecting hydrophone placed in front of the foam, a signal conditioning plate of finite size is introduced between the foam and hydrophone. This paper considers the effect of the discontinuity by the signal conditioning plate on the general properties of the sound radiation of a layered plate. Both far field sound directivity and near field sound distribution are calculated to illustrate the properties. The scattering of structural waves in the elastic foam by the signal conditioning plate is used to explain the observed properties.

9:20

2aSA4. Vibrations, energies, and power flows in complex systems. Hongan Xu and Wen L. Li (Dept. of Mech. Eng., Wayne State Univ., 5050 Anthony Wayne Dr., Detroit, MI 48202, dw0588@wayne.edu)

A general numerical method, the so-called Fourier spectral element method, is described for the dynamic analysis of complex structures. When applied to structural problems, this method treats a complex structure as an assembly of fundamental structural components such as beams and plates. Over each component, the basic displacement is sought directly in the form of a modified Fourier series which is mathematically guaranteed to converge absolutely at every field or boundary point. The Fourier coefficients are considered as the generalized coordinates and determined using the Rayleigh–Ritz method. Since this method does not involve any assumption or simplification, it is broadly applicable to the whole frequency range usually divided into low-, mid-, and high-frequency regions. Additionally, because the system model is mesh- and grid-free, a smooth transition between the different frequency regions can be automatically accomplished by switching on/off the statistical processes and/or the spatial- and frequency-averaging features. This method is particularly suitable for mid-frequency dynamic analysis of complex systems. The relationships are investigated between component energies and power flows. It is shown that the spatial- and frequency-averaging processes may not be appropriate for the mid-frequency analysis because the important dynamic characteristics of a system tends to be eliminated.

9:45

2aSA5. Energy transport and statistics in the acoustics of large irregular structures. Richard L. Weaver (Dept. of Phys., Univ. of Illinois, 1110 W Green St., Urbana, IL 61801, r-weaver@uiuc.edu)

There is a long-standing presumption that the acoustics of large irregular structures are well approximated by a theory of diffuse spectral energy density, for which one may write balance and constitutive relations, neglecting wave phase. This is the basis for SEA. But this picture fails to predict numerous important phenomena (normal modes, level repulsion, enhanced backscatter, localization, retrieval of the Greens function, etc.) Such theories are furthermore derivable from first principles only in certain special limits, e.g., statistical thermodynamics in which the acoustic fields are thermal fluctuations. The question thus arises as to whether one can derive a more accurate and universal theory. In the non-dissipative limit (which can be relaxed a posteriori) one can impose an additional constraint on acoustic transfer functions; i.e., that they conserve energy. S-matrices must be unitary. This condition re-inserts phases that SEA neglects. It is shown that, in the case in which there is a single source/receiver, this imposition has a plethora of consequences, amongst them being normal modes, level repulsion, enhanced backscatter, and localization. We discuss attempts to extend this to the case of multiple sources and receivers.

10:10—10:25 Break

10:25

2aSA6. Interior holography. Earl G. Williams (Naval Res. Lab., 4555 Overlook Ave., Washington, DC 20375, earl.williams@nrl.navy.mil)

Interior spaces with vibrating walls embrace a broad set of structural acoustic problems (planes, trains and automobiles). In the hunt for efficient measurement schemes that provide details of both the acoustic and structural fields, inverse holographic methods are very promising. These methods use a surface of measured data and mathematically back-track the field in time and space filling in a 3-D volume around the array in the process and are particularly attractive since they can image the time averaged acoustic intensity vector,
localizing and quantifying surface sources. There is a score of holographic back-propagation methods in use today. In almost all cases single measurement surfaces are not accurate due to reverberation and dual conformal surfaces are needed to separate sound fields coming from forward and backward directions. The spherical array (open or rigid) is a noticeable exception, an accurate mathematical approach is obtained with only one surface of data. This paper presents an overview of the inverse problem in interior spaces paying particular attention to the rigid spherical array, discussing improvements that extend the frequency range and volume of study, and simple regularization methods to stabilize the ill-posed nature of the volumetric reconstructions. [Work supported by ONR.]

Contributed Papers

10:50

2aSA7. Analytical and experimental study on active control of structurally radiated sound from an elastic cylindrical shell. Guoyong Jin and Shuangxiao Shi (Dept. of Power and Energy Eng., Harbin Eng. Univ., Harbin 150001, China, jgy1822@yahoo.com.cn)

This paper presents an analytical and experimental study on active control of structurally radiated sound from an elastic cylindrical shell. An analytical model is developed for the active structural acoustic control (ASAC) of the cylindrical shell. The structural response of a cylindrical shell to the harmonic point force is derived by applying the modal supposition method. Expressions for the far-field radiated sound pressure and radiated sound power are derived by using the Kirchhoff–Helmholtz integral and the appropriate Green’s function. Both global and local control strategies are considered. The optimal control forces corresponding to each control strategy are obtained by using the linear quadratic optimal control theory. Numerical simulations are performed to examine and analyze the control performance under different control strategies. Results show that using point force as the control input of the ASAC system can achieve the global sound attenuation of the cylindrical shell at resonance frequencies. Better control performance can be obtained under the control strategy of minimization of the radiated sound power in terms of the radiated sound power. However, control spillover may occur at off-resonance frequencies with the structural kinetic energy minimization strategy. ASAC experiment was implemented, good agreement was observed between the numerical and experimental results, and successful attenuation of structural vibration and radiated sound was achieved.

1:10


There are several optical methods that can be used for imaging structural vibrations; however, these often require expensive equipment and have significant limitations on the stability of the structure. We review some simple methods for imaging deflection shapes using inexpensive off-the-shelf components and introduce some modifications that make these methods even simpler. A simplified arrangement for electronic speckle pattern interferometry as well as a novel method that relies on the inherent threshold of the detection mechanism will be discussed.

11:20

2aSA8. Power transmission characteristics of a U-shaped plate structures with elastically restrained edges including in-plane vibration. Yuehua Chen, Guoyong Jin, Jingtao Du, and Zhigang Liu (Dept. of Power and Energy Eng., Harbin Eng. Univ., Harbin 150001, China, chenyuehua05341@yahoo.com.cn)

An analytical study on the power transmission characteristics of a U-shaped configuration consisted of three rectangular plates is presented. Four groups of springs, to simulate the transverse shearing forces, bending moments, in-plane longitudinal forces and in-plane shearing forces separately, are distributed consistently along each edge of the model. With general boundary conditions of both flexural and in-plane vibrations taken into account by setting the stiffness of these springs, the double Fourier series solution to the dynamic response of the structure is obtained by employing the Raleigh–Ritz method. For model validation, the natural frequency and velocity response of the model are checked against existing literature results and the ANSYS data and good agreement is achieved. The influence of several relevant parameters on power transmission of the coupled structure is then studied in detail, including boundary conditions, coupling conditions, and locations of the external force. The results show that the power transmission of the structure can be significantly affected by altering the boundary condition without changing other parameters of the model. The location of the excitation will remarkably influence the power transmission. Once the excitation is imposed on the central symmetry point of the model, the power transmitted will show a symmetrical distribution. When the location deviates from the central symmetry point, the power circumscription occurs.

11:35

2aSA10. Chloride detection in cement based materials by ultrasonic interferometry. Eduardo Cortez-Ortiz, Andres Pech Perez, Prisciliano F. de J. Cano Barrita, and Juan A. Vazquez Feijoo (Dept of Eng., CIIDIR Unidad Oaxaca-IIPN, Hornos No. 1003, Sta. Cruz Xoxocotlan, Oaxaca, Mexico, eduardo-coo@hotmail.com)

Chloride ions play an important role in the corrosion of steel reinforcement in concrete structures. In northern countries, de-icing salts are a major factor in the deterioration of highways, and in many countries, chlorides from seawater exacerbate the deterioration process. In order to assess the chloride penetration as destructive, low spatial resolution methods are routinely used. A range of ultrasound based techniques are currently being used to characterize cement based materials. However, they are intended primarily to monitor changes in mechanical properties, cracking, position of stressing tendons, and corrosion of reinforcement. This paper will demonstrate the ability of an ultrasonic interferometry technique to monitor the capillary absorption of a 16.5% NaCl solution in cement paste and mortar cube samples. The results demonstrate that this technique is able to detect small changes in the properties of fluids penetrating the porous material. Future applications will include imaging of chloride penetration in cement based materials in laboratory and field conditions.
Speech Communication and Psychological and Physiological Acoustics: Perception: Processing, Training, and Modality (Poster Session)

Valeriya Shafiro, Chair

Communication Disorders and Science, Rush Univ. Medical Ctr., 1653 W. Congress Pkwy., Chicago, IL 60612

Contributed Papers

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

2aSC1. Cross-modal talker familiarity: The search for amodal lexical episodes. Kauyumari Sanchez and Lawrence D. Rosenblum (Dept. of Psych., Univ. of California, 900 Univ. Ave., Riverside, CA 92521)

Familiarity with lip-read (visual-only) speech from a talker facilitates perception of heard (audio-only) speech-in-noise from that same talker [L. Rosenblum et al., Psychol. Sci. 18, 392 (2007)]. This could suggest that stored speech episodes are composed of lexical items that retain amodal talker-specific characteristics. However, the Rosenblum et al. study used sentential material which differed between the lip-reading and speech-in-noise tasks. That study did not examine whether cross-modal talker facilitation was based on lexical items used in both tasks. The current experiment addressed whether amodal talker-specific characteristics are stored as lexical episodes by modifying Rosenblum et al.’s design. Subjects were first asked to lip-read words from a talker and were then asked to listen to words embedded in noise spoken by the same or a different talker. Preliminary evidence showed that speech-in-noise words were identified better when spoken by the same talker as the lip-reading task. In addition, preliminary results showed a lexical frequency by word interaction, where low-frequency old words (presented at training) were better identified than low-frequency new words (not presented at training) during the speech-in-noise task. These preliminary results suggest that cross-modal talker facilitation can be based on the retention of amodal lexical episodes.

2aSC2. The role of speech readability in the intelligibility of visual speech signals produced by cued speech transliterators. Jean C. Krause and Abby N. Bennett (Dept. of Comm. Sci. and Dis., Univ. of South Florida, 4202 E. Fowler Ave., PCD 1017, Tampa, FL 33620, jeankrause@usf.edu)

Although it is common practice for deaf individuals to use interpreters as a means of accessing spoken information, very little is known about the intelligibility of visual signals produced by interpreters [Kluwin and Stewart, Odyssey 2, 15–17 (2001)]. One of the first studies to evaluate interpreter intelligibility [Pelley, MS thesis, USF (2008)] recently examined 12 interpreters who use a speech-based visual communication mode known as cued speech. The interpreters, or transliterator, ranged in intelligibility from 52% to 90% (as evaluated by deaf individuals proficient in cued speech), and the accuracy of their cues (hand signals) accounted for only 26% of the variance in intelligibility. Because mouth movements are also part of the cued speech signal, this experiment examined the contribution of speechreadability to the intelligibility of the messages. Eight individuals who were unfamiliar with cued speech were presented with visual stimuli excised from the transliterated messages and asked to transcribe the stimuli. Speech readability of the visual speech signals was measured as the percentage of words correctly received. Results show speech readability is independent of accuracy and accounts for roughly 13% (on average) or more (when transliterator experience is controlled) of the variance in intelligibility. [Work supported by NIH/NIDCD Grant 5-R03-DC007355.]


The human voice is indicative of a variety of characteristics. It may be particularly influential in judgments such as of charisma. But what features of the voice are indicators of these characteristics? Osgood et al. (1957) suggested that a vast array of stimuli are judged by humans along the three dimensions of evaluation, potency, and activity. We tried to determine whether voices are also judged according to these three dimensions. Male and female judges rated 36 short (2 s) segments of voice segments from a variety of languages on Osgood et al.’s dimensions, using 9 different scales. Listeners were able to make their judgments quickly, even though the stimuli were short. Confirmatory factor analysis failed to produce the three factor structure from the listener’s judgments of the voices. Gender and language of the voices did not help clarify the structure of the judgments. Hence, this study demonstrates that human voices are unique stimuli that may be judged using a more complex framework.

2aSC4. Spectral modulation transfer function and vowel identification. Jillian Phillips, Sarah Wiese, Chang Liu (Dept. of Commun. Sci. and Disord., 1 University Station A1100, The Univ. of Texas at Austin, Austin, TX 78712), and David Eddins (Univ. of Rochester, Rochester, NY 14642)

Psychophysical and physiological studies have demonstrated selectivity for spectral modulation frequency in the auditory system, suggesting that auditory perception of complex spectra might be based on spectral modulation channels. However, an early study [Liu and Eddins (2008)] reported that vowel identification was significantly reduced by filtering in spectral modulation frequency domain, although there was no significant correlation between spectral modulation detection thresholds and vowel identification. The latter result may have been due to the small number of listeners (n = 5) in the previous study. Using a larger number of listeners, the present study measured spectral modulation transfer functions and vowel identification performance with and without spectral modulation filtering in 11 normal-hearing listeners. Vowel identification performance was significantly correlated with spectral modulation detection thresholds at 0.5 cyc/2c (i.e., higher spectral modulation detection thresholds associated with poorer vowel identification scores). In addition, average spectral modulation detection thresholds were significantly correlated with the change in vowel identification associated with filtering in the spectral modulation domain. Both of these results highlight the relation between the perception of sinusoidal spectral modulation and vowel identification.

2aSC5. Auditory processing and speech production in children with typical development and developmental phonological disorders. Chang Liu, Barbara Davis, and Craig Champlin (Dept. of Commn. Sci. and Disord., 1 University Station A1100, The Univ. of Texas at Austin, Austin, TX 78712)

The purpose of the study is to compare auditory processing of non-speech and speech sounds in children with developmental phonological disorders (DPDs) and children who are typically developing (TD). Basic auditory processing capacities of frequency selectivity, temporal processing, and speech perception for TD, and DPD children 4–6 years of age, were measured. Auditory filter bandwidth was examined at 4 KHz using the notch-noise method, and psychometric functions for gap detection and /b-/
identification were measured. Preliminary results showed that DPD children had higher boundaries and shallower slopes in the psychometric functions of gap detection and /b/-/p/ identification than TD children, indicating that DPD children needed longer duration gaps for detection and longer voiced onset time for voiceless stop identification than TD children. There was no significant difference in auditory filter bandwidth between the two groups of children, although DPD children showed slightly higher detection thresholds than TD children. These results suggest that frequency selectivity was comparable between TD and DPD children, although DPD children have a lower central auditory processing efficiency. The relationship between auditory perception and speech intelligibility will also be discussed.

2aSC6. Effects of training and exposure on the perception of spectrally degraded speech and environmental sounds. Valeriy Shafiro, Stanley Sheft, Kim Ho (Dept. Comm. Disord. & Sci., Rush Univ. Medical Ctr., Chicago, IL, valery_shafiro@rush.edu), and Brian Gygi (Veterans Affairs of Northern California, Martinez, CA).

Loebach and Pisoni [J. Acoust. Soc. Am. 123, 1126–1139 (2008)] demonstrated perceptual improvements following a brief training with spectrally degraded environmental sounds generalized to speech. The present study investigated whether benefit could also result from mere exposure over a longer time period. Participants received two pretests, one week apart, prior to beginning a week-long environmental sound training regimen, which was followed by post test sessions, separated by another week without training. Spectrally degraded stimuli, processed with a four-channel vocoder, consisted of a 160-item environmental sound test, word and sentence tests (CNC, SPIN), and a battery of psychoacoustic and cognitive tests. Results indicated significant improvements in all speech and environmental sound scores between the initial pretest and the last post-test, with performance increments following both exposure and training. For environmental sounds (the stimulus class that was trained), the magnitude of positive change that accompanied training was much greater than that due to exposure alone, with improvement for untrained categories roughly comparable to the speech benefit from exposure. Speech and environmental sound performance were differentially correlated with tests of spectral and temporal–fine-structure processing abilities, while working memory and executive function were correlated with speech, but not environmental sound perception. [Work supported by NIH.] 2aSC7. Detection of environmental sounds in informative backgrounds. Brian Gygi (Veterans Affairs Northern California Health Care System, Speech and Hearing Res., 150 Muir Rd. 151-I, Martinez, CA 94553).

In Gygi and Shafiro (2010), it was shown that the identifiability of environmental sounds presented in naturalistic scenes can be affected by the semantic content of the background scene: under certain conditions there is an incongruency advantage for sounds which are contextually incongruent, i.e., not semantically consistent with the background scene. Detection thresholds for the same set of sounds and scenes were obtained using adaptive tracking. The 28 background scenes and the last post-test, with performance increments following both exposure and training. 2aSC6. Effects of training and exposure on the perception of spectrally degraded speech and environmental sounds. Valeriy Shafiro, Stanley Sheft, Kim Ho (Dept. Comm. Disord. & Sci., Rush Univ. Medical Ctr., Chicago, IL, valery_shafiro@rush.edu), and Brian Gygi (Veterans Affairs of Northern California, Martinez, CA).

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2aSC8. The importance of syllable position and combination in the perception of birdsong by budgerigars (Melopothaciss undulatus). Erikson G. Neilans, Thomas E. Welch (Dept. of Psych., Univ. at Buffalo, 206 Park Hall, Buffalo, NY 14206, eneilans@buffalo.edu), Ross Maddox, Barbara G. Shinn-Cunningham (Boston Univ., Boston, MA 02215), and Micheal L. Dent (Univ. at Buffalo, Buffalo, NY 14206).

Identifying a target signal in a clamorous environment is crucial for any social species. The current study examines the importance of syllable position for birds identifying a target song embedded in a chorus of masking noise. Four budgerigar subjects were trained for food reinforcement to identify three separate target zebra finch songs from a chorus masker of multiple simultaneous zebra finch songs. Following training, the birds identified individual syllable and incremental syllable stimuli as either target or masker by pecking the corresponding key. The early syllables in these songs elicited more “target” responses compared to the later syllables, suggesting that the beginning syllables are more salient in the perception of birdsongs by these birds. Further, the sequential combination of syllables resulted in a gradual increase in target bird song responses, when syllables were added from beginning to end, indicating that as more of the song is presented, the ability to hear and identify the birdsong improves. These results indicate that the perception of birdsong by budgerigars follows similar rules as the perception of speech in humans. 2aSC9. The effect of reverberation on Quick Speech-in-Noise test and Revised-Speech Perception in Noise test scores in young adults with normal hearing. Holly A. Morlas and Elizabeth M. Adams (HAHN 1119 307 N. Univ. Blvd. Mobile, AL 36688-0002).

In typical, realistic listening environments, speech perception is often degraded by various degrees of both noise and reverberation. The effect of reverberation has previously been assessed with predictability measures, such as the speech transmission index (STI). [Houtgas et al. (1980); Steeneken and Houtgas (1980).] Currently, no clinical test measure of an individual’s speech perception ability incorporates both noise and reverberation. The aim of the present study is to provide a more real-world measure of speech perception performance with the inclusion of reverberation on two speech-in-noise tests. The Revised-Speech Perception in Noise Test [R-SPIN; Bilger et al. (1984)] and the Quick Speech-in-Noise Test [Quick-SIN] were administered to young adults with normal hearing. Results of the Quick-SIN and R-SPIN tests indicate that reverberation negatively impacts speech perception but is dependent upon other characteristics of the listening environment, such as the poor SNR, sentence stimuli lacking contextual cues, or both. Overall findings suggest that the inclusion of reverberation would provide a more realistic clinical tool of speech perception accuracy and validate patient complaints of poor speech perception in degraded listening environments. 2aSC10. Binaural speech intelligibility as a function of speaker angle and interaural cross-correlation of disturbing noise and reverberation. Laura Padilla and Felipe Orduña (CCADET, Universidad Nacional Autónoma de Mexico, Circuito Exterior S/N. Ciudad Universitaria, C.P. 04510, C.de Mexico, Mexico, laura.padilla@ccade.unam.mx).

Subjective intelligibility tests were carried out that demonstrate that binaural speech intelligibility under adverse acoustic conditions, such as noise and reverberation, improves when listening at an angle, and also depends on the interaural cross-correlation of the disturbance. Speech material was binaurally recorded in an anechoic chamber using an acoustic manikin, from a loudspeaker at different azimuth angles from −60 deg (left) to +60 deg (right) in 15-deg steps. The binaural speech material was contaminated with interaurally correlated and uncorrelated noise and reverberation, and was presented to listeners through headphones. Results indicate advantages of binaural speech intelligibility under adverse acoustic conditions when listening at an angle, relative to listening at 0 deg. Additionally, speech intelligibility improves when the disturbing reverberation has high interaural cross-correlation; while on the other hand, speech intelligibility improves when the disturbing noise has low interaural cross-correlation. 2aSC11. The relative contribution to speech intelligibility from consonants and vowels when using synthesized and naturally spoken sentences. Jeffrey J. DiGiovanni, Jessica M. Prewitt (Div. of Comm. Sci. and Discord., Ohio Univ., W218 Grover Ctr., Athens, OH 45701, digiovan@ohio.edu), and Daniel R. Moates (Ohio Univ., Athens, OH 45701).

Several studies have attempted to understand why intelligibility is maintained in sentences or words in which all consonants were replaced by silent filler words and consonants were added to the end of spoken words to identify the final consonant. Cole et al. (1996) found that when sentences were presented with only vowels or only consonants intact, participants performed significantly better in the vowel only condition. Others [Kewley-Port et al. 2319 J. Acoust. Soc. Am., Vol. 128, No. 4, Pt. 2, October 2010 2nd Pan-American/Iberian Meeting on Acoustics 2319]
To evaluate potential factors underlying the speech benefit obtained when combining acoustic and electrical stimulation, the perception of temporally interrupted dual-band speech (lowpass at 800 Hz and highpass at 1600 Hz) was investigated with HINT sentences quasi square-wave gated at rates ranging from 0.8 to 12.5 Hz. For the highpass band alone, intelligibility improved with gating rate, while for the lowpass band, performance was consistently poor at all rates. Concurrent presentation of both bands interrupted at the same rate led to significantly better performance that exceeded predictions based on probability summation. Similar performance benefits were obtained when the two bands were gated at different rates. Specific aspects of dual-band performance could be accounted for through detection theory with appropriate selection of set size for 1-of-n detection and cross-band correlation. However, no single parameter set led to a complete description of results. An alternative approach is to view the consequence of combining speech bands as an enhancement of definition of stimulus structure to allow for greater information transmission. A similar effect may contribute to the benefit obtained when combining acoustic and electric speech stimulation with acoustic elements providing structure through enhancement of signal coherence and segmentation. [Work supported by NIH.]

2aSC12. Speech intelligibility in speech noise: Single versus multiple talker training. Kristin J. Van Engen (Dept. of Linguist., Northwestern Univ., 2016 Sheridan Rd., Evanston, IL 60208, k-van @northwestern.edu)

Difficulty understanding speech in noise is a primary complaint for listeners with hearing loss. However, a person’s language experience can also affect the extent to which s/he is affected by noise when listening to speech. Non-native listeners, for example, have greater difficulty coping with noise than native speakers, and several studies have shown that native listeners are more detrimentally affected by interfering speech/babble in their native language versus a foreign language. Since experience-related factors can modulate the effects of noise on speech perception, the present study aims to identify training parameters that can enhance speech intelligibility in noise. Listeners were trained on English target sentences produced by either a single talker or by multiple talkers in the presence of English 2-talker babble, Mandarin 2-talker babble, or speech-shaped noise. These training conditions were compared by administering a common speech-in-speech post-test. It was predicted that target talker variability would lead to more generalizable learning. However, preliminary results indicate that single-talker training may be more beneficial than multiple-talker training, at least in some noise conditions. When the task of target speech perception is made easier, listeners may have more opportunity to learn to tune out competing speech noise.


Psychometric function characteristics of threshold and slope are well defined for monaural speech recognition performance for both normal hearing and hearing impaired listeners. Psychometric functions for dichotic speech recognition, however, have not been characterized in the literature. Dichotic listening refers to the simultaneous presentation of two differing stimuli to the ears. Percentage scores are typically obtained for both right and left ears, as well as an ear advantage score. The purpose of the present study was to characterize the psychometric function attributes of threshold and slope for dichotic speech recognition among young adults with normal hearing. Performance-intensity (PI) functions were obtained for three-pair dichotic digits and monosyllabic words for 26 normal hearing young adults 18–30 years of age. Results revealed significantly lower mean thresholds and significantly steeper mean slopes for dichotic digit versus dichotic word stimuli, reflecting the differences in difficulty between the two types of stimuli. Relative to monaural PI functions, dichotic PI functions exhibited higher thresholds and more shallow slopes. The differences in threshold and slope between dichotic and monaural PI functions likely reflect the greater difficulty associated with processing simultaneous yet different speech stimuli.

2aSC16. The effect of disrupting intonation patterns on the perceptual segregation of competing voices. Nandini Iyer (Air Force Res. Lab., 2610 Seventh St., Area B, Bldg. 441, WPAFB, OH 45433), Douglas S. Brungart (Walter Reed Army Medical Ctr., Washington, DC 20307), and Brian D. Simpson (Air Force Res. Lab., WPAFB, OH 45433)

Relative differences in fundamental frequency and prosodic features have been extensively studied as cues to voice segregation in multitalker listening tasks. However, little is known about the effects of disrupting the intonation characteristics of a talker within a single phrase. The current experiment measured color-number keyword identification scores with the coordinate response measure (CRM) sentences under two conditions: a “normal prosody” condition, where the target and masker phrases were spoken contiguously by a single talker; and a “disrupted prosody” condition, where the target and masking phrases were generated by concatenating together individual words that were originally spoken in different sentences. The results show that the elimination of normal prosodic cues had the greatest impact on performance when the target sentence was masked by a single CRM masking phrase that was presented at the same level as the target talker. The elimination of prosodic information had only a very slight impact on performance when the target talker was masked by more than one simultaneous talker, suggesting that the use of prosodic cues for segregation may be impaired when more than one interfering talker is present in the mixture.
2aSC17. Improving speech intelligibility for hearing aid users using a speech segregation algorithm. Srikanth Vishnuhboota and Carol Espy-Wilson (3180 A. V. Williams Bldg., Univ. of Maryland, College Park, MD 20742, srikanth@umd.edu)

Users of hearing aid devices encounter great difficulty in speech recognition in noisy environments, and show inferior performance in situations with interfering background noise or competing speakers as compared to the case of a quiet background. In such devices, a pre-processing stage, which first removes the background interference, should greatly boost the speech recognition performance in noisy environments. In this work, the performance of a previously reported speech extraction algorithm is investigated for the purpose of improving the speech intelligibility of hearing aid users. Recently, we performed segregation of speech mixtures using our speech extraction algorithm and showed improvement in the speech recognition performance of normal hearing listeners. In this work, we investigate the performance of our algorithm for users of hearing aids on the same perceptual task of recognizing speech in the presence of either interfering speech or noise. We focus on tuning the segregation algorithm for hearing aid users, and emphasize on the preservation of the appropriate envelope and fine structure cues which will prove most helpful in the recognition of the target speech signal.


The purpose of this research is to develop a software-based simulation platform for hearing aid and to provide an assistive tool for clinician. User can experience hearing aid function and choose suitable hearing aid via the hearing aid simulation platform. The platform contains microphone input, noise reduction, wide dynamic range compression (WDRC), and feedback cancellation. The main purpose of this study was to simulate noise reduction and feedback cancellation by using patent documents of GN ReSound Canta, Sonic Innovation Natura, and Oticon Syncro. We evaluated the noise reduction efficiency by using subjective quality evaluation (waveform, spectrogram, listening) as well as objective quality evaluation (segmental SNR, log-spectral distance). The result shows that speech quality, signal to noise ratio, and listening comfort were improved. We also simulated feedback signals with KEMAR feedback path impulse responses and showed the three simulated feedback cancellation algorithms in good performance. Finally, we combined three noise reduction methods and WDRC with the three feedback cancellation methods of the same companies. The result shows that the effects of noise and feedback were both reduced efficiently.

2aSC19. A simple hearing screener up to 17 kHz for use in speech perception research. Elizabeth Composeo (Dept. of Linguist., Northwestern Univ., 2016 Sheridan Rd., Evanston, IL 60208), Jenna Luque, Jessica Maye, and Jonathan Siegel (Northwestern Univ., Evanston, IL, 60208)

Speech scientists are well aware that hearing ability affects the perception of sound speech. However, hearing is rarely directly tested in studies of normal speech perception. Furthermore, standard audiological testing does not test above 8 kHz, although speech spans frequencies as high as 17 kHz. Given that amplification of frequencies above 8 kHz aids speech perception in hearing impaired subjects, higher frequencies appear to play a role in speech perception. Thus, speech perception research would be improved by incorporating direct tests of speech frequency hearing in all participants. We developed a simple, computer-based hearing screener for use in speech perception research. Using easily accessible equipment and simple calibration techniques, the screener tests bilateral hearing sensitivity between 250 Hz–17 kHz in about 5 min and is easily implemented by non-audiologists. A total of 111 ears from 56 participants were tested using both the simple screener and a full audiological workup using modified Bekesy tracking up to 17 kHz. The hearing screener identified hearing loss across a wide range of frequencies with a high level of both sensitivity (0.85395) and specificity (0.83118). This study demonstrates that a quick and simple hearing screener can provide an accurate assessment of hearing ability across the speech frequency range.

2aSC20. Priming effect of hearing-impaired listeners in speech perception. Woojae Han (Dept. of Speech and Hearing Sci., Univ. of Illinois at Urbana-Champaign, 901 S. Sixth St., Champaign, IL 61820, woojaehan@gmail.com), Riya Singh (Dept. of Elec. and Comput. Eng., Univ. of Illinois at Urbana-Champaign, Champaign, IL 61820), and Jont Allen (Dept. of Elec. and Comput. Eng. and Beckman Inst., Univ. of Illinois at Urbana-Champaign, Champaign, IL 61820)

Unlike normal hearing (NH) listeners, the speech perception of hearing-impaired (HI) listeners is much too complicated to be determined for certain. That is, HI listeners provide a varied profile of consonants that NH listeners record as 100% correct, with an error of less than 1/1000. Thus, we argue that there is an important factor that dominates the consonant errors in HI listeners: grouping errors, or priming. We hypothesize that when a cochlear dead region is present, a critical cue is lost, forcing priming errors (the listener cannot identify the sound within a group). We have tested over 50 sensorineural hearing-impaired ears for consonant recognition both in quiet and in noise. Sixteen English CV syllables were analyzed. Preliminary results showed that there is a subject based trend for either temporal or spectral cue errors when explaining the priming pattern of HI ears. In particular, /gal/, /bal/, /Zal/, and /Sal/ were reported as /da/, /pa/, /Sa/, and /Ta/, respectively. Although more data collection and analysis will be needed to prove the current findings, we conclude that HI listeners hear the consonant feature differently from NH listeners. [Work supported by NIH Research Grant.]

2aSC21. Quantification of context effects in phoneme and word recognition by non-native listeners. Lu Shi and Laura Koenig (Dept. Communi. Sci. & Disord., Long Island Univ., Brooklyn Camp., One Univ. Plaza, Brooklyn, NY 11201, lu.shi@liu.edu)

Non-native listeners have difficulty perceiving context cues in speech. The current study intended to quantify the gain in non-native listeners’ perception of English speech due to context by adopting the Boothroyd–Nittrouer model [J. Acoust. Soc. Am. 84, 101–114 (1988)], in which context effects are expressed via factors j and k. These factors are mathematically equal to the total number of components in a speech stimulus with no context, but are reduced when context is present. Ten normal-hearing, non-native and 10 native listeners participated in this study. To investigate the effect of acoustic-phonetic cues, 12 lists of CVC English words and 12 lists of nonsense words were employed. To investigate the effect of semantic-syntactic cues, three sets of 20 four-word sentences were employed. These sentences differed in the amount of semantic and syntactic cues contained in the stimuli. All stimuli were presented binaurally at 45 dB HL. A concomitant speech-weighted steady-state noise was added at 39 and 45 dB HL for words and sentences, respectively. The non-native listeners’ verbal responses were recorded digitally. It was expected that j and k are larger in value in non-native than native listeners, and that the magnitude of the difference is associated with the non-native listeners’ English learning history.

2aSC22. The interrelation between stimulus range and the number of response categories in vowel categorization. Titia Benders and Paola Escudero (Amsterdam Ctr. for Lang. and Kommun., Univ. of Amsterdam, Spuistraat 210, 1012 VT Amsterdam, The Netherlands, titia.benders@uva.nl)

The categorization of speech sounds is influenced by the preceding sound context. Specifically, listeners shift their boundary between two categories toward the values of the preceding sounds. Such boundary shifts are larger if listeners can respond with fewer categories than are actually present in the stimulus. This study investigates the influence of the stimulus range on boundary shifts between vowels and whether these shifts are smaller if listeners can choose from more categories than are present in the stimuli. The F1 continuum between Spanish /i/ and /e/ was presented to Peruvian listeners in three ranges. Half of the listeners responded with /i/ and /e/; the other half chose from the five Spanish vowels. Boundary shifts between /i/ and /e/ as a function of the stimulus range were observed, which were smaller when listeners could choose from five categories. In subsequent analyses, the influence of the preceding sound context appeared smaller for listeners that
could choose from five categories. Also, these listeners shifted their boundaries after more stimuli than listeners who responded with only two response categories. It is proposed that the effect of the number of response categories on boundary shifts is due to an automatic process in speech perception.

2aSC23. Presence of preceding sound affects the neural representation of speech sounds: Behavioral data. Kathy M. Carbonell (Dept. of Speech Lang. & Hearing Sci., Univ. of Arizona, 1131 E 2nd St., Tucson, AZ 85721, kathy@email.arizona.edu), Radhika Aravamudhan (George S. Osborne College of Audiol., Salus Univ., Elkins Park, PA 19027), and Andrew J. Lotto (Univ. of Arizona, Tucson, AZ 85721)

Traditionally, context-sensitive speech perception has been demonstrated by eliciting shifts in target sound categorization through manipulation of the phonemic/spectral content of surrounding context. For example, changing the third formant frequency of a preceding context (from /al/ to /ar/) can result in significant shifts in target categorization (from /ga/ to /da/). However, it is probable that the most salient difference in context is between the presence or absence of any other sound. The question becomes whether this large change in context has substantial effects on target categorization as well. In the current study, participants were asked to categorize members of a series of syllables varying from /ga/ to /da/ presented in isolation or following /al/, /ar/, /or/, or /la/. The typical shifts in categorization were obtained for /al/ versus /or/ contexts, but the shift in response between isolated presentation and any of the audible contexts was much larger (with more /da/ responses in isolation). A similar pattern of results was obtained when the audible context was broadband noise. Additionally, slopes of categorization functions were less steep for isolated syllables than when preceded by speech. These results suggest that isolated syllables may be neurally encoded differently than the same syllables in context.

2aSC24. Presence of preceding sound affects the neural representation of speech sounds: Frequency following response data. Radhika Aravamudhan (George S. Osborne College of Audiol., Salus Univ., 8360 Old York Rd., Elkins Park, PA 19027, raramavudhan@salus.edu), Kathy M. Carbonell, and Andrew J. Lotto (Univ. of Arizona, Tucson, AZ 85721)

A substantial body of literature has focused on context effects in speech perception in which manipulation of the phonemic or spectral content of the preceding sounds (e.g., /al/ versus /ar/) result in a shift in the perceptual categorization of a target syllable (e.g., /da/ versus /ga/). In a previous study utilizing the frequency-following response (FFR) to measure neural correlates of these context effects [R. Aravamudhan, J. Acoust. Soc. Am. 126, 2204 (2004)], it was noted that the representation of target formant trajectories were much weaker when the stimulus was presented in isolation versus following some type of context. To examine this effect explicitly, a series of syllables varying from /da/ to /ga/ was presented to listeners either in isolation or following the syllables /al/, /ar/, or /or/ (with a 50-ms silent gap between context and target). FFR measures were obtained from EEG recordings while participants listened passively. The resulting narrow-band spectrograms over the grand averages demonstrated that the target formant presentations were much weaker or absent when presented in isolation. This counterintuitive result may indicate the importance of context for robust encoding of important spectral features in speech and provide a cautionary note on the use of isolated syllables in auditory testing.

2aSC25. The high redundancy of speech sounds in sentential context. Christian E. Stilp (Dept. of Psych., Univ. of Wisconsin, 1202 W. Johnson St., Madison, WI 53706, cestilp@wisc.edu)

Readers are exceptionally good at guessing which letter comes next in printed text, even with little contextual information. This was quantified by Shannon [Bell Syst. Tech. J., (1948); Bell Syst. Tech. J., (1951)], who demonstrated that printed English text is approximately 50% redundant (predictable). Here, redundancy of American English speech was examined in a modified “guessing game” paradigm, as listeners guessed which speech sound came next throughout an entire sentence. Four TIMT sentences were presented to 19 listeners, and guesses were made by clicking buttons on a computer screen corresponding to each consonant and vowel sound in American English. Listeners received feedback after each guess and heard each sentence (up to their most recent correct guess) as often as they wished. Numbers of guesses were at chance levels at sentence and word onsets, and were much better than chance with the addition of phonemic and sentential context. Equivocal speech sounds predict 0% redundancy (chance performance) and entropy of 5.29 bits/sound. Listener performance demonstrates a mean sentence redundancy of 40% (3.18 bits/sound), within a narrow range of 36–43. Behavioral results will be evaluated with reference to simple phoneme frequency and transitional probabilities between phonemes. [Work supported by NIDCD.]
Pre-lexical representations of speech sounds have been shown to change dynamically through a mechanism of lexically driven learning. [Norris et al. (2003)] Here we investigated whether this type of learning occurs in native British English (BE) listeners for a word-final stop contrast which is commonly de-voiced in Dutch-accented English. Specifically, this study asked whether the change in pre-lexical representation also encodes information about the position of the critical sound within a word. After exposure to a native Dutch speaker’s productions of de-voiced stops in word-final position (but not in any other positions), BE listeners showed evidence of perceptual learning in a subsequent cross-modal priming task, where auditory primes with voiceless final stops (e.g., [sɪt], “seat”) facilitated recognition of visual targets with voiced final stops (e.g., “seed”). This learning generalized to test pairs where the critical contrast was in word-initial position, e.g., auditory primes such as [zaun] (“town”), facilitated recognition of visual targets like “down”. Control listeners, who had not heard any stops by the speaker during exposure, showed no learning effects. The results suggest that under these exposure conditions, word position is not encoded in the pre-lexical adjustment to the accented phoneme contrast.

TUESDAY MORNING, 16 NOVEMBER 2010

Session 2aSP

Signal Processing in Acoustics: Signal Processing for Speech Recognition

Jose Luis Oropeza Rodríguez, Cochair
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Invited Papers

7:45

2aSP1. Simulation of a model of the basilar membrane in two dimensions for Spanish vowels. Mario Jiménez Hernández, M. en C., José Luis Oropeza Rodríguez, and Sergio Suárez Guerra (Res. Lab. of Digital Signal Processing, CIC, IPN, Juan de Dios Batis esq. Miguel Othon de Mendizabal s/n, P.O. 07038, Mexico)

The inner ear has the cochlea as the principal element; this is a biological element in the form of a snail, within which the mechanical energy is converted into electrical energy. This process is realized by the inner ear cells on the basilar membrane. This membrane response is different for different frequencies of excitation; the result of this process is the human audition. This paper shows a simulation of a model in two dimensions of the basilar membrane and its characteristic response when excited by the two first formants of Spanish vowels; these formants are obtained by an analysis by mixtures of Gaussians.

8:05

2aSP2. Speaker recognition infrastructure for legal context in Mexico City. Felipe Rolando Menchaca García, Dr (IPN, Morelos No. 37, Taxco de Alarcon, Guerrero FDM 40200, Mexico, fmenchac@ipn.mx)

Speaker recognition is actually a mature technology; it is part of the alternatives considered highly reliable for biometric identification, control access of computer services, and so on. However, applications to legal ambient, particularly as part of juridical processes, are still problematic, particularly in an ambient like Mexico City. High-quality infrastructure to determine with high precision whether a voice recording comes from a specific person is a topic which has to be regulated. Expert studies of this kind have to be regulated. This paper is to fulfill the need to develop a local regulation of infrastructure characteristics and standard test methods for this topic. Standard test has to take into account local language mainly.

8:25

2aSP3. Design and evaluation of reference populations with forensic purposes. Claudia Rosas (Instituto de Lingüística y Literatura, Facultad de Filosofía y Humanidades, Universidad Austral de Chile, Campus Teja S/N, Casilla 567, Valdivia, Chile) and Jorge Sommerhoff (Instituto de Acústica, Valdivia, Chile)

The present work analyzed the effect of sex, noise, vocal register, and channel on the performance of Batvox, the automatic speech recognizer employed by the Forensic Laboratory of the Investigative Police of Chile. This study transferred and applied the methodological and results of a previous and recently finished research project [Regular Fondecyt No. 1,070,210, funded by the National Commission for Scientific and Technological Research of the Government of Chile] to develop a model for the generation of speech refer-
ence populations with forensic purposes, which incorporated a set of dialect and environmental variables not considered in the selected samples currently integrating their databases. The information supplied provided methodological criteria to develop an optimized database for the biometric identification of speakers in Chile and to improve the performance of the systems applying them.

8:45


Automatic speech recognition (ASR) represents a great expectative in communications systems where interaction between computers and humans is present. In the last three decades it has been grounded. The use of computer techniques helps people who speak in a native language or with physical limitation to make transactions, leave messages, obtain information, or control some device using voice expressions. This work shows results obtained when using different techniques of ASR into the Náhuatl language. Náhuatl language is a native language spoken in Mexico and is the most important because there are many people speaking it. Náhuatl language has around 49 variants. Preserving language is important because language is the most efficient means of transmitting a culture and it is the owners of that culture who lose the most when the language dies. Mel cepstrum coefficients, vector quantization, and hidden Markov models of continuous density were used. The experiments reported 99% accuracy in ASR on Náhuatl numerals corpus employed. This work represents the first step for implementing an automatic speech recognition translator into Spanish from the Náhuatl language, with closed sentences.

9:05

2aSP5. Voice analysis for a cochlear implant. Jorge Becerra Garcia, Joel Flores Martinez, and Patricia Lorena Ramirez Rangel (Lab. of Psychoacoustics, IPN, ESIME Zacatenco, Unidad Profesional Adolfo Lopez Mateos, Col. Lindavista, C.P. 07738, DF, Mexico)

In the laboratory of acoustics in the ESIME, IPN, Mexico, a project for the design of a cochlear implant is under development. The most important part of this project is the processing of acoustical signals for the recognition of voice. The main objective is an individual analysis of perception to every participating person in an appropriate manner when such signals reach the auditory system.

9:25


One of the most important aspects related with automatic speech recognition (ASRs) systems is to find a set of characteristics that represent speech signal; nowadays, LPC, CLPC, cepstrum, MFCC, LFCC, and Melspec, among others, have been used to solve that problem. Likewise, analyzing the audible behavior of the ear has shown to have good performance in comparison with those methods that consider the pronunciation as element to obtain speech signal representation. In this paper an analysis related with Gaussian wavelets representation based on Bark and Mel audible representations is presented. For these wavelets, a Gaussian function represents the basis decomposition function; further, we need a modulator to displace the function associated with each of the scales of analysis to the central frequency into Bark and Mel audible representations. At the same time, we use the variance parameter of the Gaussian function to adapt its bandwidth to the Bark and Mel audible representations. Finally, we show a comparison between two alternatives in time and frequency representations.

9:45


The underlying assumption for spectral/temporal features for use in automatic speech recognition is that the frequency resolution should be emphasized in relation to temporal resolution. Accordingly, Mel frequency cepstral coefficients are typically computed using an approximately 25-ms frame length with a 10-ms frame spacing, and using 3–5 frames to represent temporal derivative information. In phone recognition experiments based on the TIMIT database using discrete cosine transform coefficients for spectral information and discrete cosine series coefficients for their temporal evolution, substantially higher phone accuracies were obtained with much shorter frame lengths (8 ms), much shorter frame spacings (2 ms), and much longer time intervals for capturing spectral tracks (on the order of 500 ms). Experimental results with various conditions are given for phone recognition using the TIMIT database. The implications of these results are that spectral/temporal evolution features, emphasizing the temporal aspects, are of great importance for automatic speech recognition.

10:05—10:30 Break
Speech enhancement can be achieved in temporal, spectral, and spatial domains. For practical use, spectral subtraction is a popular filter in the spectral domain, and beamforming is a representative spatial filter. However, the former assumes stationary background noise, which should be known or estimated, and the latter requires arrival directions on sound sources. The author has proposed a robust approach to the simultaneous estimation of both a direction-of-arrival (DOA) of a target signal and an amplitude spectrum of background noise [Mizumachi, Proc. 158th ASA meeting]. The proposed method has also given confidence for each DOA estimate. In this paper, a spatial filter is adaptively designed and is integrated with a spectral filter in the adaptation process on spectral-spatial space. The confidence for each DOA estimate plays an important role for designing a reasonable and efficient spectral-spatial filter under time-varying acoustic scenes. In other words, simultaneous adaptation can be achieved time by time based on the reliability of each DOA estimate for the 2-D spectral-spatial filter. [Work supported by NEDO, Japan.]

Subjective and/or objective measurements of speech quality are important in benchmarking speech enhancement algorithms. Subjective measures include ratings of speech quality by listeners, whereas objective measures compute a metric based on the clean and enhanced speech samples. While subjective quality ratings are the “gold-standard,” they are also time- and resource-consuming. An objective metric that correlates highly with subjective data is attractive, as it can act as a substitute for benchmarking and fine-tuning the noise reduction algorithms. In this paper, the performance of several noise reduction algorithms for wideband (50–7000 Hz) telephony application was evaluated both subjectively and objectively. A custom wideband noise reduction database was created that contained speech samples corrupted by different background noises at different signal to noise ratios and processed by seven different noise reduction algorithms. Speech samples in the database were subsequently rated by a group of 32 listeners with normal hearing capabilities. Several objective metrics including log-likelihood ratio, weighted spectral slope, PESQ, and the loudness pattern distortion (LPD) measure based on the Moore–Glaserberg auditory model were used to predict the subjective ratings. Results showed that the subjective ratings were highly reliable and the LPD metric correlated the best with subjective ratings of enhanced wideband speech.

Head-related transfer functions (HRTFs) are generally large datasets and thus a constraint for real-time applications. We propose a method to reduce redundancy and compress the datasets. In this method, HRTFs are transformed into time domain head-related impulse responses (HRIRs) and compressed by conversion into autoregressive (AR) filters. The AR coefficients are calculated using Prony’s method and the order is determined using the minimum eigenvalue of the HRIR covariance matrix. Such filters are specified by a few coefficients which can generate the full HRIRs. Next, Legendre polynomials (LPs) are used to compress the AR filter coefficients. LPs are derived on the sphere and form an orthonormal basis set for spherical functions. (Higher-order LPs capture increasingly fine spatial detail; the number of LPs needed to represent an HRTF, therefore, is indicative of its spatial complexity.) The results indicate that compression ratios can exceed 95% while maintaining an error of less than 10% in the recovered HRTFs. [Work funded by the EU framework program 7 under Contract No. IST 215370.]
presence of double-talk condition. The typically used DTD solutions are often a compromise between high efficiency and affordable computational cost. The algorithm introduced by the authors is based on a novel principle related to the so-called semi-fragile watermarking, making it possible to achieve satisfactory performance with limited processing burden. A hidden signature is embedded into the arriving far-end speaker signal just before it is replayed to the near-end speaker. Consequently, the presence of that signature is detected in the signal recorded by the near-end speaker.

Tuesday morning, 16 November 2010

Session 2auWa

Underwater Acoustics: Scattering from Boundaries

Dajun Tang, Chair

Applied Physics Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105-6698

Contributed Papers

8:00 2auWa1. Scattering from an arbitrarily shaped rough interface embedded in heterogeneous fluids. Dajun Tang and Darrell R. Jackson (Appl. Physics Lab, Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, djtang@apl.washington.edu)

The general formula based on first order perturbation theory for sound scattering from a rough interface separating two fluid media is derived. Possible areas of application of the formula are scattering from buried, uneven sediment rough interfaces and reverberation in a waveguide with changing water depth. Each of the two media is allowed to have space-dependent sound speed, density, and attenuation coefficient. The rough interface is assumed to be comprised of two components, a slowly varying, arbitrarily shaped surface considered as the unperturbed background, and a small roughness component superimposed on the slowly varying surface. The formula is intended for both the scattered field back to the medium where the incident wave originates and the penetration field. Properties of the formula are discussed, including reciprocity and symmetry with respect to the scattered and penetration fields. Numerical examples are presented for several scenarios which are of common interest to the underwater acoustics community.

8:15 2auWa2. Three-dimensional scattering from the ocean surface using finite elements. Sumeedh M. Joshi and Marcia J. Isakson (Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758)

Scattering from the ocean surface is a major parameter in propagation, reverberation, and coherence length models for shallow water waveguides. However, there are few models that quantify the effects of out-of-plane scattering at the ocean surface. In this work, a finite element model is used to quantify the effects of out of plane scattering by comparing 2-D models to those in three dimensions for given realizations of the ocean surface. The ocean surface roughness is described by an ocean surface spatial spectrum and the scattering of a Gaussian tapered plane wave is considered. Results for several different ocean surfaces are compared to the Kirchhoff approximation to determine its range of validity. [Work sponsored by the Office of Naval Research, Ocean Acoustics.]

8:30 2auWa3. A model for under-ice-target acoustic scattering. Garner C. Bishop (Naval Undersea Warfare Ctr., Newport, RI 02840)

A model is described that has been developed to calculate the scatter of a high-frequency acoustic pulse that originates from a stationary source and is incident on and is scattered from the under-ice surface characteristic of the Arctic and from a stationary target. The under-ice surface is modeled by an ensemble of ice keels, and ice keels are modeled by an ensemble of ice blocks with rectangular facets. The Helmholtz–Kirchhoff integral in the Kirchhoff approximation is used to calculate the scatter from an ice facet. The scatter from an ice keel is given by a coherent sum of the scatter from all the ice facets. A T-matrix is used to calculate the scatter from the stationary target. The first order terms in the multiple scattering process between the target and the ice surface are also calculated, i.e., the scattering from the target of the free field scatter from ice facets and the scatter from ice facets of the free field scatter of the target. Numerical results show the extent to which the scatter from the target in the under-ice environment differs from its free field scatter.

8:45 2auWa4. Reverberation and backscatter at very small grazing angles. Kevin L. Williams and Dajun Tang (Appl. Physic Lab, Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, williams@apl.washington.edu)

Reverberation in shallow waters can be generally considered as a combination of two-way forward scatter and a single backscatter. With increasing range, reverberation consists of backscattered signals at ever smaller grazing angles. First-order perturbation theory is a suitable choice to model boundary roughness scatter if the rms roughness height is small compared to the acoustic wavelength. With decreasing grazing angle, the first-order theory works even better. To apply the first-order approximation, a zeroth-order problem is assumed known, which is often conveniently chosen to be the case where the boundary is flat. However, if the boundary has a low-wavenumber component, the flat boundary assumption for the zeroth-order field can result in modeling error. To quantitatively assess such error, perturbation results are compared to exact numerical solutions using COMSOL. It is found that such errors are found when grazing angles are on the order of a few degrees, because at such small grazing angles the true local grazing angle is sensitive to variations of the local boundary slopes.

9:00 2auWa5. Convolving the impulse response of ocean reverberation. Henry Weinberg (23 Colonial Dr., Waterford, CT 06385) and Ruth E Keenan (SAIC, Mashpee, MA 02649)

The comprehensive acoustic system simulation, a range-dependent standard model for predicting ocean reverberation, has the ability to simulate complex pulses. However, the method is inefficient if the acoustic system is aimed at numerous directions for various pulse shapes. A recent capability uses the standard approach to determine a numerical impulse response as a function of time and azimuth. The impulse response is then convolved with the impulse signature of the double-talk speaker's talkspurts, precisely identifying the double-talk periods. The use of echo-hiding watermarking method allows fulfilling the algorithm requirements. The results of performed experiments prove the good performance of the proposed DTD solution. [Research funded by the Polish Ministry of Science and Higher Education within Grant No. PBZ-MNiSW-02-H-2007.]
a complex pulse in order to obtain the actual response. Standard and convolved responses agree for the simple test cases that were tried. The standard approach is faster for one look direction and pulse shape. Convolution is an order of magnitude faster for realistic applications. At first glance, the impulse response appeared to be a significant advantage to the standard approach, but theoretical issues have not been resolved to the authors’ satisfaction. Some methods, including the one described here, convolve acoustic pressure squared. Others convolve acoustic pressure. References tend to quote basic convolution theory or require a significant background in signal processing. The former is probably an oversimplification. The latter is significantly more complicated than standard convolution.

9:15
2aUWb1. Simultaneous Bayesian localization of multiple sources in an uncertain ocean environment. Stan E. Dosso and Michael J. Wilmut (School of Earth and Ocean Sci., Univ. of Victoria, Victoria, BC V8W 3P6, Canada)

This paper considers simultaneous localization of multiple acoustic sources when properties of the ocean environment (water column and seabed) are poorly known. A Bayesian formulation is developed in which the environmental parameters, noise variance, and locations and complex strengths (amplitudes and phases) of multiple sources are considered unknown random variables constrained by acoustic data and prior information.

9:30
2aUWb7. An airborne broadband acoustic method to measure the characteristics of a dynamically rough, air-water interface. Andrew Nichols (Dept. of Eng., Univ. of Bradford, Richmond Rd., Bradford BD7 1DP, UK, a.nichols2@bradford.ac.uk), Keith Attenborough, Shahram Taherzadeh (Acoust. Res. Group, The Open Univ., Walton Hall, Milton Keynes MK7 6BJ, UK), and Simon J. Shepherd (Univ. of Bradford, Richmond Rd., Bradford BD7 1DP, UK)

An acoustic method to measure the characteristics of a dynamically rough interface is presented. The theoretical foundation for this method has been adapted from outdoor sound propagation research whereby the boundary roughness is represented by an effective admittance. This model is used to calculate the excess attenuation of a broadband pulse emitted at near grazing incidence by a point source over a dynamically rough air-water interface. The excess attenuation is compared to that measured with an array of microphones. The effects of the water surface pattern on the recorded excess attenuation spectrum and its relation to the spatial correlation function for the dynamically rough interface are then analyzed. A Newton–Raphson root finding method is used to determine the acoustic admittance from the measured excess attenuation data in the frequency range between 0 and 20 kHz. The relationship between the mean roughness height, its standard deviation, and the real and imaginary parts of the admittance are obtained.

TUESDAY MORNING, 16 NOVEMBER 2010

Session 2aUWb

Underwater Acoustics: Acoustic Localization

Brian T. Hefner, Chair

Applied Physics Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105-6698

Contributed Papers

10:15
2aUWb1. Simultaneous Bayesian localization of multiple sources in an uncertain ocean environment. Stan E. Dosso and Michael J. Wilmut (School of Earth and Ocean Sci., Univ. of Victoria, Victoria, BC V8W 3P6, Canada)

Two approaches are considered for estimating source locations and strengths. The first approach, referred to as localization, maximizes the posterior probability density (PPD) over all parameters using an adaptive hybrid optimization. The second approach, referred to as marginalization, integrates the PPD to produce marginal probability distributions for source positions and strengths, which quantify localization uncertainties. In this approach, 2-D Gibbs sampling is applied to source ranges and depths, and Metropolis–Hastings sampling is applied in principal-component space for environmental parameters. In both approaches, closed-form maximum-likelihood expressions for source strengths and noise variance allow these parameters to be sampled implicitly rather than explicitly, reducing the dimensionality.
of the inversion. Examples are presented of both approaches applied to single- and multi-frequency localization of multiple sources in an uncertain shallow-water environment.

10:30
2aUWb2. Navigation on an unmanned surface vehicle using an acoustical spiral wave front beacon. Benjamin R. Dzikowicz (Naval Res. Lab., Physical Acoust. Branch Code 7136, 4555 Overlook Ave. SW, Washington, DC 20375, benjamin.dzikowicz@nrl.navy.mil), Brian T. Hefner (Univ. of Washington, Seattle, WA 98105), and Robert A. Leasko (Naval Surface Warfare Ctr., Panama City, FL 32407)

An acoustical spiral wave front beacon described previously [Dzikowicz and Hefner, J. Acoust. Soc. Am. 127, 1748 (2010)] is tested using an unmanned surface vehicle (USV). The beacon transmits from two transducers, one with a signal whose phase varies with aspect and a reference signal whose phase is constant with aspect. A remote hydrophone’s aspect relative to the beacon’s can be determined by comparing the phase of the two signals. The beacon is positioned at a fixed depth, location, and orientation in the acoustic test pond at the Naval Surface Warfare Center, Panama City Division. A remotely operated USV is equipped with a hydrophone at the same depth as the beacon. The vehicle is also equipped with a differential global positioning system (DGPS) receiver to determine its exact position. The USV is driven in several routes including areas of varying water depth and reverberation levels. The DGPS positions are then compared to the aspect as calculated by the spiral wave front beacon. Single, dual, and swept frequency beacon signals are tested and compared. [Work supported by the Office of Naval Research.]

10:45

An acoustical spiral wave front transducer radiates a field whose phase varies linearly with the azimuthal angle around the transducer. The phase of the field, therefore, carries information about the location of a receiver relative to the spiral transducer. There are two transducer designs that can produce this type of wave front. The first is a “physical spiral” transducer that creates the phase change by physically deforming the active element of the transducer. While this is the simplest design, the physical deformation produces a discontinuity in the active element that affects both the amplitude and phase of the outgoing field. The effect of this discontinuity is examined through both a finite element model and a baffled source approximation. The second technique uses an array of elements, each driven with an appropriate phase to produce a “phased spiral” transducer. The output of this transducer depends on the wavenumber of the outgoing field as well as on the radius and number of the elements of the transducer. Simulations and approximations show that by tuning these parameters, the variations in either the amplitude or the phase of the outgoing field can be minimized. [Work supported by the Office of Naval Research.]

11:00
2aUWb4. Eccentricity discrimination of hyperbolic localizations to minimize positioning uncertainty. J. Vallarta (JASCO Appl. Sci., 432-1496 Lower Water St., Halifax, NS B3J 1R9, Canada)

Conventional hyperbolic positioning locates a source at the intersection of hyperbolas that are based on time difference of arrival (TDOA) of a signal at each receiver pair. There are times when the intersections of these hyperbolas may delineate regions of uncertainty rather than a point, causing inaccurate locations due to the ambiguity of source positions. Localization accuracy can be improved by selecting hyperbolas based on their eccentricity (i.e., curvature). Since the eccentricity of a hyperbola is greater than 1, localization curves with eccentricity equal to or less than 1 are discarded as candidates of greater uncertainty. Curves with eccentricity tending to infinity are also discarded only when no other curves intersect with it. If the separations between receivers and the sound velocity are known, the hyperbolic eccentricity can be used as a function of the TDOA to minimize the ambiguity of source localizations. This method was applied to shallow water scenarios where 2-D localization approximation was sufficient (minimum three receivers required). Validation results of this method are shown based on bowhead (Balaena mysticetus) vocalizations recorded in the Chukchi Sea in autumn 2009.

11:15
2aUWb5. Automated estimation of marine engine rpm using underwater acoustic sensors. Gary R. Wilson and Martin L. Barlett (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, wilson@arlut.utexas.edu)

The effect of the underwater noise made by boats and ships on marine mammals is a current issue of research. The source of the noise is primarily the engine, and the intensity and spectrum of the noise are directly related to the rpm of the engine. For systems that monitor and record boat noise, it is useful to be able to estimate the rpm of the engine in order to correlate it with the noise characteristics. In this paper a processing method will be presented that automatically estimates engine rpm from the narrowband characteristics of the boat noise. This method was applied to recordings of a number of different boats in different environments. The accuracy of the results will be demonstrated by comparing the estimates with independent measurements of the engine rpm using instrumented test boats. General trends in broadband spectral intensity versus rpm will be shown for a wide variety of boats.

11:30
2aUWb6. Simultaneous passive and active detection and localization of humpback whales using data recorded on a towed horizontal receiving array. Zheng Gong, Hari Chauhan (Dept. of Elec. and Comput. Eng., Northeastern Univ., 360 Huntington Ave., Boston, MA 02115), Nicholas Makris (Massachusetts Inst. of Tech., Cambridge, MA 02139), and Purnima Ratilal (Northeastern Univ., Boston, MA 02115)

A large number of marine mammal vocalizations, most likely from humpback whales, were passively recorded by a towed horizontal receiver array during the ocean acoustic waveguide remote sensing (OAWRS) Gulf of Maine 2006 Experiment. Spectral analyzes show that most of the vocalizations were random broadband signals with short time duration, typically lasting 1–2 s in the 300–600-Hz frequency range. The array invariant method [Lee and Makris, J. Acoust. Soc. Am. (2006)] is applied to localize the whales in bearing and range after conventional plane-wave beamforming and matched filtering of passively recorded vocalizations on the array. The source levels of a sequence of whale calls in a 110-min time period, belonging to a number of individuals, are estimated based on the localization results. The estimation results are consistent with previous studies. We also examine the feasibility of actively detecting and localizing whales with OAWRS at low-to-mid frequencies in a random range-dependent ocean waveguide. A full-field waveguide scattering model is applied to determine the scattered field level from whales, as a function of range and depth location of the whale, and compared to environmental reverberation measured in the Gulf of Maine. The dominant source of scattering from whales at these frequencies is their air-filled lungs.

11:45
2aUWb7. Developing a matched-filter kernel for localization of complex sources in a dispersive ocean waveguide with array invariance. Hari Chauhan (Dept. of Elec. and Comput. Eng., Northeastern Univ., 264 Egan Res. Ctr., 360 Huntington Ave., Boston, MA 02115, chauhan.h@husky.neu.edu), Zheng Gong, Duong D. Tran (Northeastern Univ., Boston, MA 02115), Nicholas Makris (MIT, Cambridge, MA 02139), and Purnima Ratilal (Northeastern Univ., Boston, MA 02139)

The array invariant method was previously developed by Lee and Makris [J. Acoust. Soc. Am. (2006)] for instantaneously localizing both transient and continuous broadband acoustic sources after conventional plane-wave beamforming and matched filtering array measurements in an ocean waveguide. The approach has been demonstrated with experimental data for a source transmitting linear frequency modulated pulses and theoretically for multiple uncorrelated broadband noise sources. Spectral analyzes of field-recorded data acquired during the ocean acoustic waveguide
remote sensing Gulf of Maine 2006 Experiment show that acoustic signatures from many random sources, such as marine mammal vocalizations, are complex and exhibit irregular patterns that vary over time. Furthermore, the signals embedded in noise may also contain interfering sources correlated with the signal over some time duration. Here, we extend the array invariant method for range and bearing estimation of such complex sources. A robust matched filter kernel based on the acoustic properties of field-recorded whale vocalizations is developed to maximize the signal-to-noise ratio. Examples that simulate the beamformed and matched filtered array measurements of complex whale vocalizations embedded in noise in a random ocean waveguide are provided to demonstrate that range and bearing estimation using array invariance shows good agreement with true whale location.

TUESDAY AFTERNOON, 16 NOVEMBER 2010

GRAND CORAL 1B, 1:10 TO 4:45 P.M.

Session 2pAA

Architectural Acoustics: Acoustics of Precolumbine Buildings

David Lubman, Chair

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

Chair’s Introduction—1:10

Invited Papers

1:15

2pAA1. Archaeoacoustics: Suggestions for a methodological framework. Andrés Medina, Pablo Padilla, Lizette Alegre, Guadalupe Caro, Alejandro Ramos, Tonatiuh Ruiz, and Francisca Zalaquett (UNAM, cd. Universitaria, Mexico City 04510, Mexico, pabpad@gmail.com)

We attempt to present a unified approach to archaeoacoustics: its basic questions and methods, as well as the interaction of the several disciplines involved in this subject (archaeology, physics, engineering, mathematics, musicology, etc.) focusing on the role of the contribution of each of them. We illustrate with specific examples from pre-Colombian buildings in Mesoamerica what, from our point of view, might be called a working methodological framework. Our aim is to provide a general framework for archaeological studies in which anthropological aspects are taken into account, together with scientific and technical issues that involve the formulation of mathematical and physical models, performing experiments, and so on. Within a global formulation, we argue that discussing general problems such as intentionality or functionality of certain buildings can be better addressed.

1:45


Inspired by on-site observations and measurements, a computational acoustic model of the interior architecture of the 3000-year-old ceremonial center at Chavín de Huántar, Perú is presented. The model addresses the foundational study by Lumbreras et al. (1976) which posited an acoustic system integral to Chavín architecture involving “a network of resonance rooms connected by sound transmission tubes.” We propose a translation of the topology of Chavín gallery forms to a modular computational acoustic model based on bi-directional digital waveguides, representing the corridors and ducts, connected through reverberant scattering junctions, representing the small rooms. This approach combines known architectural dimensional and material data with representative measured acoustic data, thus economizing the collection of impulse response measurements required to accurately simulate site acoustics. Applications include virtual acoustic reconstruction of inaccessible or demolished site structures, and auralizations of hypothesized architectural forms, allowing any desired sound sample to be “played back” in the modeled acoustic context.

2:15

2pAA3. Acoustical solutions to archaeological mysteries at Chichen Itza’s temple of Kukulkan. David Lubman (DL Acoust., 14301 Middletown Ln., Westminster, CA 92683-4514, dlubman@dlacoustics.com)

Archaeoacoustics adds value when it solves problems that stump archaeologists. At the spring equinox, a mysterious zigzag shadow creeps down a staircase balustrade at the temple of Kukulkan. The shadow represents the Mesoamerican plumed serpent god Kukulkan—or Quetzalcoatl—descending from the heavens. Is that famous shadow accident or design? Despite compelling ethnographic ties linking the equinox display with ancient Maya culture, archaeologists have found no evidence of intentional design. Such evidence was found by relating a peculiar aural clue at the temple’s chirped echo-to Maya ethnohistory. The chirped echo emulates the call of the quetzal—the venerated bird messenger of the Maya gods, whose spectacular diving behavior at the spring equinox gives it the visage of a flying serpent. The echo is of cognitive interest as well. Echoes are normally taken in stride; not so the chirped echo. To many visitors it seems unnatural, even supernatural, for an echo to sound radically different than the stimulus, as if one’s handclap was answered by an invisible sentient being. If this echo can evoke numinous (spiritual) experiences in contemporary listeners without ties to the Maya, was it more powerful to the ancients for whom the sound was iconic?
2pAA4. Acoustical characterization of the archaeological sites “Plazuelas” and “Peralta” in Guanajuato, Mexico. Alejandro Ramos (Laboratorio de Cibernética, UNAM, Av. Universidad 3000, 04350, Mexico, alejora@gmail.com) and Andres Medina (Facultad de Ciencias, Av. Universidad 3000, 04350, Mexico)

This paper will present the work to be done on the sites “Plazuelas” and “Peralta”, both in Guanajuato, Mexico. The objective is to develop an archaeoacoustical characterization of the archaeological areas, as well as the musical instruments found in these sites. Furthermore, an ethnomusicologic and historiographic investigation will be done. In addition, it will present the methodology and protocol to be followed in this investigation.

2pAA5. Analysis of the acoustic response generated by the stairs of El Castillo of Chichen-Itza in relation to the sound source. A. Tonatihu Ruiz, Alejandro Ramos, Andrés A. Medina, and Pablo Padilla (Facultad de Ciencias, UNAM, Universidad 3000 Circuito Exterior S/N, DF, C.P. 04510, Mexico, ttonaru@gmail.com)

When a person claps in front the rebuild stairs of El Castillo in Chichen-Itza, a chirp echo similar to the song of a quetzal is produced. This acoustic effect has been studied by many authors and some of them have proposed physical models to explain it. The simulations generated by these models were compared only with a recording using a handclap as sound source. In this work we analyze the response of the acoustic effect to different sound sources, and we propose a model based on a linear array of N point sources to simulate this response. A series of Gaussian pulses with different spectral distributions was reproduced and recorded in situ. It was found that above a frequency related to the steps dimensions, the acoustic response is temporally prolonged, and depending on the spectral distribution of the pulse, the response may result in a frequency sweep with one or more harmonics. Comparing the recordings with the simulations, it was found that the general characteristics of the acoustic response were reproduced by the model. With the results of this work, it is possible to know how the sound source, the stair dimensions, and receptor position are related with the acoustic effect.

3:15—3:30 Break

3:30

2pAA6. Pyramids and basements. Sergio Beristain (ESIME, IMA, P.O. Box 12-1022, Narvarte, 03001 Mexico City, Mexico, sberista@hotmail.com)

Among pre-Columbian buildings there are a series of sound effects which resemble some particular sounds that can be related to religion or political affairs. Ancient pyramids were employed by priests and politicians as a high basement in order to make his voice heard in the distance, addressing their people, during the religious ceremonies and sometimes by the government or for social events or to promote a war or a defensive action; this was done when public address systems were not available, so it was a good medium to communicate with large audiences. Sometimes directly, and sometimes taking advantage of the stone walls behind the speaker to reinforce his speech level, and of course always without grassing incidence to avoid sound level reduction.

3:50

2pAA7. The Northern Group Plaza in Palenque: An archaeoaoustic study. Francisca Zalaquett (Centro de Estudios Mayas, Universidad Nacional Autónoma de Mexico, Circuito Mario de la Cueva s.n., Ciudad de la Investigación en Humanidades, Ciudad Universitaria, Delegación Coyocacán, DF, C.P. 04510, Mexico)

This investigation presents and proposes a methodology for archaeoacoustic analysis of public rituals of plazas in the Mayan area, which includes detailed studies of frequencies on musical instruments, white and pink sound emanations and their reverberations in the plazas, and laboratory studies on the stucco that covered the temples. As a result of the Northern Group study, we got rooms that could be used as scenarios where a musician or a speaker stood and this place allowed the frequency’s increase, causing an effect of “horn” in the sound and quality, so they were able to deliver messages at a greater distance, and therefore more people. On the plaza, we establish areas of musical activity and areas where they could locate the spectators with great acoustics and visibility of representations expressed. In addition we found that in this case, the design and construction of the structures by the Maya of Palenque involved a great knowledge of the sounds and behavior.

4:10

2pAA8. Ethnomusicology and musicology, a dialogue: Approaching interdiscipline research in Mexican archaeoacoustics. Lizette Alegre and Guadalupe Caro (Programa de Maestría y Doctorado en Música, UNAM, Av. Cuauhtémoc 130 depto. C-PH03 Col. Doctores, C.P. 06720, DF, Mexico, guadalupecaro@gmail.com)

Recently, ethnomusicology and musicology, scientific disciplines of music, share not only a study object, but also have begun a dialogue about methodology, approaches, and theoretical frameworks, which have benefited both musical disciplines, expanding their epistemological and analytic horizons. This dialogue empowered a better collaboration, in an interdisciplinary way, with other academic fields, such as archaeoacoustics. This work presents how the work of this musical discipline may have contributed to the archaeoacoustical work and investigations, by the objective review and analysis from different perspectives such as ethnographic and historiographic analysis and appreciation of sources, iconographic analysis, inter alia. It will approach archaeoacoustic methodology, from the ethnomusicology and musicology perspective.
In the Laboratory of Psychoacoustics of the Zacatenco IPN-ESIME-ICE Zacatenco studies of acoustic answer to an enclosure of the XVI century of Franciscan extraction were carried out. The opposing results revealed that the acoustics presented inside the enclosure is quite capricious since the form was truly deliberate for the projection of the word and the musical sounds of that time.
parameters. Additionally, avoidance reactions might result in displacement away from potential fishing grounds and lead to reduced catches. However, reaction thresholds and therefore the impacts of pile driving on the behavior of fish are completely unknown. Pile-driving noise was played back to cod and sole held in two large (40 m) net pens located in a quiet bay. Movements of the fish were analyzed using a novel acoustic tracking system. Received sound pressure level and particle motion were measured during the experiments. The results show significant movement responses to the pile-driving stimulus in both species at relatively low received sound pressure levels. This might indicate a rather large area of avoidance during real pile-driving operations. The results of the study have important implications on regulatory advice and the implementation of mitigation measures in the construction of offshore wind farms.

2:05

2pAB4. Wave energy and underwater noise: Assessment and monitoring aspects, Sofia Patricio (Wave Energy Ctr., Avenida Manuel da Maia, 36, r/c-Dto, 1000-201 Lisbon, Portugal, sofia@wave-energy-centre.org) and Cristiano Soares (MarSensing, 8005-139 Faro, Portugal)

After several decades of research, wave energy has reached a pre-commercial stage, emerging as a potential industry for the future. Despite the wave energy being considered as an environmentally friendly activity, it will probably have positive and negative aspects. The marine environment is an important area of biological diversity, hence the concern about the possible effects of underwater noise has increased among developers and promoters. While it is not expected that each individual device will produce a high level of noise, the deployment of several devices in the same farm, operating day and night, may have an effect on the fauna. However, the acoustic characterization of wave energy devices (WEDs) is not trivial; the noise generated and propagation will depend on oceanographic characteristics and operational conditions. The acoustic signature of each WED is expected to be produced from a variety of different components (mechanical or other moving parts) related to the device itself or by its interaction with the environment. Just after the characterization of the noise emitted by the WEDs and potential coincidence with the hearing sensitivity range of marine animals, a first assessment of possible impacts on the animals can be attempted.

2:25

2pAB5. New and developing seafloor petroleum extraction technologies: The noise of processing equipment operating in extreme environments. Michael Stocker (Ocean Conservation Res., P. O. Box 559, Lagunitas, CA 94938)

Fossil fuel demands are driving petroleum extraction operations into deeper marine settings and into oil reservoirs that are also deeper beneath the sea bed. To facilitate this new operating paradigm, operators are commonly placing processing equipment on the sea floor at 5000 feet or deeper where depth pressures can easily be 100–200 atm (1400–2800 psi). In areas such as the Gulf of Mexico, the reservoir pressures alone can reach 20,000 psi at the well head. The extracted product can be a mix of oil, gas, sand, and water, which all route through a chain of processing equipment such as separators, pumps, de-sanders, and routing manifolds under high pressures and high volumes. This paper will evaluate the new and developing fossil fuel extraction and production equipment to determine the scope and scale of the mechanical noises that these new technologies generate.

2:45

2pAB6. What it takes to meet the International Council for the Exploration of the Sea underwater noise limit for research vessels. Michael Bahtiarian (Noise Control Eng. Inc., 799 Middlesex Turnpike, Billerica, MA 01821, mikeb@noise-control.com)

Prior to 2010, the International Council for the Exploration of the Seas (or ICES) is the only organization to have developed and specified underwater (radiated) noise limit for commercial research vessels. European vessels were among the first to be built to this specification, which was given in a 1995 report. Since 2004, the US has built five ICES capable vessels, four of which were Fishery Research Vessels (FRV) for the National Oceanic and Atmospheric Administration (NOAA). In the US two more ICES capable vessels are in construction, another two are under consideration, and another three vessels are being built to an underwater noise requirement similar to ICES. This paper discusses the types of propulsion, machinery elements, and noise control features that are required in these vessels in order to meet the ICES underwater noise requirement. Design features and results will be presented from two specific, new vessel construction programs. The first is the NOAA FRV-40 program, which resulted in four fishery research vessels built at VT Halter Marine in Pascagoula, MS. The second program was for a smaller single research vessel for the University of Delaware.
There is a growing need for real-time monitoring of marine life and floating debris during a wide variety of commercial operations. These include seismic exploration for oil and gas, explosive removal of offshore structures, pile driving for the installation of marine structures including offshore wind farms, and the operation of tidal turbines and wave power generation devices. Active acoustics is likely the best method for monitoring where there is a high-danger region with a limited range around the activity. The SSI Swimmer Detection Sonar Network was originally designed as a human swimmer and diver detection and tracking system. However, extensive trials have demonstrated that the system is also capable of tracking marine life ranging from a large fish or marine mammals to schools of smaller fish. Analysis of the detection characteristics and movement behavior of marine life is being conducted as a means of tracking and classification at ranges out to roughly 500 m. The intent is to use it for monitoring during potentially harmful military and commercial activities such as high-powered Navy sonar, oil and gas exploration, explosive removal of offshore structures, seismic exploration for research and oil and gas, pile driving to install maritime structures including offshore wind farms, and marine hydrokinetic energy devices such as free turbines. The system will consist of multiple single-beam sonars, each transmitting and receiving a unique signal in a narrow beam. The hardware and software will initially be ported to a Xilinx FPGA, with future versions done within a custom made ASIC. Initial hardware and software research and development will be described, as well as the work done to create computationally efficient signal processing algorithms. This work will potentially lead to the first ever prototype of a sonar-on-a-chip implementation. [Work supported by the NAVAIR SBIR Program and ONR.]

Mohsen Badiey, Chair
College of Earth, Ocean, and Environment, Univ. of Delaware, Newark, DE 19716

Chair’s Introduction—1:00

Contributed Papers

1:05
2pAO1. Horizontally refracted acoustic signals on the hind side of a solitary internal wave in shallow water. Mohsen Badiey (Univ. of Delaware, College of Earth, Ocean and Environment, Newark, DE, 19716), Boris Katsnelson (Voronezh State Univ., Voronezh, Russia), Ying-Tsong Lin, and James Lynch (Woods Hole Oceanograph. Inst., Woods Hole, MA 02543)

Interference patterns of low-frequency, horizontally reflected sound signal from an approaching nonlinear internal wave (NIW) front toward an acoustic track was shown earlier. [J. Acoust. Soc. Am. 127, 1974 (2010).] When the first wave of NIW crosses the acoustic track, a new interference pattern emerges before the appearance of the acoustic shadow zone. Similar to the approaching behavior shown earlier, the interference pattern on the receding side of the NIW is related to the modal interference due to the interaction of low-frequency acoustic waves and the NIW. In this paper, the modal results of the receding behavior are compared with the approaching behavior and the modal order and exchange in each case is discussed. [Work supported by ONR.]

1:20
2pAO2. Intensity fluctuations dominated by horizontal refraction during a strong nonlinear internal wave event. Georges A. Dossot, James H. Miller, Gopu R. Potty (Dept. of Ocean Eng., Univ. of Rhode Island, Narragansett Bay Campus, Narragansett, RI 02882), Kevin B. Smith (Naval Postgrad. School, Monterey, CA 93943), James F. Lynch, Ying-Tsong Lin (Woods Hole Oceanograph. Inst., Woods Hole, MA 02543), and Mohsen Badiey (College of Marine and Earth Studies, Univ. of Delaware, Newark, DE 19716)

During the Shallow Water 2006 (SW06) experiment, a J-15 acoustic source deployed from the Research Vessel Sharp transmitted broadband (100–500 Hz) chirp signals 15 km away from a vertical line array. The array was intentionally positioned near the shelf-break front and in an area where internal waves are known to occur. During the same time an internal wave, “Event 44,” passed through the sound field such that the internal wave front was near parallel to the acoustic transmission path. Measured data show substantial intensity fluctuations that vary over time and space due to complex multimode and multipath (both 2- and 3-D) interference patterns. This...
presentation compares 3-D modeling results using the experimental geometry, acoustic signal parameters, and a simulated oceanographic environment based on environmental moorings and ship-born sensors to mimic the measured internal wave event. A modified version of the 3-D Monterey-Miami parabolic equation (MMPE) code which incorporates a user-defined sound speed field is used. Measured and modeled intensity fluctuations are compared during dominating horizontal regimes such as refraction, ducting, and “anti-ducting.” Modal-dependent time-arrival analysis during the different horizontal regimes is examined. [Work sponsored by the Office of Naval Research.] 

1:35

2pAO3. Horizontal interference structure of the sound field in the presence of moving internal waves and estimation of angle of horizontal refraction. Boris Katsnelson, Valery Grigorev (1, Universitetskaya sq, Voronezh Univ., Voronezh 394006, Russia), Mohsen Badiey (Univ. of Delaware, Newark, DE 19716), and James Lynch (Wood's Hole Oceanograph. Inst., Woods Hole, MA 02543)

The experimental results of Shallow Water 2006 experiment (SW06) and theoretical estimations of the key parameters are presented for observation of non-stationary interference pattern provided by “direct” and reflected acoustic signals (acoustic Lloyd mirror effect) in shallow water. Propagation of low-frequency LFM chirp signals with the center frequency of 300 Hz and the bandwidth of 30 Hz on an acoustic track of about 20 km was studied. Refraction in the horizontal plane is caused by the wave front of the nonlinear internal waves (NIWs) moving across the acoustic track. Analysis of the non-stationary interference pattern on the horizontal and vertical line arrays and theoretical estimations give the angle of horizontal refraction (angle between direct and reflected rays) and the velocity of NIW. Results of estimations are in a good agreement with experimental data. [Work supported by ONR.]

1:50

2pAO4. Space-time parabolic equation in horizontal plane for description of non-stationary horizontal refraction in shallow water. Boris Katsnelson (1, Universitetskaya sq, Voronezh Univ., Voronezh 394006, Russia)

Propagation of the sound signals in inhomogeneous anisotropic shallow water environment is studied. Because the waveguide dispersion shape of the radiated pulse is being changed, waveguide modes have different group velocities and, generally speaking, propagate along different ray paths in the horizontal plane. It can lead to simultaneous redistribution of the sound pulse intensity in the horizontal plane and compression/decompression in time. For a description of the spatial and temporal variation of the traveling pulse, space-time horizontal rays and space-time parabolic equation in the horizontal plane are derived using a standard approach used in electro-dynamics based on integral non-local connection between field and response. Non-stationary parabolic equation (PE) is an extension of the well known approach of vertical modes and PE in the horizontal plane. This equation is analyzed and its solutions are constructed for different models of perturbation: nonlinear internal waves, coastal slope, etc.

2:05


Internal waves, bottom sand waves, and ocean surface waves are known to have an impact on low-frequency 3-D acoustic propagation in shallow water. It has been shown that at certain horizontal grazing angles, acoustic normal modes can be ducted in between such waves and thus propagate along these waves with less spreading loss. In coastal areas, internal waves are often seen as packets with curved wave fronts. Different internal wave packets sometimes superimpose, creating a crossing structure. This work is focused on 3-D acoustic propagation through crossing waves that are the result of either internal wave packets crossing or combinations of internal waves, bottom sand waves, and surface waves crossing at various angles. An idealized shallow water column model is used for understanding the mode coupling physics and acoustic horizontal refraction. 3-D parabolic equation numerical modeling of acoustic propagation through a more realistic environment with crossing waves is then performed for quantitative analysis of the mode coupling effects.

2:20

2pAO6. Modeling of broadband acoustic energy evolution in dynamic ocean environments using high-performance computing resources. Natalia Sidorovskaia (Dept. of Phys., Univ. of Louisiana at Lafayette, UL Box 44210, Lafayette, LA 70504-4210, nas@louisiana.edu)

In recent years there has been considerable interest in predicting the acoustic energy temporal evolution over large 3-D volumes. Most of the standard Navy models are inherently 2-D and produce the acoustic pressure distribution of a point harmonic source in the vertical source-receiver plane. There are several computational challenges that have to be addressed when using these models for quantitative modeling of the acoustic pressure distribution from natural or man-made sources: broadband and complex spatial structure of a source, dynamic propagation environment, a need for 3-D dynamic acoustic energy reconstruction, and an efficient visualization of a large amount of generated data. An attempt to resolve any of the mentioned issues makes the computational time on a personal state-of-the-art computer up to several days (or weeks) and strongly dictates a need for upgrading these models to the high-performance computing (HPC) environment. The range-dependent acoustic model (RAM) was transferred into the HPC environment and upgraded to model broadband acoustic energy distribution from spatially distributed sources. The MPI domain decomposition approach to run the model on several cluster nodes is discussed. The data demonstrating computational time efficacy for multi-node multi-processor runs versus standard serial implementation are presented. [Research supported by TeraGrid Pathways Fellowship.]

2:35


Acoustic modes propagating in shallow oceans interact with internal-wave-induced oscillations in the thermocline to produce fluctuations in the acoustic field that govern spatial and temporal coherence. Such interactions are described by both adiabatic-mode path-integral theory and coupled-mode scattering-matrix theory. In both theories there occurs an integral over the water depth that sums the interaction between the buoyancy, acoustic modes, and internal-wave modes. This paper explores the strength of that interaction for the Transverse Acoustic Variability Experiment, which took place in the northern limit of the East China Sea in 65–80 m of water. A towed CTD chain provided measurements of the density and sound-speed profiles required to compute the necessary acoustic and internal-wave modes. Within this environment, two main conclusions pertain. First, with respect to internal-wave mode contributions, the first mode contributes most significantly with a lesser contribution from the second mode and negligible contributions from higher modes. Second, with respect to acoustic mode contributions, the adiabatic mode contribution is most significant with lesser contributions from adjacent modes, and negligible contributions from widely separated mode pairs. [Work supported by the Office of Naval Research.]
different portions of the thermocline. The single bottom-interaction does not show multiple paths, possibly due to a steeper angle of interaction with the thermocline. Multiple thermistor strings show the spatial and temporal changes in thermocline for this geometry. The observed thermocline variations are used to approximate the conditions under which the multiple direct paths were observed. By using ray-based and parabolic equation models, the interface conditions necessary for generating the multiple direct paths are investigated. Results show that modulations in the spatial and temporal variability of the thermocline are a potential forcing mechanism for the observed interference patterns. [Work supported by ONR.]

3:05

Nonlinear internal waves have 3-D random variability arising from smaller scale physical oceanographic processes. Internal wave fronts form acoustic waveguides, and we examine propagation in directions nearly parallel to the fronts. Intensity fluctuations arising from the variability are treated by 3-D radiative transport methods, which provide insight into the parametric dependence on waveguide randomness. We focus on situations where the frozen field approximation applies, the acoustic modes represent the vertical energy distribution in the waveguide, and the modes propagate adiabatically. Under these conditions, horizontal modal variations are specified by 2-D radiative transport equations. With additional assumptions, the transport equations simplify to diffusion-type equations. An alternative energy scattering interpretation describes the duct in terms of a 2-D Galton’s box, assuming discrete scattering events occur as modes propagate down the duct. The resulting equations also reduce to diffusion-type equations in the continuous limit. We use appropriate spatial variability scales from the SW06 experiment to investigate properties of acoustic intensity. [Work supported by the ONR.]

TUESDAY AFTERNOON, 16 NOVEMBER 2010
CORAL GALLERY 1B/2B, 1:00 TO 3:05 P.M.

Session 2pBBa

Biomedical Ultrasound/Bioresponse to Vibration, Physical Acoustics, and Signal Processing in Acoustics: Biomedical Applications of Time Reversal Acoustics

Armen Sarvazyan, Cochair
ARTANN Laboratories, 1459 Lower Ferry Rd., Trenton, NJ 08618

Mathias Fink, Cochair
ESPCI, 10 rue Vauquelin, 75005 Paris, France

Chair’s Introduction—1:00

Invited Papers

1:05
2pBBa1. An overview of time-reversal acoustics in biomedical engineering, Mathias Fink (Institut Langevin, ESPCI, 10 rue Vauquelin, Paris, France)

Time-reversal is a very powerful method for focusing ultrasound through complex and heterogeneous media and shows very promising results in biomedical applications. Here, we review the main applications investigated in medicine in this last decade. From the iterative time-reversal processing to the diagonalization of the time-reversal operator, we will discuss the various approaches for correcting distortions induced by different aberrators (skull, ribs, and abdominal layers). A complete discussion about the various targets available in tissues will show that in many situations time-reversal focusing can be used very efficiently. Several examples in ultrasound therapy and in ultrasound imaging will be presented.

1:25
2pBBa2. Time-reversal techniques in ultrasound-assisted convection-enhanced drug delivery to the brain: Technology development and in vivo evaluation, George K. Lewis, Jr. (Dept. of Biomedical Eng., Cornell Univ., 108 Olin Hall, Ithaca, NY 14853, george@cornellbme.com), Laurent Fillinger (Artann Labs., Lambertville, NJ 08530), George K. Lewis, Sr. (Transducer Eng. Inc., Andover, MA 01810), William L. Olbricht (Cornell Univ., Ithaca, NY 14853), and Armen Sarvazyan (Artann Labs., Lambertville, NJ 08530)

We describe a drug delivery method that combines time-reversal acoustics (TRA) with convection-enhanced delivery (CED) to improve the delivery of therapeutics to the interstitium of the brain. The ultrasound-assisted CED approach (UCED) circumvents the blood-brain barrier by infusing compounds through a cannula that is inserted into the brain while simultaneously delivering ultrasound to improve the penetration of pharmaceuticals. CED without ultrasound-assistance has been used to treat a variety of neural disorders, including glioblastoma multiforme, a malignancy that presents a very poor prognosis for patients. The mixed clinical success of CED suggests that the technique needs to be improved. We describe a novel TRA system that is used to infuse fluids into the brain parenchyma while simultaneously exposing the tissue to safe levels of 1-MHz, low-intensity, ultrasound energy. The system includes a com-
bined infusion needle-hydropone, a 10-channel ultralow-output impedance amplifier, a broad-band ultrasound resonator, and MATLAB-based, TRA control and user-interface. TRA allows easy coupling of ultrasound therapy through the skull without complex phase-correction and array design. The smart targeting UCED system has been tested in vivo and results show that it provides 1-mm spatial resolution for UCED and improves tracer distribution in the brain six to eight times over CED alone.

1:45

2pBBa3. Focusing with time reversal of speckle noise. Jean-Luc Robert (Philips Res. North America, 345 Scarborough Rd., NY 10530, jean-luc.robert@philips.com) and Mathias Fink (Universite Denis Diderot, 75005 Paris, France)

The FDORT method (French acronym for decomposition of the time reversal operator using focused beams) is a time reversal based method that can detect point-scatterers in a heterogeneous medium and extract their Green’s function. It is particularly useful when focusing in a heterogeneous medium. In this presentation, the theory of the FDORT method is generalized to random media (speckle), and it is shown that it is possible to extract Green’s functions from speckle signal using this method. Therefore it is possible to achieve a good focusing even if no point scatterers are present. Moreover, a link is made between FDORT and the Van Cittert Zernike theorem. We deduce from this interpretation that the normalized first eigenvalue of the focused time reversal operator is a well-known focusing criterion. The concept of an equivalent virtual object is introduced, which allows the random problem to be replaced by an equivalent deterministic problem and leads to an intuitive understanding of FDORT in speckle. Applications to aberration correction are presented. The reduction of the variance of the Green’s function estimate is discussed.

Contributed Papers

2:05

2pBBa4. Three dimensional focusing through the ribs using the decomposition of the time reversal operator. Etienne Cochaud, Claire Prada, Jean-François Aubry, and Mathias Fink (Laboratoire Ondes et Acoustique, Institut Langevin, ESPCI, CNRS UMR 7587, 10 rue Vauquelin, 75005 Paris, France)

Thermal ablation induced by HIFU is a promising non invasive technique to treat liver tumors. However, skin burns have been reported due to the high absorption of ultrasonic energy by the ribs. We propose to use the decomposition of the time-reversal operator to build an acoustic focusing field that spares the ribs. This method requires a fully programmable multi-channel electronic device and a 2-D array of 121 elements working at 370 kHz. In the first step, the array response matrix is measured. The dominant singular vectors of this matrix are associated with the ribs (strongest scatterers). The weight vector is then calculated as the projection of a focusing vector onto the subspace orthogonal to these eigenvectors. This procedure can be repeated for any focusing law. It is, therefore, possible to steer around the whole tumor while avoiding the ribs. This procedure is better than successive angulations of a time-reversed signal measured at the center of the tumor which may send energy on the edge of the ribs. The rib sparing and the effect of angulation have been investigated and compared to time reversal focusing.

2:20

2pBBa5. Waveform interpolation for applications of time-reversal acoustics. Raghu Raghavan (Therataxis, Johns Hopkins, Eastern Bldg., Ste. B305, 1101 East 33rd St., Baltimore, MD 21218, raghu@therataxis.com) and James G. Bryman (Univ. of California, Berkeley, CA 94740)

Acoustic waves can force fluids and small objects along the direction of sound propagation (streaming). Potential applications include forcing the flow of fluid and small particles for drug delivery into the brain, or of underground fluids, such as oil, toward a borehole collection point. Well-focused beams are needed for amplitudes to be large enough to be effective. Time-reversal acoustics (TRA) permits focusing of sound in heterogeneous media, but requires a sensor near a point of desired focus. There is a need, addressed in this talk, to send signals off the grid of available sensors so that forces may be delivered in regions between receivers, and no receiver need be at a point where focus is desired. The Acoustic Shepherd introduces interpolation or data matching of TRA-derived Green functions for this purpose. One particular scheme of geometric-mean interpolation is analyzed, and the curve along which the focus moves from one receiver to another as the interpolating weights are varied is shown. General approxima-
tion methods from statistical estimation theory are also described. These methods are expected to work better in heterogeneous media (having un-
known acoustic properties) than well-known beam forming methods designed for homogeneous media.

2:35

2pBBa6. Time reversal acoustic approach for non-lethal swimmer deterrent. Alexander Sutin (Stevens Inst. of Technol., Hoboken, NJ 07030, Alexander.Sutin@stevens.edu) and Yegor Sinelnikov (Misonix Inc., Farmingdale, NY 11735)

Protection against surface and underwater threats from swimmers constitutes one of the most challenging aspects of port security. The envisioned risk mitigation consists of passive acoustic detection and localization and active diver deterrent, which can be done by focusing high intensity acoustic waves. The main goal of this study is to explore the possibility of using the time reversal acoustics (TRA) sound focusing system for non-lethal swimmer neutralization. This breakthrough technology enables the precision targeting of a hostile diver with minimum impact to the marine environments. The acoustic noise radiated by the diver is used to focus the acoustic energy, while the moving diver acts as an active self-disclosing acoustic beacon. In a shallow water environment the radiated noise, direct and multi-path interference, is rich and is exploited through the use of TRA. The effectiveness of the TRA focusing and the diver deterrence zone are modeled taking into consideration the frequency range, harbour depth, and geometry of the transducer array. The real diver noise and hydroacoustic signature parameters collected by Stevens Institute of Technology are used in the theoretical analysis. It is demonstrated that outside the focal region, the TRA system produces acoustic intensities that are not harmful for marine life.

2:50


Dual-mode ultrasound array (DMUA) systems have been recently shown to be capable of refocusing high-intensity focused ultrasound beams in the presence of strongly scattering objects, e.g., the ribs. A refocusing approach based on a modified forward/inverse propagation operator accounting for the ribs was experimentally demonstrated. The propagation operator is obtained from DMUA-based images of the target region, including the ribs. This image-based approach to refocusing implicitly accounts for the heterogeneity by defining a propagation operator to the proximal rib surface using DMUA directivity vectors assuming a homogeneous medium. In this paper, we address the consequences of this simplification on the quality of the refocused beam. We also present an adaptive refocusing approach that identifies the inhomogeneous directivity vectors to the target location(s) in the presence of the ribs. Example experimental results will be presented using a true real time DMUA system where the adaptive refocusing is achieved within milliseconds. Implications on real time adaptive refocusing in the context of scanning multiple target points and motion compensation will also be discussed with illustrative experimental data.
TUESDAY AFTERNOON, 16 NOVEMBER 2010

Coral Gallery 1B/2B, 3:30 TO 5:00 P.M.

Session 2pBBb

Biomedical Ultrasound/Bioresponse to Vibration: Biomedical Applications of Radiation Force

George K. Lewis, Chair
Dept. of Biomedical Engineering, Cornell Univ., Ithaca, NY 14853

Contributed Papers

3:30
2pBBb1. Three types of nonlinearity in physics of radiation force and acoustical streaming. Oleg Rudenko (Moscow State Univ., Moscow, Russian Federation, and Blekinge Inst. of Technol., Karlskrona, Sweden) and Armen Sarvazyan (Artann Labs., Lamberti, NJ 08530)

Thousands of papers on radiation force (RF) and acoustical streaming (AS) have been published since Faraday, Rayleigh, and Langevin. There have been more papers on RF in biomedicine published in the last decade than were written in the entire 20th century. Nevertheless, some nonlinear acoustic phenomena significant for engineering and biomedical applications are still misunderstood. The presentation contains discussions regarding frequently occurring inaccuracies and contradictions. In particular, it features three types of nonlinearities which can appear independently. First, equations for RF, which are nonlinear by themselves, can contain the nonlinear modulus of a medium. Second, the nonlinear distortion of the acoustic field, depending on sound intensity and geometric and physical nonlinearities, can govern the RF and AS. Third, hydrodynamic nonlinearity often appears at the development stage of streaming. The latter nonlinearity may be observed in weak acoustic fields if RF acts for a long period of time on a low-viscosity fluid. Evaluations and formulas are given, indicating conditions for separate observation of each type of nonlinearity. Some contradictions existing in the literature are resolved. Experimental data illustrating developed theory are shown.

3:45
2pBBb2. On the acoustic radiation force in soft tissues induced by a focused ultrasound beam with transient modulation envelope. Egor Donsov and Bojan Guzina (Dept. of Civil Eng., Univ. of Minnesota, 500 Pillsbury Dr. SE, Minneapolis, MN 55455, guzina@wave.ce.umn.edu)

Building on a recently developed plane-wave model for Newtonian fluids, this work studies the acoustic radiation force (ARF) in soft tissues generated by an ultrasound beam, modulated by a transient modulation envelope whose dominant frequency (relative to that of the ultrasound carrier) is on the order of the Mach number. Following earlier treatments of the featured “narrow-beam” problem, the analysis focuses on the spatio-temporal distribution of the ARF in the focal region and approximates the tissue’s responsive to the usual time-averaging procedure owing to the fact that the mean acoustic fields remain plagued by the rapid oscillation features. The asymptotic treatment is pursued within the framework of plane waves via scaling approach that splits the temporal variable into “fast” and “slow” time, permitting one to track the contribution of (time-harmonic) sound and its modulation separately in the solution. In this setting the second-order solution, written in terms of “fast”-time averages, is shown to (i) be free of rapid oscillations and (ii) permit compact formulation in terms of “modulation-corrected” ARF that accurately captures the effect of sound modulation. The developments are illustrated by the analytical solution for a sinusoidal modulation envelope with quiescent past, which both exposes the limitations of earlier treatments and highlights the possibility of ARF generation in a lossless fluid when a modulated, high-intensity sound field is propagated through it.

4:15
2pBBb3. Effect of low-frequency sound modulation on the acoustic radiation force in Newtonian fluids. Bojan Guzina and Egor Donsov (Dept. of Civil Eng., Univ. of Minnesota, 500 Pillsbury Dr. SE, Minneapolis, MN 55455, dons002@umn.edu)

This study investigates generation of the acoustic radiation force (ARF) in Newtonian fluids when the amplitude of high-intensity sound field is modulated using frequencies that are, relative to the frequency of sound, on the order of the Mach number. This configuration, which is frequently deployed by emerging biomedical sensing technologies, turns out to be unresponsive to the usual time-averaging procedure owing to the fact that the mean acoustic fields remain plagued by the rapid oscillation features. The asymptotic treatment is pursued within the framework of plane waves via scaling approach that splits the temporal variable into “fast” and “slow” time, permitting one to track the contribution of (time-harmonic) sound and its modulation separately in the solution. In this setting the second-order solution, written in terms of “fast”-time averages, is shown to (i) be free of rapid oscillations and (ii) permit compact formulation in terms of “modulation-corrected” ARF that accurately captures the effect of sound modulation. The developments are illustrated by the analytical solution for a sinusoidal modulation envelope with quiescent past, which both exposes the limitations of earlier treatments and highlights the possibility of ARF generation in a lossless fluid when a modulated, high-intensity sound field is propagated through it.

4:45
2pBBb4. Generalized response of a sphere embedded in a viscoelastic medium excited by ultrasound radiation force. Matthew W. Urban, Ivan Nenadic, Shigao Chen, and James F. Greenleaf (Dept. of Physio. and Biomed. Eng., Mayo Clinic College of Medicine, 200 First St. SW, Rochester, MN 55905, urban.matthew@mayo.edu)

Characterization of viscoelastic soft tissues and materials has been a subject of active research for the past two decades. Techniques involving measuring the motion of embedded spheres in viscoelastic media have been employed to measure the viscoelastic material properties of the media surrounding the sphere. The published work using this technique has been restricted to using the Kelvin–Voigt model, which has worked for gelatin-based materials. We report an extension of this method using different viscoelastic rheological models such as the Maxwell, generalized Maxwell, Zener, and Kelvin–Voigt fractional derivative models. We performed experiments using a focused 3-MHz transducer to produce radiation force and a 7.5-MHz pulse-echo transducer to measure the sphere motion. Two phantoms were tested; one made of gelatin and another made of rubber with stainless steel spheres embedded in the materials. The response was fitted with the generalized theoretical model to find the material properties over bandwidths of 25–3000 Hz. The median absolute errors ranged from 0.0036–0.0421 and 0.0365–0.0879 for the gelatin and rubber phantoms.
This paper presents the design process for a stereophonic loudspeaker system. The enclosures provide a base that is inert to vibration and an effective medium for the uniform formation of acoustic waves. Each enclosure is divided into two parts: the base for the high-frequency transducer (tweeter) and the enclosure for the low-frequency one (woofer). The former consists on a solid marble sphere with a copper heat radiator for the tweeter. This one lies on a rail that allows it to move forward and backward to be synchronized in time and phase. The second enclosure incorporates a rigid and massive quarter wavelength type MDF structure that controls the impedance of the woofer by means of a tube and reduces the amplitude of the sound from the back of the woofer diaphragm while favoring the omnidirectional characteristic and controlling diffraction with the baffle. Simulations in Matlab from King’s tables for the woofer enclosure and others in Matlab for the filter are presented. This paper shows, as a result, a loudspeaker system of high specifications designed with the use of simple low-cost materials while considering the restrictions of designs of the same type in the industry.
This work presents a modal analysis of acoustic enclosures by dual reciprocity—boundary element method. The dual reciprocity formulation allows the use of known Laplace fundamental solutions. The boundary element method possesses the main characteristic that reduces the dimension of the problem upon the discretization of the boundary to calculate the acoustic eigenfrequencies. The DR-BEM formulation has been evaluated with the use of interior nodes and some known radial basis functions. The results obtained by this technique have been compared with the more extensively diffused competitor, the finite element method (FEM). Results show good agreement between both methods.

A method for measuring the sensitivity of loudspeakers in diffuse field conditions is proposed. This method uses sweeps as excitation signals capturing information related not only to the loudspeaker but also related to the room (reverberant chamber). To approximate this measurement to a free field condition, digital signal processing techniques are used to remove the components that are related to the room. The results are compared with measurements done following the method described in the international standard IEC 60268-5 for a free field condition.

This paper compares the properties of a rectangular electrostatic speaker with a circular one. To achieve this, different prototypes were built with the same radiation area (approximately 230 cm²). The radiation diagrams were drawn and distortion measurements were taken so as to establish a correlation between the THD and the different coating resistances of the diaphragm. From the results obtained, the intention is to optimize the design of a large electrostatic transducer.

2:30


Magnetostrictive drives require features that are not standard in piezoelectric motors: a magnetic return path, a biasing mechanism, and eddy current mitigation. In spite of this, there is not a standard design for a self-contained magnetostrictive drive. This work investigates a configurable high-power magnetostrictive design drive built from Galfenol that can be tailored to fit a variety of applications. Based on this design, a prototype drive has been constructed from laminated Galfenol structures and neodymium biasing magnets. In this paper, the author will discuss the fabrication and characterization of the device and compare the results to simulated performance.

2:45

2pEA7. Multidomain ferroelectric actuator. Victor Klymko (Dept. of Phys. and Astronomy, Univ. of Mississippi, University, MS 38677, vick@olemiss.edu), Andriy Nadtochiy (Natol. Taras Shevchenko Univ., Kiev, Ukraine), David Sedorook, and Igor Ostrovskii (Univ. of Mississippi, University, MS 38677)

An improved performance of a multidomain PZT actuator compared to a single domain actuator is demonstrated. The multidomain actuator is fabricated out of a 650-μm-thick single domain plate. The internal electric field is inverted by applying 1 kV/mm external electric field to 1/3 of the plate area. The remaining 2/3 of the area is left with the original internal electric field. The actuator is excited by applying 1 kV ac voltage to the electrodes deposited on the top and bottom surfaces of the plate. The amplitude of acoustical displacement at the tip of the actuator is calculated with the aid of a finite element model developed for the piezoelectric plate. In an experiment, the amplitude of tip vibration is measured under the microscope. The experimental resonance frequencies are consistent with the calculations. At the second order resonance, the calculations predict 25% higher amplitude of vibrations for the multidomain actuator than for the single domain actuator. The results for several actuators with different configurations of ferroelectric domains are compared to the single domain actuator.

Resolution of circuits from a physical point of view. Juan C. Gimenez de Paz (Nobel 678, 1718 San Antonio de Padua, Buenos Aires, Argentina, gimenezdepa@gmail.com)

The resolution of electrical circuits and its acoustic homologous from a physical point of view is presented here. Based on the structural analogy with a mechanical system in equilibrium, it is possible to associate a Lagrangian depending on the generalized forces in a viscous medium with friction coefficients and with a set of holonomic constraint functions. By means of the Hamilton principle applied to this Lagrangian associated with a circuit, an equation of a function of a set of independent variables can be obtained, which turns out to be the condition for the resolution of a typical problem of linear programming. This minimized expression represents, in turn, the power dissipation. From all the possible values that could take that set of speed and pressure, the principle picks out one and only one (the true values) with the condition of producing the least power dissipation. All
acoustic devices or systems that could be represented by an electric circuit may be solved by this method with the physical meaning described here.

3:45
2pEA10. Three-dimensional polar analysis of a directional endfire microphone array on a measurement manikin due to microphone drift.
Thomas H. Burns (Starkey Labs, Inc., 6600 Washington Ave. S., Eden Prairie, MN 55344)

A dual microphone endfire array is a directional system commonly used in hearing aids. The directional performance of such systems is sensitive to sensor mismatch and drift. In this study, a pair of matched omnidirectional microphones in a delay-and-sum configuration are mounted in a hearing aid and their directional response is measured in-situ on KEMAR at 10 resolution in azimuth and elevation. The resulting 3-D polar balloons, directivity indices, and unidirectional indices are computed as a function of frequency. The measured transfer functions are then perturbed with sensor mismatch responses acquired empirically from typical lots of hearing-aid microphones. The resulting polar benchmarks are evaluated and compared to the original benchmarks. A detailed analysis, both visual and numerical, will be presented.

4:00
Francois M. Guillot (George W. Woodruff School of Mech. Eng., Georgia Inst. of Technol., 771 Ferst Dr., Atlanta, GA 30332-0405, francois.guillot@me.gatech.edu), Haskell W. Beckham, and Johannes Leisen (Georgia Inst. of Technol., Atlanta, GA 30332-0295)

Most energy harvesting devices rely on traditional piezoelectric materials integrated in structures subjected to mechanical vibrations. A new kind of commercially available composite sensors consists of several PZT fibers embedded in a passive polymer matrix with interdigitated electrodes deposited on the polymer. The main advantage of these sensors is that they combine the large transduction capabilities of PZT with the mechanical flexibility of polymers. The present study evaluates the performance of single PZT fibers, with electrodes directly deposited onto the ceramic and with individual polymer jackets. This approach should optimize the amount of electrical energy available from the material by placing the electrodes on its surface and by maximizing the active piezoelectric volume. Fibers with tubular and “crimped” cross-sections were electroded, poled, and glued to a vibrating cantilever beam equipped with a strain gauge. The performance of the fibers was compared to that of the composite sensors, and the influence of the polymer jacket on the strain transmission to the fiber was also evaluated. A potential application is to incorporate these fibers into fabrics that would be capable of harvesting mechanical energy from the environment (wind, currents) or from human motion. [Work supported by the National Textile Center.]

4:15
Dawnielle Farrar (Appl. Phys. Lab., Johns Hopkins Univ., 11100 Johns Hopkins Rd., Laurel, MD 20723), James E. West, Michael S. Yu (Johns Hopkins Univ., Baltimore, MD 21218), and Wonkyu Moon (Pohang Univ. of Sci. & Technol., Pohang 790-784, Korea)

Composite films and microfibers based on the biopolymer poly(ã-benzyl á,L-glutamate) (PBLG) and their fabrication and piezoelectric properties are presented. By simultaneous poling and curing of PBLG/methylmethacrylate, we fabricated a flexible composite film with approximately 20% of the PBLG molecules oriented normal to the film surface exhibiting (d33 = 20 pC/N), and its Young’s modulus was 1 GPa. In an effort to optimize net dipole orientation in PBLG, we produced piezoelectric microfibers (diameter: 100 nm) with nearly all of the PBLG dipoles oriented along the fiber axis, evidenced by x-ray diffraction. The PBLG fibers showed high piezoelectricity (d33 = 32 pC/N) and an elastic modulus of 570 MPa. Both the piezoelectric film and fiber systems can be fabricated directly from solution in a mold or on a substrate. Due to the versatility in the fabrication process and the high piezoelectricity, these materials show great promise as transducer materials.
Fourier theorem and the acoustical uncertainty principle—are comprehended without difficulty. They are applied in the perceptual analysis of some simple signals and, almost immediately, to the analysis of some musical passages specially chosen. As a result, the student sees that the acoustical science is related to the musical practice, stimulating their interest in the discipline.

1:20

2pED2. Musical acoustics as a general education course. Andrew C. Morrison (Dept. of Phys., DePaul Univ., 2219 N. Kenmore, Chicago, IL 60614-3504)

Musical acoustics is a highly interdisciplinary field of study, drawing elements from music, physics, psychology, engineering, architecture, computer science, and electronics. As such, a course in musical acoustics for non-science majors is an ideal option for a general education science course with laboratory at many colleges and universities. Students with a predisposition against science courses may be more willing to engage in an acoustics class if they perceive that the course will overlap with their interest in music. An overview of a typical course in musical acoustics will be presented. A summary of topics, classroom demonstrations, laboratory activities, and active learning techniques will be presented.

1:35

2pED3. Civil engineering program in audio and acoustics at Universidad Tecnologica de Chile INACAP. Roberto Muñoz and Cristian Jimenez (Audio and acoustics area, Perez Rosales campus, Univ. Tecnologica de Chile INACAP, Brown norte 290, Ñuñoa, Santiago, Chile, rmunoz@inacap.cl)

The Audio and Acoustics Engineering program at Universidad Tecnologica de Chile INACAP is quite broad and it has a 12 semester duration. It includes physical acoustics, engineering acoustics, underwater acoustics, electroacoustics, noise and vibration control, environmental acoustics, architectural acoustics, sound recording, ultrasound, environmental noise control, digital signal processing, and the design of reinforcement systems. The facilities of the laboratories include an anechoic chamber, electro-acoustic laboratory, and recording studios. Our students gain experience in field measurement techniques, computer calculation, and modeling. Our graduated students are able to work in areas such as recording, production, sound reinforcement, programming, environmental acoustics, and acoustic conditioning for recording studios, media, film production companies, television, and video. Also, they are able to design noise and vibration control systems. The paper will present a comprehensive overview of the program of audio and acoustics at Universidad Tecnologica de Chile INACAP. In part, the paper will contain the following: (1) description of the acoustics and related courses offered; (2) examples of the facilities available at the campus; and (3) listing of graduate thesis, completed and in progress.

1:50

2pED4. Programming abilities description for engineering students. Federico D. Felipe, Maria Elena M. Acevedo, and Ignacio S. Martinez (ESIME, IPN, Av. IPN s/n, Col. Lindavista, C.P. 07738, Ciudad de Mexico, Mexico, ffelipe@ipn.mx)

This paper focuses in specifying competences for high-level language programmers. It is very important to train engineering students in programming, because this forms the basis for solving many problems. It is not necessary to write programs in a high-level language for the solution for many problems; however PDS programs are used for acoustics, telecommunications, computer science, and many other fields. There are specified seven competences for programmers and there is a proposal for basic exercises to obtain these competences.

2:05

2pED5. Latin American universities’ future. Pablo Lizana, Dr (Acoust. Lab., ESIME, IPN, Antonia 15-15, San Jeronimo Lidice, DF, Mexico, plizana@ipn.mx)

Latin American universities are at a disadvantage as compared to those of the rich countries. The number of graduates in relation to population, number of research programs in science and technology, budget, etc., reveal their flaws. This paper examines some proposals from the flexible training point of view and also overlooks the organization that usually is based on administration over the academic matters. With few exceptions, the Latin American universities are still pre-modern: the pre-eminence does not generate academic merit, and concern for the infrastructure problems occur instead of the teaching ones, making the results not competitive on the international level. The investigation is insufficient or even nonexistent, and full time researchers are very few and often give opinions that agree with the administration. Some changes are suggested.

2:20

2pED6. First empirical works in acoustics done by freshman students at a public university in Argentina. Pablo Kogan and Mayra Cardozo (Universidad Nacional de Tres de Febrero, V. Gomez 4752, Caseros, B1678ABH, Prov. de Bs. As., Argentina)

A Bachelor of Science in Sound Engineering of five years was created at the Universidad Nacional de Tres de Febrero in Argentina in the year 2007. The first course of the specialty that students should take is Introduction to Acoustics and Psychoacoustics. It has a four month duration, is located in freshman year, and does not require any previous course of math or physics. The course intention is to give beginners a global point of view of acoustics and its branches, the nature of sound, some elementary concepts of acoustics, and the basic methodological directions to elaborate academic projects on acoustics. As a very early stage training in scientific method and project development, the undergraduates have to choose a topic to perform a project during the semester. At the middle of the course, the pupils ought to consign a project progress report. At the end of the course, they submit the final report and give an oral presentation to their classmates. This work expounds the empirical work in class and summarizes some student projects carried out during the courses taught between years 2007 and 2010.
2:35

2pED7. A successful undergraduate program in acoustical engineering. E. Suarez (Universidad Austral de Chile, Instituto de Acustica, General Lagos 2086, Valdivia, Chile, enriquesuarez@uach.cl)

Since 1967 undergraduate programs on acoustics have been offered at the Univ. Austral de Chile. At the beginning a 4-year program on Sound Engineering was offered, which was developed and expanded in 1981 to a 5-year program on Acoustical Engineering. In 2001, the curriculum was again extended to a 6-year program on Civil Engineering in Acoustics. So far, more than 300 graduates have completed training in the programs: 65 sound engineers, 230 acoustical engineers, and 10 civil engineers in acoustics. Although these programs have been unique in Latin America for many years, more recently some other programs are offered in Chile and other countries. However, most of these new programs have had the cooperation of our former graduates as faculty members and directors. The current curriculum includes mathematics, physics, music, economics, and administrative, computing, and engineering sciences, as well as acoustics and sound engineering technology. The working field of these graduates has been very wide, including areas such as recording and sound reinforcement, noise and vibration control, environmental protection, and acoustic design of architectural spaces. Many of our graduates have obtained advanced degrees abroad, working with success not only in Chile but also in several countries in Latin America, Europe, and the U.S.

2:50

2pED8. A brief history of the undergraduate acoustical engineering programs at the University of Hartford. Michelle C. Vigeant and Robert D. Celmer (Acoust. Prog. and Lab., Dept. of Mech. Eng., Univ. of Hartford, 200 Bloomfield Ave., W. Hartford, CT 06117, vigeant@hartford.edu)

Located in West Hartford, CT, the University of Hartford’s undergraduate mechanical engineering program has had required coursework in acoustics and vibrations for over 40 years. The evolution of two undergraduate degree programs with a focus in acoustics will be discussed: (1) the Bachelor of Science in Mechanical Engineering (BSME) with Acoustics Concentration and (2) the interdisciplinary Bachelor of Science in Engineering (BSE), Acoustical Engineering & Music, which requires acceptance into the University’s music conservatory, The Hartt School. The programs provide both theoretical instruction on the concept of acoustics and vibrations as an inherent element of engineering design, and the benefits of applied, sponsored full-semester design projects. The program has evolved to incorporate real-world projects during the sophomore (second) year of the 4-year program, in addition to the well-established design project coursework during senior (fourth) year. A description of the acoustic and vibration facilities and equipment used by our students will be given, in addition to the results of a recent ABET accreditation visit.

3:05—3:20 Break

3:20

2pED9. Is acoustic comfort underestimated by architects? Elvira Viveiros (GAAMA, Univ. of Santa Catarina, CxP 476, Florianopolis, SC 88040-900, Brazil, elvira@pq.cnpq.br)

A survey was carried out among architects and undergraduate students in order to identify their understanding of acoustics. It can be seen that some myths in architectural acoustics are solidly established. Simultaneously, the viewpoint of lay people was also looked into and it was found that their demands are often in opposition to what architects consider relevant. The reason why what is expected in buildings is not delivered by the architects are discussed and their academic profile is investigated.

3:35

2pED10. Ideas for effectively teaching architectural acoustics to students of architecture. Lily M. Wang (Architectural Engr. Prog., Univ. of Nebraska—Lincoln, Omaha, NE 68182-0681, lwang4@unl.edu)

Because the process of designing and constructing buildings often has the architect in the role of project manager, it is crucial that architects be trained in the fundamentals of architectural acoustics. Students of architecture, though, are often not interested in that required course on acoustics (if there is one), as their education tends to push them to be more visually inclined. This paper provides some ideas on how to effectively teach architectural acoustics to students of architecture, based on the author’s own experiences. In particular, (a) engaging the students to experience sound in rooms through a course-long listening journal assignment, (b) pulling in case study examples, and (c) promoting a collaborative environment between architects and consulting engineers have seemed to be successful techniques. A brief review of the Robert Bradford Newman Student Award Fund, whose goal is to promote the study of architectural acoustics among those in the building industry, will also be provided.
3:50

The presentation describes the experience of designing and teaching the course “The Physics of Music and Language” at Universidad de Costa Rica. Even though the course is taught at the Physics Dept., it has drawn large enrollment from music students as well as other majors such as education, biology, and the social sciences. The course has now been taught five times and has become part of the mandatory curriculum for music students. In this course, a traditional approach to acoustics is extended to include topics on psychoacoustics as well as a discussion of the anthropology of music and the relationship between music, language, and poetry. This makes the course truly transdisciplinary. Rather than despairing at the heterogeneity of the audience, it is argued that the situation should be seen as an opportunity to generate debate among students from diverse majors.

4:05

For decades the technology of binaural beats has been used effectively to produce several brain states associated with profound psycho-physical relaxation and altered states of consciousness for learning. These rhythms were discovered by Dove in 1839, but only when it was published by Oster in 1973 was its significance appreciated. It seems that these states are created by the beats through the nucleus of the medulla to the cerebral cortex, with frequencies below 30 Hz, by the combination of two higher frequencies, which generates a third one in the required range. Then the brain follows this signal. Although these sounds are used for example, to promote creative statements, improve concentration and memory, and to help eliminate anxiety, insomnia, depression, chronic fatigue, headaches, or psycho-physical, it must be kept in mind that not everyone respond the same way to the same stimuli. The general idea is to apply it in order to improve learning.

4:20
2pED13. Information technologies applied to acoustics learning. Maria Teresa Franco Martinez, Amparo Vazquez Saldaña, and Jose de Jesus Negrete Redondo (ESIME, IPN, ICE, ACUSTICA, Maestranza Lote 12, Mza I Col. San Fdo., Huizquilucan Edo. C.P. 52765, Mexico, acusticafranco@gmail.com)

The rather new information and communication technologies are computing tools that transmit, process, stores and present information in several different ways. The teaching process based on these techniques can promote diverse experiences due to interaction with peers and collaborative learning. This paper is developing within the acoustics field in the topic Electroacoustics and Transducers, where two groups of the seventh semester (BSc level) participate in the experiment, one as a control group subject to regular teaching, while the other works with the techniques described. At the end, an evaluation of both groups will show whether there is a significant difference.

1:30
2pMU1. A method for comparing sounds of similar instruments. Sandra Carral (Univ. of Music and Performing Arts, Vienna, Inst. of Musical Acoust., Anton von Webern Platz 1, 1030 Vienna, Austria)

Although the radiated sound of similar musical instruments (e.g., two recorders made by the same maker) can be compared using an artificial excitation apparatus to eliminate a player-induced effect on the sound, this introduces another problem. Even if differences are found, the question remains as to whether these differences are musically significant: Would the musician not be able to compensate for these differences, making them irrelevant? One way of overcoming this problem is to have a musician play the instruments in a manner as close to that of real performance as possible. This paper presents a method for comparing the sounds of similar musical instruments, whereby a musician is asked to play a known piece many times on each instrument. The radiated sounds are recorded and subsequently spectrum-analyzed using the program SONDAN. Later on, time-varying parameters such as the rmx amplitude, fundamental frequency, and spectral centroid as well as average and standard deviations of these parameters are calculated for each recorded sample for each instrument. Statistical tests are made in order to establish whether statistically significant differences between the instruments exist. Examples of this method will be presented.
2pMU2. Visually supported analysis of music. B. Kostek (Multimedia Systems Dept., Gdansk Univ. of Technol., Narutowicza 11/12, 80-230 Gdansk, Poland, bozenka@sound.eti.pg.gda.pl)

Subjective tests are highly regarded means for determining musical sound timbre, sound and music quality, audio-visual correlations, etc. The multimedia development created a new domain of interests, namely, quality of experience, in which subjective evaluation has proven to be one of, if not, the most important tool. It should be pointed out that experience is not only subjective but also context-dependent. In addition to the external factors related to the environment aspects, perception and cognitive state clearly play a critical role in the context. Despite significant advances in objective measuring methods, the only way to evaluate subjective audio-video signal quality is to acquire opinions of subjects, and then to quantify resulting choices. Many multimedia applications stimulated new methods of subjective evaluation. Also, audio-visual perception evaluation requires a new methodological approach, which can be fulfilled while employing gaze-tracking technology. Gaze-tracking-based experiments consist in determining the part of the screen that the user is looking at and in superimposing it against an audio-video content. This method allows for measuring the attention and performance of the subject, thus making the subjective tests more reliable. The proposed methodology based on the gaze-tracking technique applied to audio-visual correlation tests will be shown.

2:10

2pMU3. Perception and acoustical analyzes of traditionally orchestrated musical structures versus non-traditional counterparts. Roger A Kendall (Music Cognition and Acoust. Lab., Univ. of California, Los Angeles, Los Angeles, CA 90095, kendall@ucla.edu) and Pantelis N. Vassilakis (Columbia College, Chicago, IL 60605)

Many studies use single-tone or dyadic-unison stimuli in the exploration of musical timbre. Musical contexts, however, often employ larger numbers of simultaneous voices. Recently, we have conducted perceptual/acoustical studies of consonance, dissonance, and roughness in multitimbral triads often finding that chord quality (major, minor, diminished, augmented) had perceptual primacy over timbre. The present research extends this work to include major and minor triads in similarity scaling, blend, and identification experiments. In addition, four-tone chords were orchestrated according to procedures articulated in orchestration texts: juxtaposition, interlocking, enclosure, and overlapping. Sampled instrument tones (Sibelius/Kontakt) of oboe, flute, and clarinet were used, allowing comparison to previous research, and non-traditional combinations with saxophone and trumpet were included. Triads with each tone assigned a different timbre provided nontraditional orchestrations to be compared and contrasted with traditional orchestrations that duplicated timbres within a chord. Perceptual data were correlated to acoustical analyzes of spectral distribution and time-variance.

2:30

2pMU4. Estimation of reed flow signal from instrument performance. Tamara Smyth (School of Computing Sci., Simon Fraser Univ., 250-13450 102nd Ave., Surrey, BC V3T 0A3, Canada tamaras@cs.sfu.ca) and Jonathan S. Abel (Stanford Univ., Stanford, CA 94305-8180)

In this work we present a technique for estimating the reed flow signal, typically a periodic sequence of pulses, from the recorded sound of a reed instrument. The instrument is modeled as a reed coupled to a 1-D waveguide having unknown filter elements that must first be determined before constructing the instrument reed flow transfer function. As pressure waves make two round trips from the mouthpiece to the bell and back for each reed pulse, the output periodic pressure has two distinct halves: the second half being roughly the first half filtered by the instrument’s propagation losses. Estimation of these losses is not simply a spectral ratio, as the two halves are not temporally disjoint and the beginning of the reed pulse period is often unclear. The running autocorrelation of the recorded signal is zero phase and naturally provides the beginning of the period of the recorded signal as well as clear first and second phases that may be analyzed to estimate the round-trip losses in the instrument. Combining these losses with the direct measurements of the bell reflection function, a filter is developed which inverts the implied waveguide to produce the reed flow estimate.

2:50


In previous waveguide synthesis models for piano, longitudinal waves have been neglected, although it is known that there is audible coupling from transverse to longitudinal vibration in piano strings [e.g., Conklin lectures, http://www.speech.kth.se/music/5/\_lectures/]. A general method for accurate 3-D string simulation is the mass-spring chain [Rowland and Pask, Am. J. Physics, (1999)], which reduces to half-coupled wave equations at low amplitude. At yet lower amplitudes, when string slope times string curvature can be neglected, the longitudinal and transverse waves reduce to linear superposition, for which digital waveguides are most efficient for simulation [http://ccrma.stanford.edu/~jos/pasp/]. In this presentation, a hybrid piano-string model is proposed which employs a mass-spring chain at the hammer and a digital waveguide model elsewhere. The spatial force vector at either edge of the mass-spring section drives (and couples) three digital waveguide models corresponding to one longitudinal and two transverse vibrational components. At the bridge, a 3-D force vector is formed which drives the vertical, horizontal, and longitudinal transfer functions of the soundboard. A simplified 2-D model neglects one of the transverse planes of vibration. Beyond piano strings, the model can transition adaptively (across time and/or space) among digital waveguides, half-coupled finite-difference schemes, and full mass-spring-chain models.
3:10—3:25 Break

3:25

2pMU6. Playing the Continuum Fingerboard: New performance techniques for a continuous instrument. Lippold Haken (Elec. and Comput. Eng., 330G Everitt Lab, Univ. of Illinois, Urbana, IL 61801, L-Haken@illinois.edu) and Edmund Eagan (Twelfth Root, Ottawa, ON K1Y 3E5, Canada )

The Continuum Fingerboard is a Midi performance controller that offers real-time 3-D continuous control for every finger that is placed on its playing surface. Its novel design presents unique performance challenges and possibilities. Like a fretless instrument, the performer must rely on audio feedback, finger memory, and manual dexterity for accurate intonation and expression. The Continuum requires its own technique, different from any other instrument. With a traditional music keyboard, it is normal for the performer to feel the key hit a hard stop for each note, even if the performer is playing quietly. Also, traditional keyboards are usually velocity sensitive; a single velocity value is determined by the speed of the key movement. In contrast, the Continuum is both velocity sensitive and, more importantly, pressure sensitive. It initially outputs a single velocity level, and then continually outputs a stream of pressure, pitch, and front-back position updates until the note is terminated. The Continuum is designed for a fast and relatively lighter touch than other musical instruments, so it is unusual to hit the hard stop (or “bottom out”) except for the very loudest notes. Various Continuum-specific performance techniques utilized by performers will be presented.

3:45

2pMU7. Sound design for the Continuum Fingerboard: Effective mapping of finger data to synthesis parameters. Edmund Eagan (Twelfth Root, 953 Gladstone Ave., Ottawa, ON K1Y 3E5, Canada, ed@twelfthroot.com) and Lippold Haken (Univ. of Illinois, Urbana, IL 61801)

The Continuum Fingerboard tracks the position and pressure of fingers on its playing surface. Its internal computer encodes this finger placement information into Midi data streams. The Continuum has an internal dsp that provides high-quality on-board timbres, but often performers expand on this built-in sound palette with sounds from external synthesizers. The Continuum connects to synthesizers via Midi, and/or to analog modular synthesizers via the Continuum voltage converter. It can be used to control a wide variety of sound synthesis algorithms. When designing sounds for the Continuum, it is important to keep in mind that it is not a keyboard. Sounds that work well with keyboards generally do not work well with the Continuum and will not exploit the full performance potential of the instrument. Sound design techniques for producing high-quality synthesis algorithms suitable for the Continuum will be presented.

4:05—4:10 Break

4:10—4:40

Mini-Concert on the Continuum Fingerboard

Edmund Egan will give a demonstration performance on the continuum fingerboard

TUESDAY AFTERNOON, 16 NOVEMBER 2010

Session 2pPA

Physical Acoustics: General Physical Acoustics II

Claire Prada, Chair

CNRS, Institut Langevin/LOA, ESPCI, 10 rue Vauguelin, 75005 Paris, France

Contributed Papers

2:00

2pPA1. Zero group velocity Lamb modes and edge resonance in a semi-infinite plate. Maximin Ces, Dominique Chorenc, Daniel Royer, and Claire Prada (Laboratoire Ondes et Acoustique, Institut Langevin, ESPCI, CNRS UMR 7587, 10 rue Vauguelin, 75005 Paris, France)

The local resonances of an elastic plate with a free edge are investigated using laser ultrasonic techniques. It is known that the long wavelength vibrations of a plate can be interpreted in terms of Lamb modes. Low-frequency vibrations can be ascribed to the flexural A0 mode, while high-frequency shear or stretch vibrations occur at the cutoff of higher order Lamb modes. These thickness resonances correspond to an in-phase motion of the whole plate, associated to vanishing Lamb wave number. An elastic plate can also support local vibrations at nonzero values of the wave number for which Lamb mode dispersion curves undergo a minimum. Recently, experiments using laser ultrasonic techniques demonstrated that the local resonance of an infinite elastic plate is entirely governed by these zero group velocity (ZGV) Lamb modes. For a semi-infinite plate, ZGV modes can be observed at a distance as close as two thicknesses from the edge. Coming toward the edge, ZGV resonances slowly vanish, whereas the edge mode resonance appears, whose frequency depends on the thickness and on the bulk velocities of the plate. Using a laser excitation on the edge, the amplitude profile of the normal displacement of the edge was determined.
2:15

2pPA2. Non contact thin film local thickness measurements through a thick plate using zero group velocity Lamb modes. Maximin Ces, Claire Prada, Dominique Clorennec, and Daniel Royer (Laboratoire Ondes et Acoustique, Institut Langevin, ESPCI, CNRS UMR 7587, 10 rue Vauquelin, 75005 Paris, France)

It has been shown that a homogeneous isotropic plate can support nonpropagating Lamb modes having a zero group velocity (ZGV) for nonzero values of their wave number. Optical means such as a pulsed laser source and a heterodyne interferometer can be used for non-contact generation and detection of these elastic waves. With this experimental setup, local and accurate measurements of material parameters such as bulk wave velocities and very small plate thinning detection were obtained from the resonance spectrum of ZGV Lamb modes. A thin layer deposited on the plate induces a significant down shift of the set of ZGV resonance frequencies. Experimentally, thin gold layers down to 100 nm were detected through a 1.5-mm thick Duralumin plate. Optical measurements were performed at the same point as the generation on the non-covered face of the thick plate. The frequency shift, which is typically 1 kHz for the $S_2$-Lamb mode at 1.9 MHz, can be approximated by a formula, which allows us to calculate the layer thickness. Since the material parameters are temperature-dependent, the influence of temperature was also studied.

3:00—3:15 Break

3:15

2pPA5. Squeal acoustic emissions and the stick-slip effect. A. J. Patitsas (Dept. of Phys. and Astronomy, Laurentian Univ., Sudbury, ON P3B 3E4, Canada)

The origin of the squeal acoustic emissions when a chalk is rubbed on a blackboard, or better on a ceramic plate, and those when a wet finger is rubbed on a glass surface is sought in the stick-slip effect between the two surfaces. The elastic agency that determines the frequency of the stick-slip process is sought in a thin shear band between the two surfaces, characterized by a very low shear modulus. In the case of the squealing chalk, the shear band is a layer of chalk powder, about 0.3 mm thick, forced to slide over the ceramic plate surface. Similarly, in the case of the wet finger, the shear band is a water layer forced to slide over the glass surface. The stick-slip resonance effect becomes coupled to the mode, in the shear band, characterized by high stability and by an anti-node of the particle displacement at the sliding interface. The application of such concepts to the acoustic emissions from singing (sand) grains impacted by a rod will also be discussed.

3:30

2pPA6. Interferometric nanogold sensor with sensibility-controlled acoustical signal. Carlos Torres-Torres (SEPI-ESIME-IPN, Zacatenco, DF 07738, Mexico, cttorres@ipn.mx), Reydezel Torres-Martinez (CICATA-IPN, Qro 76900, Mexico), Javier Moreno-Valenzuela (CITEDI-IPN, Tijuana, BC 22510, Mexico), and Martin Trejo-Valdez (ESIQIE-IPN, Zacatenco, DF 07738, Mexico)

Experimental and numerical results for acoustical wave detection performed by a nanostructured interferometric system with optical excitation wavelengths near the surface plasmon of resonance of Au nanoparticles are presented. A colloidal gold sample located at the arm of a Michelson interferometer was used. It is shown that the presentation of metallic nanoparticles with different sizes in a fluid can be driven by acoustical frequencies in order to induce changes in the properties of the sample. The measurements are based on the identification of the optical transmission associated with the distribution density that results when an acoustical signal is in propagation through the sample. The modification on the absorptive and refractive optical response of the sample containing Au nanoparticles was investigated. It is demonstrated that a high accurate stabilization of the optical properties of the sample can produce important differences in the sensibility of the interferometric system for detecting other physical perturbations by remote sensing. [Financial support from CONACyT through Grant nos. 80024 and 82708; from IPN through Grant nos. SIP-20100800, SIP20100836, SIP20100553, and SIP 20100564; and also from COFAA is gratefully acknowledged.]
length. An important benefit of the above approach is that the devices can generate electricity by means of piezoelectric elements; this provides energy conversion from heat to sound to electricity. The working fluid is air at 1 atm.

4:00
2pPA8. Low-cost thermoacoustic co-generator for biomass-burning cook stoves. Paul J. Montgomery, Jr. and Steven L. Garrett (Grad. Prog. in Acoust., Penn State Univ., P.O. Box 30, State College, PA 16804)

Three billion people cook daily on an open fire or primitive stove that creates so much pollution that it has been shown to be directly responsible for 1.6×10⁶ deaths each year, almost exclusively among women and young children. Atmospheric models suggest that black carbon and methane released during biomass combustion are also the primary anthropogenic contribution to global warming from the populations of Southeast Asia, sub-Saharan Africa, and India. Recent research demonstrates that fan-aided combustion creates a significant decrease in products of incomplete combustion that degrades indoor air quality and exacerbates global warming. The use of a fan also increases efficiency, improves fuel flexibility, and reduces cooking times. Unfortunately, 1.6×10⁶ biomass burners have no access to electricity, yet each cook stove produces 4–8 kW of heat. This presentation will describe a very simple co-generator prototype that extracts a small fraction of the stove’s heat and uses air at atmospheric pressure in a lumped-element, stack-based, thermoacoustic engine to vibrate an inexpensive moving-coil electrodynamic loudspeaker acting as a linear alternator that can generate the few watts of electrical power necessary to operate the fan and charge a battery. [Work supported by Paul S. Veneklasen Research Foundation.]

4:15
2pPA9. Soil profile meter with ultrasonic sensor. Ana Trinidad Garcia Martinez, Jose Rodolfo Martinez y Cardenas, and Jaime Ruiz Vega (CIIDIR Unidad Oaxaca IPN, Calle Hornos No. 1003, Col. Noche Buena, Santa Cruz, Xoxocotlan, CP 71230, Mexico, ana.garcia.martinez@hotmail.com)

The Agricola research at CIIDIR Oaxaca requires characterization of the line profile made by the seeding appliances, in order to allow the people involved to get a surface description for quality control purposes. This paper presents the design of a profile meter with a non contact ultrasonic sensor, so the soil surface is not altered. The system is programmable, simple to use, and stores the measurement results.
2pSC1. A comparison of suprasegmentals and segmentals in Indian English, Hindi, and Telugu. Hema Sirsa and Melissa A. Redford (Dept. of Linguist., Univ. of Oregon, Eugene, OR 97403-1290, hsirsa@uoregon.edu)

Indian English (IE), an official language of India, differs suprasegmentally and segmentally from other English dialects. This study explores whether IE sound structure varies with the divergent L1s of its speakers, as suggested by Wiltshire and Harnsberger (2006), or whether its sound structure is similar regardless of speakers L1. To investigate this question, measures of rate, rhythm, and final lengthening were taken in Hindi (Indo-Aryan), Telugu (Dravidian), and in IE, which was produced with native fluency by the same five Hindi and five Telugu speakers. Vowel and obstruct segments, common to all languages, were also extracted from the 13 stimulus phrases and acoustically analyzed. The results indicate some influence of the different L1s on IE rhythm structure and VOT, but none on IE speech rate, final lengthening, or vowel and /l/ production. Mostly, the results indicated that when language differences did exist (and there were many similarities across the languages), these were more likely to be between Hindi and IE or between Telugu and IE or between Hindi and Telugu than between the IE produced by speakers of different L1s. Such results suggest that IE is a pan-India dialect of English, not merely an L1 influenced L2 for Indians.


Individuals become specialized through development to perceive native-language speech sounds, but it is not certain whether this specialization occurs solely at the level of phonological categorization or whether precategorical auditory processing also becomes specialized. The present study examined this issue using 11 continua based on English /w/-/v/. The stimuli formed a graded set from naturalistic to non-speech by progressively making the source spectrum less natural and removing acoustic cues. The aim was to examine how close a stimulus must be to natural speech in order for cross-language differences to emerge. Native Hindi speakers, who have /w/-/v/ identification difficulties when learning English as a second language, and native English speakers were tested in terms of discrimination and...
characterizing talker similarity spaces are similar in the first and third phases for both groups of listeners.

2pSC6. Linguistic effects on speaker discrimination. Chandan R. Narayan (Dept. of Humanities, Univ. of Toronto Scarborough, 1265 Military Tr., Toronto, ON M1C 1A4, Canada; chandan.narayan@utoronto.ca) and Molly Bab- 

We report the results from experiments aimed at measuring linguistic in- fluences on speaker discrimination. In experiment 1, listeners heard two speakers in each trial (10 total speakers balanced within and across gender), each saying a single English word with a 100 ms ISI. Listeners were asked to determine whether the two words were spoken by the same speaker or not in a 2AFC design. In some cases the two words formed a common comp- pound (i.e., year-book) and others a nonsense compound (i.e., year-crow). Listeners’ response and reaction times were measured as a function of seman- tic cohesion. Experiment 2 examined speaker effects on listeners’ knowledge of compounds. Listeners were asked whether two words, spoken by either the same or different speaker, formed a compound or not. Results are discussed in terms of the facilitative and inhibitory effects of semantic cohesion on speaker discrimination.

2pSC7. The impact of talker and prosodic variability on speech perception in noise for children with dyslexia. Valerie Hazan, Stuart Rosen, and Souhila Messaoud-Galusi (UCL Speech, Hearing and Phonetic Sci., Chandler House, 2 Wakefield St., London WC1E 1PF, UK; v.hazan@

This study investigated whether children with dyslexia (DYS) are more affected by talker and prosodic variability when perceiving speech in noise than age-matched average readers (ARs). 34 DYS and 25 AR children car- ried out syllable-initial consonant perception tests, with natural consonant- vowel (CV) tokens in multi-talker babble noise (0 dB signal-to-noise ratio). Twelve-alternative identification tests were presented in four conditions varying in degree of talker and intonation variability. The discrimination of place (/bi-di/) and voicing (/bi-p/i) contrasts was investigated with the same test conditions. When identifying CV syllables in noise, the DYS group made more errors than the AR group, but only for conditions with variable intonation. This group difference was primarily due to poorer perception by DYS children of fricative voicing and greater /nl/-/l/ confusions, i.e., con- trasts with less salient acoustic differences. DYS children showed lower dis- crimination scores than AR children for both the voiceless and place contrasts, across the single/multiple talker and single/multiple intonation conditions. These results are compatible with the view that some children with dyslexia have speech perceptual acuity at the lower end of a normal range, which may be exacerbated by non-sensory factors, and thus affected by the type of task used. [Work funded by the Wellcome Trust (076499/Z/05/Z)].

2pSC8. Predicting the effect of talker differences on perceived vowel category. Antonia D. Vitela, Brad H. Story, and Andrew J. Lotto (Dept. of Speech, Lang., and Hearing Sci., Univ. of Arizona, 1131 E. 2nd St., Tucson, AZ 85721; adv1@email.arizona.edu)

It has been clear for more than 50 years that listeners sometimes shift their categorization of target vowels as a function of the information in pre- ceding carrier phrases. These results have been interpreted as evidence for a talker normalization process that allows listeners to tune their speech per- ception to the peculiarities of individual talkers. However, it is still not clear what information is being extracted about the talker from the context phrases. In the current set of studies, carrier phrases (“He had a rabbit and a”) spoken by different “talkers” were produced by manipulating the length and oral/pharyngeal cavity ratios of an articulatory synthesizer. [Story, J. Acoust. Soc. Am. 117, 3231–3254 (2005).] These phrases preceded a series of target stimuli varying from “bit” to “bet” produced by the standard vocal tract for the synthesizer. Shifts in the perceived target vowel were obtained for some of these vocal tract manipulations but not others. The presence /absence of effects was most readily predictable from the long-term spectral average of the carrier phrase. These results support the idea that listeners are extracting general acoustic information from the context and not specific in- formation about the talker’s vocal tract. [Work supported by NIH-NIDCD.]
Previous research has shown that 14-month-olds are able to form arbitrary word-object associations. [Weker et al. 1998;] However, merely associating words and objects is insufficient for the higher level concept that words “stand for” objects. In order to investigate at what age referential understanding emerges, we tested 14- and 17-month-olds on their ability to comprehend the correct meaning of a word when the word and object are presented separately from one another. We used an infant-controlled version of the visual fixation procedure presented using HAPI x [Cohen et al. (2004).] Infants heard audio passages containing two target nouns. Subsequently, infants were presented with silent animations of four words—two with which they were familiarized (target) and two they had not heard during familiarization (non-target). Looking time to target and non-target animations were averaged and compared using paired t-tests. The 14-month-olds looked longer at the target animations although this result was not significant. In order to identify at what age infants understand the referential relationship between words and objects, we are currently testing 17-month-olds on the same task. The results will be discussed in the context of emerging referential understanding of nouns during development.

Infants' word segmentation is facilitated by bottom-up cues, such as prosodic cues (stress-patterns, position in a sentence, etc.). Research shows that infants also use top-down cues, i.e., known words, to segment new words [Bortfeld et al. (2005); “mommy” and the child’s name]. In two experiments, infants were tested on their ability to use a known function word “the” to segment vowel-initial words (the ability to segment them does not appear until 13.5–16 months), without any additional bottom-up cues. In experiment 1, using the headturn preference procedure, monolingual English-learning 11-month-olds were tested on their ability to segment sentence-medial vowel-initial words that followed the function word “the.” First, infants were familiarized with passages containing the target words “ash” and “eff.” Next, infants were presented with two familiar and two novel isolated words. Mean listening time to the familiar and unfamiliar words was significantly different [(t(12) = 4.220, p < 0.001). Out of 13 infants, 12 listened longer to the familiar words compared to the novel words. In experiment 2, monolingual English-learning 8-month-olds are currently being tested using the same task. Preliminary results show three out of five 8-month-olds listen longer to the familiar words compared to the novel words.

The results of experiment 1 are interpreted as advancing the claim that humans have an unlearned, domain-specific attentional bias in favor of phonetic cues. [Bortfeld et al. 2005] In experiment 1, monolingual English-learning 8-month-olds are currently being tested using the visual fixation procedure. Monolingual 4- and 8-month-old infants were presented with side-by-side videos of a bilingual speaker producing English and Spanish sentences. Infants’ looking time to each video was calculated and compared statistically. Preliminary data from this hypothesis is indirect (e.g., infants take longer to learn contrasts with overlapping acoustic cue distributions) and the more direct evidence, from laboratory training studies, is not straightforward. Two published papers show that infants’ perception is affected by different frequency distributions along voice onset time [Maye et al., Dev. Sci. 11, 122–134 (2008);] and children’s perception of non-native vocalic VOT discontinuities, whereas category formation is harder when such discontinuities are not available. If so, different contrasts would be learned at different rates. To test this hypothesis, a continuous training paradigm was developed to assess the timecourse of infant perceptual learning of categories that depend on very different acoustic correlates (e.g., stop voicing; Mandarin tones T2–T4).

Many models of infant category learning attribute a crucial role to distributions of acoustic cues. Unfortunately, most of the evidence in favor of this hypothesis is indirect (e.g., infants take longer to learn contrasts with overlapping acoustic cue distributions) and the more direct evidence, from laboratory training studies, is not straightforward. Two published papers show that infants’ perception is affected by different frequency distributions along voice onset time [Maye et al., Dev. Sci. 11, 122–134 (2008);] and children’s perception of non-native vocalic VOT discontinuities, whereas category formation is harder when such discontinuities are not available. If so, different contrasts would be learned at different rates. To test this hypothesis, a continuous training paradigm was developed to assess the timecourse of infant perceptual learning of categories that depend on very different acoustic correlates (e.g., stop voicing; Mandarin tones T2–T4).
4-month-olds show that monolingual English-learning infants look longer to the English stimuli. Results will be discussed in the context of the influence of visual cues in auditory speech perception.

2pSC15. Speech discrimination of English, Spanish, and Mandarin consonants in monolingual and bilingual infants: An event related potentials experiment. Adrian Garcia-Sierra, Nairan Ramirez-Esparza, and Patricia K. Kuhl (I-LABS, Univ. of Washington, Fisheries Ctr. Bldg., Box 357988, Seattle, WA 98195-7988, gasa@uw.edu)

The goal of this investigation was to compare speech discrimination in 11–14-month-old infants who are being raised in either monolingual English (N=22) or bilingual Spanish-English (N=22) homes. The participants’ neural activity associated with the ability to discriminate English, Spanish, and Mandarin Chinese consonants was assessed using event related potentials (ERPs) in separate ERP sessions. The goal of the experiments is to compare ERPs for the phonetic units that infants have experienced as opposed to those they have never been exposed to. The native language neural commitment (NLNC) hypothesis holds that, by 11 months of age, infants respond differently to the phonetic units of native language(s) as opposed to non-native language(s). Comparisons between monolingual and bilingual infants should help resolve whether monolingual and bilingual infants are on the same timetable regarding NLNC. Our results will provide evidence about how early language exposure influences speech discrimination in monolingual and bilingual infants. [Work supported by an NSF Science of Learning Center grant to the UW LIFE Center.]

2pSC16. Neural correlates of dialect perception in early infancy, Natalia Egorova (Univ. of Groningen, Broerstraat, 5 P.O. Box 72, 9700AB Groningen, Netherlands, and LSCP, 29, rue d’Ulm, 75005 Paris, France, natafka@gmail.com), Alejandra Cristiá, Ingrida Vendelin, Luca Filippini, Bria Long (LSCP, 75005 Paris, France), Judit Gervain (45, rue des Saints-Pres, 75006 Paris, France), Yasuyo Minagawa-Kawai (Keio Univ., Tokyo 108-8345, Japan), and Emmanuel Dupoux (LSCP, 75005 Paris, France)

A wealth of behavioral research suggests that infants become increasingly specialized in their native dialect/language in infancy. In fact, several studies document how this early specialization is reflected in neural activation, most of which have compared familiar and unfamiliar languages, and none focused on different dialects. This study aimed to fill that gap, focusing on cerebral activation in temporal areas, as measured with near infrared spectroscopy. Audiovisual infant-directed speech was recorded from talkers of either Parisian or Quebecois French. These videos were presented to 5-month-old Parisian infants in blocks within which videos from two talkers alternated in one of two ways. In pure blocks, both talkers were either Parisian (pure-familiar) or Quebecoens (pure-unfamiliar). In mixed blocks, the two talkers had different dialects. A robust and mostly bilateral activation was found for both mixed and pure blocks. Follow-up comparisons revealed a stronger activation for mixed blocks than for pure ones, and a trend for more activation in response to the unfamiliar dialect than to the familiar one. These findings are congruent with behavioral studies showing early sensitivity to dialects and extend the brain imaging literature on early neural attunement to the language as spoken in the infant’s environment.

2pSC17. A chimpanzee (Pan troglodytes) recognizes spoken words synthesized as sine-wave speech. Lisa A. Heimbauer, Michael J. Beran, and Michael J. Owren (Dept. of Psych. and The Lang. Res. Ctr., Georgia State Univ., P.O. Box 5010, Atlanta, GA 30302-5010)

The human ability to understand speech in the absence of traditional acoustic cues to phonetic content has been argued to be evidence of specialized processing. To determine whether this capability is unique to humans, perception of sine-wave speech was examined in a 23-year-old, language-trained chimpanzee named Panzee. This animal was reared from infancy by human caregivers in a speech-rich environment and identifies more than 125 spoken words using graphical symbols (lexigrams). Test trials presented one of 48 familiar words via computer in either natural or sine-wave form, with Panzee choosing a corresponding lexigram from among four alternatives. Her performance on sine-wave words was well above chance levels, in spite of receiving no reward or any other feedback when responding to these sounds. The chimpanzee also showed above-chance accuracy on trials that cumulatively represented the first instances of hearing words in sine-wave form. While she was less accurate with sine-wave words than with natural versions, human listeners hearing the same stimuli were less accurate as well. Panzee’s performance indicates that experience with spoken language and general auditory-processing mechanisms alone can be sufficient for perceiving even highly impoverished synthetic speech.


The purpose of this study was to examine how interlingual homophones of Canadian English (CE) and Canadian French (CF) differ along acoustic dimensions as a prelude to perceptual tests. Two male Canadian monolingual speakers of English and French produced sentences in which interlingual homophones were embedded (e.g., two /tou/ and /tou/ meaning all). Voiced onset times (VOTs), formant frequencies (F1 and F2), and vowel-inherent spectral changes (VISCs) of each respective speaker were examined. First, it is expected that the CE monolingual speaker will have aspirated (long-lag) voiceless stops, while the CF monolingual speaker will have unaspirated (short-lag) voiceless stops regardless of the context. Second, it is expected that CF vowels will have lower F1 values for lax vowels and will be produced more peripherally (with respect to F2 values) than CE vowels (i.e., French front vowels will be more advanced, and French back vowels will be more posterior). Lastly, it is expected that CF vowels will have less vowel-inherent spectral change than CE vowels. Results from this study will provide insight on the phonetic representation of interlingual homophones, enabling the understanding of acoustic-phonetic influences on the processes of speech perception in bilinguals (CE and CF).

2pSC19. Perceived similarity of English, Korean, and Japanese stops for native English, Korean, and Japanese listeners. Takeshi Nozawa (Lang. Education Ctr., Faculty of Economics, Ritsumeikan Univ. 1-1-1 Noji-higashi, Kusatsu, Shiga 525-8577, Japan) and Sang Yee Cheon (Univ. of Hawaii at Manoa, Honolulu, HI 96822)

 Stops in /Ca/ context produced by four native speakers of English, Korean, and Japanese were presented in an AXB discrimination task to examine and compare interlingual stop identification, where the listeners heard three stimuli in sequence and decided whether the second (X) is more similar to the first (A) or to the third (B). Fifteen native speakers of the three languages heard three stops per trial, each of which was produced by a native speaker of a different language. The results revealed the effects of the listeners’ respective L1. While English and Korean listeners perceived Ko- rean aspirated stops as closer to English voiceless stops than to Japanese voiceless stops, Japanese listeners perceived Korean aspirated stops as closer to Japanese voiceless stops (which are said to be unaspirated or weakly aspirated). English and Korean listeners perceived English voice stops closer to Korean tense stops rather than Japanese voiceless stops, while Japanese listeners perceived English and Japanese voice stops as similar. These results suggest that perceived similarity of interlingual stops are affected by the phonotactical system of one’s L1, and may imply that perceived boundaries between voiceless and voiced stops and between aspirated and unaspirated stops are different across the listeners’ L1.

2pSC20. Perception and production boundaries between fricative and affricate in Japanese speakers. Shigeki Amano and Kimiko Yamakawa (Faculty of Human Informatics, Aichi Shukutoku Univ., 9 Katahira Nagakute, Nagakute-cho, Aichi-gun, Aichi 480-1197, Japan, psy@asu.aas.ac.jp)

Perception and production boundaries between a fricative /ts/ and an affricate /ts/ in Japanese were investigated to clarify their characteristics and relationships. In a perception experiment, 40 Japanese native speakers listened to 2-D stimulus continua coordinated with a rise time and the sum of steady-state and decay time of [st]ts [st]ts in 1–4-mora Japanese words. They made two-alternative forced choice between /ts/ and /ts/, and the identification ratio of these phonemes was obtained. In a production experiment, one female Japanese speaker pronounced 403 Japanese words in 1–4 moras long having /ts/ or /ts/, The pronounced words were digitally stored with 16-bit quantization and 16-KHz sampling frequency. A rise time and the sum of steady and decay to show how the words were measured with a waveform editor. Logistic regression analysis for perception data and discriminant analysis for production data revealed that the perception and production
boundaries were both represented by very similar linear functions with the rise time and the sum of steady and decay time of [s][ts]. This result shows a correspondence of perception and production boundaries between /s/ and /ts/, suggesting a close connection between speech perception and production.

**2pSC21. Perception boundary between fricative and affricate in Korean speakers.** Kimiko Yamakawa and Shigeaki Amano (Faculty of Human Informatics, Aichi Shukutoku Univ., 9 Nagakute Katahira, Nagakute-cho, Aichi-gun, Aichi 480-1197, Japan, jin@asu.aasta.ac.jp)

Previous study has shown that perception boundary between fricative /s/ and affricate /ts/ in Japanese speakers is represented by a linear function with two variables: a rise time and a sum of steady and decay time [Amano et al. (2009)]. This study investigated characteristics of the boundary in non-native Japanese speakers such as Koreans, because they often have trouble in distinguishing /s/ and /ts/. Thirty-two Korean speakers listened to 2-D stimulus continua coordinated with a rise time and the sum of steady and decay time of [s][ts] in 1–4-mora Japanese words and made two-alternative forced choice between /s/ and /ts/. A logistic regression analysis of the identification ratio of these phonemes showed that the perception boundary in Korean speakers was well represented by a linear function with the same variables as in Japanese, but the boundary shifted to a smaller value both in the rise time and in the sum of steady and decay time, which indicates a greater identification ratio of /ts/. The results suggest that Korean speakers’ perception of [s][ts] are affected by their native language, which does not have /ts/.

**2pSC22. The influence of language background on the relative perception of vowels.** Matthias J. Sjerps (Max Planck Inst. for Psycholinguistics, Wundtlaan 1, 6525 XD Nijmegen, The Netherlands, matthias.sjerps@mpi.nl) and Rajka Smiljanic (The Univ. of Texas at Austin, Austin, TX 78712-0198)

Individual differences in vowel production result in an extensive overlap of vowel categories in F1 × F2 vowel space, especially in languages with large vowel inventories. When perceiving vowel categories, listeners compensate for this variability by identifying vowels relative to the voice characteristics of individual speakers. The present study examines the effect of language background on the relative perception of vowels, i.e., whether listeners engage in vowel normalization differently in languages with vowel inventories of varied sizes and with different amounts of category overlap due to across-speaker variability. To that end, we compare English and Dutch, both with 10+ vowels and a substantial category overlap, and Spanish, with only 5 vowels and less category overlap. Vowel targets from an /aL–al/ continuum were embedded in sentences spoken by native speakers of all three languages. F1 of the precursor sentences was modified to be either high or low. Seventy-two listeners participated in the vowel categorization task. Preliminary analyzes reveal that listeners of all three languages compensate for F1 context similarly regardless of vowel inventory size and independent of their proficiency in another language. These results suggest that the normalization processes may be automatic and involve a language general mechanism.

**2pSC23. Acoustic characteristics of central vowels in American English.** Phoebe E. M. Allen and Gary G. Weismer (Dept. of Communicative Disorders, UW-Madison, 1975 Willow Dr., Madison, WI 53706, pallen1@wisc.edu

Surprisingly little is known of the acoustic characteristics of the central vowels caret ([t]) and ([a]), even though they occur frequently in English and contribute significantly to its rhythmic characteristics [Umeda, J. Acoust. Soc. Am. 58, 434–445 (1975)] Moreover, analyses of schwa in Dutch [van Bergem, Speech Comm. 16, 329–358 (1995)] show that its acoustic characteristics are surprisingly variable and very much affected by surrounding context. This paper reports temporal and spectral measurements of the vowels /t/ and /a/ produced by 10 speakers (5 male and 5 female) from central Wisconsin. These vowels were produced in real words in a variety of phonetic environments in both a reading task (three different rates of speech) and in spontaneous speech. Descriptive analyzes will be presented as well as discriminant function analysis in which the acoustic measures are used to classify the two vowels. The effects of speech rate and speech material on the two vowels will also be examined.

**2pSC24. Predicting vowel backness variability in Tommo-So.** Laura McPherson and Kevin Ryan (Dept. of Linguist., UCLA, 3125 Campbell Hall, Los Angeles, CA 90095, lmcpherson@ucla.edu)

Tommo-So (Dogon, Mali) has highly restrictive vowel phonotactics in stems. While any of its seven vowels may occur in the first syllable, the set of possible subsequent vowels is small, and noninitial nonlow vowels often exhibit a wide range of gradient variation in backness (while their phonological height remains fixed). Our study examines the second vowel of disyllabic words at all nonlow heights where (1) V1 = V2, (2) V1 is high and V2 is mid, and (3) V1 is mid and V2 is high, as well as the definite enclitic in various consonantal contexts. Preliminary acoustic analysis reveals that backness variation in V2 covers the whole range of front to back and yet is significantly less bimodal than would be expected if it were variation between discrete front and back allophones. Moreover, the majority of the variation cannot be attributed to coarticulation; that is, the same speaker will produce the same stimulus differently at different times. Nonetheless, if the first vowel is identical, variation is much restricted. We predict that grammatical morphemes such as the definite will display greater variation than lexical items.

**2pSC25. Regional dialect variation in standard Italian.** Terrin N. Tamati (Dept. of Linguist., Indiana Univ., Memorial Hall 322, Bloomington, IN 47405, ttmati@indiana.edu)

Italy is a linguistically diverse country where a standard language (standard Italian) is in contact with various regional languages. Within this setting, the regional languages have influenced the standard Italian spoken throughout Italy and regional dialects of Standard Italian have emerged. Previous studies on the regional dialects of Standard Italian have predominantly taken a descriptive approach discussing the phonological differences between the dialects and a broad acoustic study of the dialects is lacking. The present study seeks to identify several acoustic characteristics of regional varieties of spoken standard Italian. In the study, an acoustic investigation of vowels produced by male and female talkers from Northern, Central, and Southern Italy was carried out. Acoustic measures obtained included vowel duration and first and second formant frequencies. These measures were used to examine possible dialectal differences in vowel quality, vowel duration, and the overall acoustic vowel space. From these measures, several dialect-specific phonetic features were identified. These dialect-specific acoustic properties will be discussed. The results of the acoustic analysis will serve as a basis for future studies investigating the acoustic properties of the regional varieties of standard Italian and for identifying the types of cues listeners might use to distinguish the dialects.

**2pSC26. Regional variation in vowel acoustics in children.** Ewa Jaciewicz, Robert Allen Fox (SPA Labs, Dept. of Speech and Hearing Sci., The Ohio State Univ., 1070 Carmack Rd., Columbus, OH 43210, jaciewicz.1@osu.edu), and Joseph Salmons (Univ. of Wisconsin, Madison, WI 53706)

This study examines the acquisition of the American English vowel system by children who grew up in one of the three distinct dialect areas in the United States: western North Carolina, central Ohio, and southeastern Wisconsin. Of interest is the extent to which children acquire dialect-specific vowel dispersion patterns and dynamic formant movements (vowel-inherent spectral change). Forty-nine children (8–12 years) and 93 adults (51–65 years), males and females, produced 13 vowel categories in isolated hVd words. The results for children clearly show the presence of dialect-specific features found in adult speakers. The regional positional relations among vowels are generally maintained in children. For some vowels, the correspondence in the amount of spectral change and formant trajectory shape between children and adults is remarkable. Two major changes in children are common across dialects: reduction of formant movement in selected monophthongs and more uniform production of diphthongs in which dialectal differences present in adults tend to disappear. These results show that children are able to adopt and reproduce dialect-specific acoustic vowel characteristics quite well despite their exposure to highly variable input.
This study focuses on acoustic characteristics of vowels in the regional variety of Southern American English spoken in western North Carolina. It examines the cross-generational pattern of positional vowel changes with special reference to vowel-inherent spectral change. 140 speakers, males and females, ranging in age from 8-year-old children to adults in their early 90s, produced 13 vowel categories in isolated hVd words. All participants were born and raised in the area and speak the local dialect. The results show that the reorganization of the vowel system occurs gradually, generation by generation, and there is a correspondence between positional vowel changes and the changes in formant dynamics. In each age group, females represent a more advanced stage of vowel change compared to males. These data clearly document both the systemic nature of vowel change and close inter-dependent relationships between the nominal monophthongs and diphthongs. The categorical distinction between monophthong and diphthong changes across generations. The oldest speakers produce monophthongal diphthongs and diphthongal monophthongs, whereas younger speakers show increased monophthongization of the former and diphthongization of the latter. [Work supported by NIH.]
2pSP3. On the use of instantaneous acoustic intensity for the classification of defects in air-filled pipes. Kirill Horoshenkov, Tareq Bin Ali, and Simon Tait (School of Eng., Univ. of Bradford, Bradford BD7 1DP, UK)

The sound pressure in an air-filled pipe has been measured using six pairs of matched microphones. These data have then been 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, cracks, and water level 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 and the intensity spectrograms. The results are compared against that obtained via a standard 2-D cross-correlation method. The proposed MFP 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.

2:20


The problem of reproducing a desired sound field with a planar array of loudspeakers is addressed. The array is modeled by a continuous distribution of monopole-like secondary sources arranged on an infinite plane, and the reproduced field is represented by a single layer potential. The target field is defined on a plane parallel to the secondary source layer. An expression for the solution is derived and this can be regarded as a linear superposition of propagating and evanescent plane waves. It is shown that the contribution of the evanescent waves does not represent the evanescent component of the desired field and is not limited to the near-field of the array. For this reason this component of the field is called pseudo-evanescent. The case of the so-called focused sources is analyzed. It is shown that an exact solution of this problem does not exist, but an approximate solution can be computed using regularization techniques (for example, by avoiding an exact reproduction of the pseudo-evanescent component of the target field). The time-reversal technique, widely used by wave field synthesis systems for the reproduction of focused sources, is also studied.

2:40

2pSP5. Localization of sound sources in rooms using an improved version of steered response power algorithm. Jose J. Lopez, Maximo Cobos, and Amparo Marti (ITEAM, Universidad Politecnica de Valencia, Camino de Vera, 46022 Valencia, Spain, jjlopez@doc.upv.es)

Localization of sound sources inside a room is a challenging problem. The possible applications involving speech-based source localization systems range from teleconferencing to home automation systems. For example, the localization of a speaker inside a conference room can be very useful to place the speaker in a remote room by means of a spatial audio reproduction system. Also, new applications will appear in home automation if accurate source localization systems are available in the future. One of the most robust approaches to source localization is the SRP-PHAT algorithm, which has shown to provide very good localization results inside rooms with moderate reverberation. However, the computational cost is highly dependent on the spatial sampling and the number of microphones, making very difficult the localization of sound events if a coarse spatial sampling is used. In this paper, we propose an improvement of this method where sound events are not missed even if a very coarse grid is used. The improvement is based on a previous calculation of the existing cross-correlation lags between spatially adjacent points in the grid, assuming that the non-sampled space is covered in terms of cross-correlation lags between microphone pairs when running the algorithm. Several experiments conducted in different rooms with complex acoustics confirm the validity and benefits of the proposed method.

Contributed Papers

2:45

2pSP6. Feasibility study of underwater acoustic communications between buried and bottom-mounted sensors. Geoffrey F. Edelmann (U.S. Naval Res. Lab., 4555 Overlook Ave. SW, Code 7145, Washington, DC 20375, geoffrey.edelmann@nrl.navy.mil) and Alenka G. Zajić (School of Comput. Sci., Georgia Tech, Atlanta, GA 30332)

This paper presents a feasibility study of underwater communications between buried sensor network nodes. To investigate this problem, two experiments have been performed, where some or all sensor network nodes are buried in the sediment. The orthogonal frequency division multiplexing (OFDM) communications have been chosen to test underwater communications because of its unique strength in handling transmissions over long dispersive channels. It is demonstrated that error-free performance can be achieved between buried sensor nodes using the proposed OFDM receiver. [Work supported by the Office of Naval Research.]

3:00


The shallow water acoustic channel is challenging to estimate and track due to the ill-conditioned nature of the problem, the need to optimize over a complex field, and the time-varying nature of the channel coefficients. We have previously presented a geometric mixed norm approach to estimate and track the channel delays and delay-Doppler spread function coefficients that exploit the often sparse distribution of the channel coefficients. In this work, we present the effectiveness of our approach over a range of field data collected at 15 m depth over ranges of 60, 200, and 1000 m at various wind conditions.
3:45

2pSP8. Underwater communication system. Carlos Barroeta, Floriberto R. Ortiz, and Juan Francisco Novoa Colin (ESIME, IPN, cbarroet@hotmail.com)

When a pressure wave hits the sea bottom, or another object, reflected signal is transmitted back and is detected by the system. Transmission underwater is limited by frequency and other causes. Information lost can be measured in parameters like range, accuracy, and reliability. The sound radiated from a source is propagated in the water. The wave propagates in different directions and the sound intensity gets lower when the distance becomes larger. Water absorbs part of the energy from the wave due to absorption coefficient, and it is also affected by the distance. The wave has an axis beam with a high directivity index. A communication system based on this principle is presented.

4:00


The current state-of-the-art in high-rate acoustic communications is represented by adaptive multi-channel equalization of single-carrier wideband signals. An alternative technique with significant potential for achieving high bit rates over multipath-distorted (frequency selective) channels is multi-carrier modulation (MCM). There are obvious technical advantages of MCM. Each frequency channel of MCM can be transmitted using a relatively narrow-band, high-efficiency transducer, and low-power amplifier, which allows achieving a very broad bandwidth. Use of narrow-band channels increases the tolerance for time synchronization errors in multipath array processing. Each frequency channel has a relatively low delay spread when measured in symbol units and can be equalized with low-complexity algorithms and simplified array processing. The reduction of the ISI span in each sub-channel allows for the application of fast decision decoders optimal for joint channel estimation and data recovery such as maximum-likelihood sequence detectors (MLSDs) and adaptive BCJR MAP algorithm. The adaptive form of the MAP algorithm is considered. The combination of channel shortening equalization and a practical MLSD algorithm with dual loop feedback and pre-survivor processing were tested in shallow water acoustic communications experiments and shown a very good performance. [Work supported by ONR Grant N00014-07-1-0229.]

4:15

2pSP10. Development of a measurement system for obtaining head relative transfer function of a head and torso simulator. Agustín Vistosi, Nilda Vecliatti, and Daniel Gavininovich (Acoust. and Electroacoustic Lab., Eng. Faculty Univ. of Buenos Aires, Paseo Colón 850, 1063 Buenos Aires, Argentina, laceac@fi.uba.ar)

The human being, because of his complex audition system, has the ability to identify the spatial position of any sound from the surrounding environment. The objective of this work is to develop a measurement system that enables to mathematically model the hearing system by using a set of transfer functions. These equations, called head related transfer function (HRTF), synthesize mathematically the binaural transformation that the listener produces (principally from the torso, the head, and the pinna) in the audition process. To find the HRTF, the measurement system uses an acoustic dummy called head and torso simulator. With two microphones located in their ears, this dummy captures the sound with the same binaural properties as a real listener. The mathematical process to obtain these transfer functions is based on the LTI systems theory, using as excitation signal the sinesweep. Matlab was chosen for the processing environment, taking advantage of its flexibility on the digital sign process and their powerful computational tools.

4:30


Methods for capturing arbitrary 3-D sound fields using spherical microphone arrays have advanced to the point where commercially available arrays with performance up to fourth order are available. However, the reproduction of these captured fields for human listeners is still held back by issues such as the presence of scatterers (humans, furniture, etc.) within a loudspeaker array which distort the reproduced field, the fact that the field is only captured accurately up to an aliasing frequency of around 1–3 kHz, and technical issues dealing with sampling the sphere and constructing the array properly. This work looks at the reproduction of the acoustic field as a set of beamforming optimization problems and proposes various constraints to address these issues. The proposed subband design leverages the advantages of mode-matching at low frequencies with other optimization methods at mid and high frequencies that take into account the presence of scatterers and perceptual cues. Comparisons with VBA, conventional Ambisonics, and dual-band decoding based on Gerzon’s localization theory are also presented.

4:45


Due to the characteristics of acoustic intensity beamforming, it is desirable to reduce the noise level of the measured spectrum. In this paper, eigenvector noise suppression is adaptively applied to a vertical line array of underwater acoustic vector sensors. Eigen decomposition is applied to the cross-spectral density matrix (CSDM) of the beamformed complex spectrum from an array of Wilcoxon VS-206 vector sensors. The array elements are integrated sensors with a triaxial accelerometer and an omni hydrophone. The beamformer steering vector is applied to each of the eigenvectors for a range of steering directions within the noise field of interest. Using a known target bearing, the signal-to-noise ratio (SNR) is computed for the beamformed eigenvectors as the frequency averaged ratio between the target steering direction and all other steering directions. The eigenvectors with an SNR below a certain level are determined to be dominated by interference. These eigenvectors are subtracted from the CSDM using a projection matrix to suppress the interference. The results using calibrated source level data are presented. [Work supported through ILIR grant from Carderock Division, NSWC.]
Session 2pUW

Underwater Acoustics and Acoustical Oceanography: Compressional and Shear Wave Dispersion in Unconsolidated Media II

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Invited Papers

1:00


A number of researchers have considered what roles rotational and non-orthogonal grain motion play in the elastic response of a granular medium. Recently, Chotiros and Isakson [J. Acoust. Soc. Am. 121, EL70–76 (2007)] proposed that these motions lead to an increase in the inertia, producing a decrease in the sound speed of the elastic waves. They model this increase in inertia as an acoustic virtual mass by considering the coupling of the translational and angular motion at the granular level. Closer examination of the theory for the acoustic virtual mass reveals that it fails to take the macroscopic strain in the medium into account and hence does not actually capture the physics described by the authors. If one argues that the theory can be treated as an approximation for the grain response in the long wavelength limit, an inconsistency in the mathematical development of the model brings the final result into question. This inconsistency can be overcome within the framework of the model, and the new result introduces a frequency dependence to the virtual mass term. [Work supported by the Office of Naval Research.]

1:20

2pUW2. Newton’s third law and seismic boundary conditions in poroelasticity. Pratap Sahay (Dept. of Seismology, Centro de Investigacion Cientifica y de Educacion Superior de Ensenada, Ensenada, Baja California 22860, Mexico)

It is known that the application of Newton’s third law of motion on porous-porous welded contact implies that the total force exerted on the interface by one side is balanced by an equal and opposite force exerted on the interface by the other side. Here it is shown that Newton’s third law of motion also amounts to the continuity of the solid and the fluid phasic forces of the two sides at the interface as well as the equalization of the solid and fluid phasic forces on each side. That is, at the interface all forces are equal to one another, and that holds true irrespective of the alignment of the pores at the interface. That, in turn, also suggests that the velocity field associated with the total mass of poro-continuum is continuous at the interface. In conjunction with the requirement that the total energy must be conserved these yield the continuity of a weighted difference of solid and fluid velocities at the interface. Here, the weight factor is the product of the reciprocal density of the poro-continuum and the difference of the reciprocal densities of the two phases. The boundary conditions thus constructed uphold Newton’s third law of motion.

1:40

2pUW3. Wave propagation prediction with elastic boundary conditions. Cathy Ann Clark (Code 1513, NUWC-DIVNPT, 1176 Howell St., Newport, RI 02841, cathy.clark@navy.mil)

A bottom model which includes compressional and shear wave transmissions and reflections through a sediment layer is utilized to derive a single bottom reflection coefficient which is shown to successfully reproduce resonance effects due to shear wave conversion in various types of sediments. The consolidation of an infinite number of reflections and transmissions into a single coefficient is accomplished by formulating an infinite sum of matrices and expressing the result as a convergent series. The resultant approach achieves significant computational efficiency in comparison to performing an iterative search for mode eigenvalues. Comparisons of boundary calculations to other published results are presented and propagation predictions obtained by implementing the method in a normal mode model are compared to a number of measured data sets. [Work supported by the Office of Naval Research under the In-House Laboratory Independent Research (ILIR) Program.]

2:00

2pUW4. The effect of shear wave dispersion on bottom loss from unconsolidated sediments with rough interfaces. Marcia J. Isakson and Nicholas P. Chotiros (Appl. Res. Labs., The Univ. of Texas, 10000 Burnet Rd., Austin, TX 78713)

Shear waves are often neglected for unconsolidated ocean bottom sediments like sands; however, the shear mode in these sediments can influence propagation and bottom loss especially for relatively high shear wave speeds and rough sediment/water interfaces. Although measured shear wave speeds are low, these measurements were made at frequencies much lower than the operating frequencies of most imaging or bathymetric sonars. New models for unconsolidated sediments suggest that shear wave speed may have a steep
dispersion implying significant shear wave speeds at operating sonar frequencies. [Chotiros and Isakson, J. Acoust. Soc. Am., 116, 2011–2022 (2004).] Additionally, measurements for bottom loss suggest a dependence of shear speed on frequency. [Isakson, Yarbrough, Chotiros, and Piper, J. Acoust. Soc. Am., In review.] Lastly, although there is often low coupling between waves of vastly different sound speeds for shallow grazing angles on flat interfaces such as in long range propagation, interface roughness can steepen the local grazing angle leading to increased coupling. In this study, the effects of predicted shear wave dispersion and coupling through rough interface scattering on bottom loss and propagation for shallow water waveguides with unconsolidated sediment bottoms will be explored using finite element modeling. [Work sponsored by Office of Naval Research, Ocean Acoustics.]

2:20


The author proposed a gap stiffness model incorporated into the Biot model; that is, the BIMGS model. Reported large velocity dispersion is predicted by this model. However, at higher frequencies, there are some reported data for negative velocity dispersion and attenuation proportional to the first to fourth power of frequency. In this study, the velocities and attenuation in six kinds of water-saturated glass beads and four kinds of water-saturated silica sands with different grain sizes are measured in the frequency range of 80–140 kHz and 300–700 kHz. The measured results are compared with the calculated results by using the BIMGS model plus some acoustic sediment models, such as the BICSQS model plus high-frequency correction, nonlocal model, porosity dynamics model, and multiple scattering model. It is shown that the velocity dispersion and attenuation is predicted by using the BIMGS model in the range of $k d \ll 0.5$ ($k$ is wavenumber in water, $d$ is grain diameter) and by using the BIMGS model plus multiple scattering effects in the range of $k d > 0.5$, in which a negative velocity dispersion appears. Finally, the relations between the measured velocity dispersion and attenuation are confirmed by using the twice-subtracted Kramers–Kronig relations.

2:40


Intrinsic attenuation is ideally measured for plane compressional waves in an unbounded homogeneous medium. For typical marine sediments, at frequencies below 5000 Hz, sufficient realization of this idealization cannot be achieved. Reported indirect measurements typically have source and receiver in the water column and require hypothesized theories and previously measured parameters. An extended theory-based discussion is given of various experiments with substantially different designs that have led to widely different results. Relevant comparisons are the attenuation at 1 kHz and the general dependence on frequency over the range 100–1000 Hz. Data from a single experiment are typically amenable via approximate graphing as being proportional to frequency raised to some power $n$, usually between 1 and 2. Some theoretical notions say that $n$ should be exactly 2, others that it should be exactly 1 or perhaps only slightly greater than 1. It is pointed out that most experimental designs involve questionable assumptions. If the true exponent is 2, then the attenuation at very low frequencies is going to be especially difficult to infer. Other physical mechanisms can enter into an indirect inference of attenuation, and mechanisms that would lead to a perceived linear frequency dependence seem to be ubiquitous.

3:00—3:15 Break

Contributed Papers

3:15


Comparative sediment sound-speed and attenuation measurements were conducted in the northern Gulf of Mexico using three different methods. The first method uses ship noise generated by the attending research vessel, recorded by a hydrophone near the seafloor. The second method uses a towed chirp sonar and estimates sound-speed and attenuation from the record by a hydrophone near the seafloor. The second method uses ship noise generated by the attending research vessel, re-conducted in the northern Gulf of Mexico using three different methods. The third method uses four acoustic probes and measures sound-speed and attenuation at 0.6 m below seafloor. Auto-correlation processing of ship-noise data provided subbottom profiling up to 40 m penetration and the results were in agreement with those of ship sonar data. The measured sound speed and attenuation are typical to clayey and silt/ clayey sediments that agree with the available core data. The frequency dependence of sound speed and attenuation were also estimated within a wide frequency band using the data from co-located ship noise (0.2–5 kHz), chirp-sonar (2–12 kHz), and sediment acoustic-probe (5–120 kHz) measurements. Observed small frequency dependence of sound-speed and linear frequency dependence of attenuation are in agreement with those predicted by the Biot–Stoll model for silt/ clayey sediments. [Work supported by ONR.]

3:30

2p UW8. The cardhouse theory of mud: Determination of shear wave speed. Joseph O. Fayton (Rensselaer Poly. Inst., 110 8th St., Troy, NY 12180, faytoj@rpi.edu), Allan D. Pierce, William M. Carey (Boston Univ., Boston, MA 02215), and William L. Siegmann (Rensselaer Poly. Inst., Troy, NY 12180)

Platelets in high porosity ocean sediments carry negative charge due to isomorphous substitution. When immersed in an ion dense environment such as sea water, positive charges congregate near each surface. The high porosity combined with the charge configurations suggests a cardhouse structure where platelets repel face to face and attract face to end. Understanding the nature of interactions between platelet faces and edges is important for determining how the cardhouse responds to small amplitude acoustic disturbances. To quantify such interactions, a sequence of fundamental problems with electric charge distributions near and around charged platelets is considered, following general procedures of Verwey and Overbeek (1948). Analytical and computational results support a hinged joint model, for which the amount of energy required to separate perpendicular platelets is large compared to the energy required to rotate an edge about the contact line with another platelet. The implication is that mud will weakly resist shear, because the work done to achieve shear distortions results in an
increase of electrostatic energy within the cardhouse. From the above considerations estimates of the shear bulk modulus and the shear wave speed are made. [Work partially supported by the ONR.]

3:45

2pUW9. Sound speed dispersion measurements and their interpretation in the presence of a shallow buried layer. John C. Osler and Paul C. Hines (DRDC Atlantic, P.O. Box 1012, Dartmouth, NS B2Y 3Z7, Canada, john.osler@drdc-rddc.gc.ca)

Sharing the same experimental apparatus, two complementary techniques were developed to measure the frequency-dependent speed of sound in marine sediments during the SAX04 sea-trial. The first technique enabled direct time-of-flight measurements of acoustic wave speed along all three Cartesian axes. The second technique determined the acoustic wave speed based on the arrival angle of pulses generated in the water column and refracted upon entry into the seabed. None of the results could be modeled or explained when the seabed was parametrized as a sand half-space. However, both techniques suggested the presence of a thin muddy layer, 0.05–0.2 m thick within the top 1 m of the sediment. For the time-of-flight technique, in the vertical direction, the interference from the layer created considerable apparent variability in sound speed measurements. In the horizontal direction, the layer caused the measured sound speeds to be lower than what a simplified poro-elastic model would predict, unless one accounts for the higher porosity material in the layer. For the arrival angle technique, the layer explains the complicated frequency- and geometry-dependent results; however, the sound speed dispersion was a second order effect that could not be unambiguously extracted.

4:00

2pUW10. Studies on the effect of shear on compressional wave dispersion. Gopu R. Potty and James H. Miller (Dept. of Ocean Eng., Univ. of Rhode Island, Narragansett, RI 02882)

The results of a study to examine the effect of shear on compressional wave dispersion are presented. Modal arrival time data from different shallow water sites will be analyzed to examine the influence of shear. The modal travel times corresponding to the Airy Phase regions are extremely sensitive to shear. Simple inversion schemes will be developed to estimate the shear speed in the sediment by comparing theoretical predictions with experimental data. The estimated shear speeds will also be compared with shear speeds calculated from core data. The effect of shear on compressional wave attenuation will also be investigated. Synthetic data will be generated for elastic bottom, with different shear speeds, and these data will be inverted for compressional wave attenuation. One of the most promising approaches to estimate shear speed is to invert the relation between seismo-acoustic interface waves (Scholte waves) that travel along boundaries between media and shear wave speed. The propagation speed and attenuation of the Scholte wave are closely related to shear-wave speed and attenuation over a depth of 1-2 wavelengths into the seabed. A shear measurement system being developed at the University of Rhode Island based on this concept will be presented. [Work supported by the Office of Naval Research.]

4:15

2pUW11. Shear wave attenuation in underwater granular media. Nicholas P. Chotiros and Marcia J. Isakson (Appl. Res. Labs., Univ. of Texas at Austin, P.O. Box 8029, TX 78713-8029, chotiros@arlut.utexas.edu)

Compared to compressional waves, there are relatively few measurements of shear waves in underwater sediments. In sandy sediments, shear waves are known to be slow and lossy compared to compressional waves. They contribute to the reflection loss at the water-sediment interface and they are known to determine the optimum propagation frequency in a shallow water wave guide. The frequency dependence of shear attenuation is known to be proportional to frequency, while that of the compressional wave is proportional to the square of frequency. This difference suggests that they are subject to different loss mechanisms. The linear frequency dependence is likely associated with solid contact losses, while the squared frequency dependence is indicative of viscous losses associated with the pore fluid. A broad band model of shear wave propagation should include both loss mechanisms. The frequency at which one mechanism gives way to the other may be estimated from existing models and measurements. [Work supported by the Office of Naval Research, Ocean Acoustics Program.]