Plenary Session and Awards Ceremony

Presentation of Certificates and Awards by Acoustical Society of America

George V. Frisk, President
Acoustical Society of America

Presentation of Certificates to New Fellows

William E. Brownell
H. Timothy Bunnell
Emily Buss
Cathy Ann Clark
R. Lee Culver
Bertrand Delgutte
Timothy F. Duda
Donald K. Eddington
Toshiro Irino
James J. Jenkins
Bruce K. Newhall
Charlotte M. Reed
Mark E. Schafer
Noral D. Stewart
Mario Zampolli

Presentation of Awards

Rossing Prize in Acoustics Education to Jerry H. Ginsberg

Silver Medal in Signal Processing in Acoustics to Edmund J. Sullivan

Silver Medal in Speech Communication to David B. Pisoni

Pioneers of Underwater Acoustics Medal to George V. Frisk

Presentation of Awards by Mexican Institute of Acoustics

Sergio Beristain, President
Mexican Institute of Acoustics

John William Strutt, 3rd Baron of Rayleigh Medal to Thomas D. Rossing

Herman Ludwig Ferdinand von Helmholtz Medal to Octavio Razcón

Session 4aAA

Architectural Acoustics: Auralization, Modeling, and Computational Methods

Norman H. Philipp, Chair
School of Architecture, Univ. of Kansas, 1465 Jayhawk Blvd., Lawrence, KS 66045

Chair's Introduction—8:45

Invited Papers

8:50

4aAA1. Perceptions of auralization in the design process. Norman H. Philipp (School of Architecture, Univ. of Kansas, 1465 Jayhawk Blvd., Lawrence, KS 66045, nphilipp@ku.edu)

The application of auralization as a part of the architectural design process first requires an understanding of the current perceptions of auralizations by the design community. From this inquiry on a national scope, the level of education necessary to allow architects to understand auralizations and embrace them into their design processes may be determined. This paper summarizes the preliminary phases in the development of a national survey toward the aforementioned goals.

9:10

4aAA2. Investigation of the subjective impression of listener envelopment with both binaural recordings and auralizations. Michelle C. Vigeant, Robert D. Celmer, Madison D. Ford, and Carl K. Vogel (Acoust. Prog. and Lab., Mech. Eng. Dept., Univ. of Hartford, 200 Bloomfield Ave., West Hartford, CT 06117, vigeant@hartford.edu)

Listener envelopment (LEV) is the sense of being fully immersed in a sound field and can be used to compare the listening experience in different concert halls. LEV has been shown to correlate with the objective parameter late lateral sound level (GLL) through the use of simulated sound fields generated with delays and reverberators. The primary purpose of this study was to investigate this correlation using both binaural recordings made in a 900-seat hall and auralizations made in an ODEON v9.20 model with both measured and predicted GLL values. In addition, the ratings of the actual recordings and simulations were compared to determine equivalency. A subjective study was carried out using 35 musically trained test participants who rated 24 stimuli, which varied as a function of both receiver position and hall setting. The ratings of the binaural recordings were found to have a linear correlation with both the measured and simulated GLL values, while the ratings of the auralizations were not found to have a clear linear relationship with GLL. When the ratings of the recordings and auralizations were compared, however, only two cases were found to be significantly different. [Work supported by a University of Hartford Greenberg Junior Faculty Grant.]

Contributed Papers

9:30

4aAA3. Combining auditory and tactile inputs to create a sense of auditory space. Ross W. Deas, Robert B. Adamson, Philip P. Garland, Manohar Bance, and Jeremy A Brown (SENSE Lab., 1276 South Park St., Rm. 3189, Dickson Bldg., VG Site QEII Health Sci. Ctr., Halifax, NS B3H 2Y9, Canada)

To localize a sound, the auditory system uses multiple cues, including binaural differences in timing and level that arise from the separation of the ears by the solid mass of the head. It has repeatedly been shown that the ability to utilize these cues is plastic and experience-based. Vibrotactile input shares many common features with auditory signals, and there is some overlap between the frequency range of the sensitivity of the ear and skin. In this study, we examine whether the auditory system is capable of combining auditory and tactile inputs to localize sounds using a multi-speaker array. To induce deficits in azimuthal localization, one ear was plugged. To examine cross-modal localization, the input level to the plugged ear was recorded via microphone, and a vibratory signal that was perceptually equal in intensity was presented to the shoulder on the same side as the plugged ear. The participant's ability to localize low-pass, band-pass, high-pass, and broadband sounds was measured. Results showed that relative to baseline (plugged) conditions, localization performance improved, suggesting that listeners can combine auditory and tactile information to create a sense of auditory space.

9:45

4aAA4. Real-time interactive room acoustic simulation engine using graphics processors. Zuofu Cheng and Lippold Haken (Dept. of Elec. and Comput. Eng., Univ. of Illinois at Urbana-Champaign, 1406 W. Green St., Urbana, IL 61801-2918, zcheng1@illinois.edu)

Presented is a software system for real-time simulation of acoustic spaces on consumer graphics technology. This is an API that uses modern graphics processing units (GPUs) to derive the high-order reflected and transmitted paths to a stereo listener from multiple sound sources within a room of arbitrary geometry and wall properties. The system processes the entire audio chain on the GPU, computing the listener's stereo mix from raw input waveforms stored in video memory and a polygonal room model and material and scene description. For the intended application which is video game audio, the room acoustics are computed at 15–30 frames/s and interpolated to account for fast listener or source motion to coincide with the graphics rendering, which may be done concurrently on the same processor. Provisions are provided within the API for dynamic rooms to handle gameplay elements such as doors, platforms, and physically modeled objects. An

acoustic scene editor is presented, which also uses the API to simulate a first person game environment.

10:00—10:15 Break

10:15

4aAA5. Correct and rapid impulse response calculation as a precondition for auralization. Wolfgang Ahnert, Stefan Feistel, and Alexandru Miron (AFMG Technologies GmbH, Arkonastr. 45-49, D-13189 Berlin, Germany, wahnert@ada-amc.eu)

For almost 20 years auralization procedures have become more and more common, but the simulated acquisition of needed impulse responses (IRs) is still a crucial point during a design process. Different methods have been developed over time with different assumptions and approximations. Until now we observe time-consuming procedures to calculate true IRs especially in large halls. In this paper new algorithms are reported to calculate IRs in multi-thread mode with one computer or better via a network with a set of CPUs working in parallel. So calculations in large halls may be reduced from 8–10 h with one CPU to 10–20 min with multi-core computers. Via internet routines a master-slave network is used to allow parallel networking on the same file. Examples are shown using eight remote-controlled quad-core computers, i.e., 32 CPUs at the same time. This way the auralization procedure is becoming a daily and quickly usable function. It opens the way to convert an auralization toy into a real tool.

10:30

4aAA6. Acoustic analysis and visualization using real time three-dimensional parametric modeling. Dave Rife (Arup Acoust., 155 Ave. of the Americas, New York, NY 10013)

As we move further toward the 3-D world in building design, specifically for non-standard shaped performance spaces, it is becoming more and more useful to easily transition between 3-D models and simple acoustic calculations. Using RHINOCEROS, a widely used and flexible 3-D modeling program, and GRASSHOPPER, a parametric modeling plugin, tools have been developed to streamline simple acoustic analysis and visualization directly from an architectural model. These real time parametric tools help facilitate a clearer understanding of room-specific acoustic phenomenon to the designer, and have proven to be particularly useful in efficiently communicating acoustic principles to non-acousticians. Examples are shown, including visualizing reflection sequences, time delays, distance attenuation, and reverberation time calculations, which are all parametrically driven and updated in real time within the 3-D modeling software.

10:45

4aAA7. Parameter-based mathematical optimization using MATLAB and CATT-A software. Dave Rife (Arup Acoust., 155 Ave. of the Americas, New York, NY 10013)

Mathematical optimization techniques have been applied to room acoustic modeling within CATT-A. Stochastic optimization methods such as genetic algorithms, pattern search, and ant colony optimization are used via software integration between MATLAB and CATT-A, allowing for acoustic parameter specific optimization of variable room geometry. The optimization routines rely on results from geometrical ray tracing, which informs the designer on aspects of room acoustics that empirical formulas cannot take into account. An example is presented showing an algorithm that optimizes placement of sound absorbing treatment to obtain a fixed speech transmission index using minimal sound absorbing material in a classroom setting.

11:00

4aAA8. Sound energy decay analysis in multiple coupled volume systems. Ning Xiang, Philip Robinson, and Yun Jing (Grad. Prog. in Architectural Acoust., Rensselaer Polytechnic Inst., 110 8th St., Troy, NY 12180)

There has been an increasing need in analyzing sound energy decays consisting of more than one decay slope, so-called non-single-exponential decays. It has been considered very challenging to estimate parameters associated with double-slope decay characteristics, even more challenging when the coupled-volume systems contain more than two decay processes. To meet the need of characterizing energy decays of multiple decay processes, this work reports investigations using both acoustical scale-models and numerical models of three coupled volumes. Two levels of Bayesian inference: model selection and parameter estimation, have been applied to detection and characterization of sound energy decays. Experimental results measured in acoustical scale models and numerical results from diffusion-equation-based models are used to validate the Bayesian analysis methods. The analysis methods are then applied to geometric-acoustics simulations of a conceptual concert hall. This work demonstrates that the analysis method within the Bayesian framework is capable of determining more than two decay slopes and estimating the corresponding decay parameters and parameter uncertainties.

11:15

4aAA9. A simplified two-dimensional model of acoustic diffusion. Martín Sequeira and Víctor Cortínez (CIMTA, Univ. Tecn. Nac., Bahia Blanca, 8000 Buenos Aires, Argentina, martins@frbb.utn.edu.ar)

The acoustic control is a topic of growing interest in workspaces. For the purpose of designing acoustic solutions, it is necessary to make correct predictions of sound levels in order to advise the most appropriate acoustic treatments. There are many methodologies to predict the noise levels in industrial rooms, from the simplest empirical models based on diffuse sound fields assumption or experimental data to a more robust method such as the ray tracing technique. A new concept for the acoustic enclosure prediction was recently proposed. [Picaut (1997).] This model may be considered as an extension of the classical theory of reverberation and has been applied to a lot of situations with good agreement and low calculation time. The objective of this work is to provide an equivalent 2-D diffusion model achieving similar prediction results in terms of sound pressure level distributions with a lower calculation time.

11:30

4aAA10. Modal reverberation time in acoustic systems with absorptive walls. Helvio Mollinedo, Hector Merodio, Itzala Rabadan, and Jose Angel Ortega (SEPI-ESIME-IPN Zac., Depto de Ing. Mecanica, UPALM, Ed. No 5-3er Piso, Del. Gustavo A. Madero, Mexico DF, C.P. 07738, Mexico helviomollinedo@yahoo.com)

The early works about the reverberation of room acoustics start with geometrical tecniques. Subsequent works in this field showed that the reverberation has characteristic frequencies of the normal vibration modes of the room, and not necessarily the frequency of the source that starts the vibration; the acoustic response of the room can be understood in terms of normal modes and the decay coefficient associated with each one of those modes. This work presents an analysis of a known procedure to compute the decay coefficient and the modal reverberation time of interior acoustic rooms with lightly damped absorptive walls by means of the finite element method. This procedure was performed for several wall-lined configurations in 3-D geometries. These calculations were carried out in a computational program prepared in MATLAB. The results show how this useful technique could predict the modal response and reverberation time depending on the absorption wall configuration.

11:45

4aAA11. Modeling and shaping specular and diffuse reflection sound fields in enclosures. Krista Michalis, Donald Bliss, and Linda Franzoni (Mech. Eng., Duke Univ., Durham, NC 27708, kam49@duke.edu)

Steady-state sound fields in enclosures, with specular and diffuse reflection boundaries, are modeled with an energy-intensity boundary element method using uncorrelated broadband directional sources. The specular reflection case is solved using an iterative relaxation method extended from a 3-D diffuse reflection solution. When chosen properly, only a few directivity harmonics are required. Starting with a diffuse calculation, higher harmonics

are estimated by post-processing, and the original diffuse influence matrix is refined accordingly and converges in a few relaxation steps. The method is higher-order accurate, computationally efficient, and similar to the diffuse case in that it includes strict enforcement of energy conservation. The simulations give accurate results compared to exact modal solutions. Sample calculations are presented to show that the mean-square pressure spatial variation can be tailored using a combination of diffuse and specular surfaces for the design of acoustic spaces.

THURSDAY MORNING, 18 NOVEMBER 2010 CORAL GARDEN 1, 7:55 A.M. TO 12:00 NOON

Session 4aAB

Animal Bioacoustics: Tropical Animal Bioacoustics

Edmund R. Gerstein, Chair
Leviathan Legacy Inc., 1318 SW 14th St., Boca Raton, FL 33486

Chair's Introduction—7:55

Invited Papers

8:00

4aAB1. Case studies of bioacoustics research in Brazil. R. S. Sousa-Lima (Bioacoustics Res. Program, Cornell Lab. of Ornithology; Inst. Baleia Jubarte, Caravelas, BA, Brazil; Inst. de Ciências Biológicas, Univ. Federal de Minas Gerais, Belo Horizonte, Brazil, RSL32 @cornell.edu)

Brazilian researchers have a growing interest in animal sounds as more and more exposure is given to this subject in the worldwide literature and concerns about noise pollution are on the rise. Some case studies focusing on research that I have been involved in, i.e., studying manatees (*Trichechus inunguis* and *T. manatus*), humpback whales (*Megaptera novaeangliae*), manned wolves (*Chrysocyon brachyurus*), primates (*Callitrix penicillata*), and fish (*Prochilodus* sp. and *Leporinus* sp.), will be presented. Studies of the acoustic ecology of these species have focused on determining the importance of acoustic communication, vocal identity, and differences in the acoustic behavior of groups of animals to better understand animal populations in the wild and in urban areas. Such studies help to support overall conservation efforts and highlight the importance of the potentially negative effects of masking on animal communication systems in the tropics.

8:20

4aAB2. A comparison of acoustic soundscapes within and among three tropical habitats: Can soundscape heterogeneity be used as an index of alpha diversity? L. J. May-Collado, T. M. Aide (Dept. of Biology, Univ. of Puerto Rico, P.O. Box 23360, San Juan, PR 00931, lmaycollado@gmail.com), C. Corrada, and R. Alvarez (Univ. of Puerto Rico, San Juan, PR 00931)

Rapid biodiversity assessment methods are needed to ensure that adequate policies and management strategies are established to protect biodiversity. In this study, we evaluate the reliability of a recently developed acoustic index for alpha diversity by determining the within and among site variations from three permanent automated recording stations located in Sabana Seca (wetland) and El Verde (moist forest), Puerto Rico and in La Selva (wet forest), Costa Rica. All stations sampled the soundscape for 1-min every 10-min, 24 h per day throughout the year. For this analysis, we evaluated recordings from the dry season and wet season for each site. The analyzes were carried out on each individual recording and then mean values were calculated for each month. Independent of the season or time, La Selva had the highest acoustic diversity and El Verde had the lowest. Both Sabana Seca and El Verde showed a significant increase in acoustic richness after dusk, reflecting the peak of amphibian activity. This acoustical index is a useful tool for identifying important areas of biodiversity without relying on individual species identification. Our results indicate that soundscape heterogeneity is a good indicator of alpha diversity.

Contributed Papers

8:40

4aAB3. The role of the environment on the acoustic radiation patterns of mating calls of the túngara frog. Ximena E. Bernal (Dept. of Bio. Sci., Texas Tech Univ., Lubbock, TX 79409-3131), Rachel A. Page (Smithsonian Tropical Res. Inst., MRC 0580-06, Apartado 0843-03092 Panamá, República de Panamá), Michael J. Ryan (Section of Integrative Biology, Univ. of Texas at Austin, Austin, TX 78712), Theodore F. Argo, IV, and Preston S. Wilson (Mech. Eng. Dept. and Appl. Res. Labs., Univ. of Texas at Austin, Austin, TX 78713-8029)

The present work is a follow-up to previously reported [J. Acoust. Soc. Am. **122**, 2981(A) (2007)] laboratory measurements of both the horizontal and vertical frequency-dependent directivity of the mating calls of túngara frogs, *Physalaemus pustulosus*. Band-limited directivities are significantly greater than broadband directivities, with a maximum directivity of 20 dB in the vertical plane for harmonics near 6 kHz. This result is unexpected given that female frogs, the intended receivers of this call, are on the horizontal plane rather than on the vertical plane where unintended receivers (predators and parasites) reside. New numerical finite element modeling of the radiation, including the effects of the environment, will be reported. Túngara frogs only call while partially submerged in water. Modeling results indicate that the reflecting boundary provided by a finite-sized air-water interface dominates the radiation, and results in the observed directivity. There is comparatively little directivity dependence on the nature of the surrounding terrain, be it acoustically hard or soft, smooth or rough. These modeling results indicate that the radiation patterns found in laboratory measurements, although unexpected, are likely to be found in nature too.

8:55

4aAB4. Noisy neighbors: A role for masking interference in the evolution of communication systems in neotropical dendrobatid frogs. Adolfo Amézquita (Departamento de Ciencias Biológicas, Universidad de los Andes, Bogotá, Colombia), Albertina Pimentel Lima (Ecologia, Instituto Nacional de Pesquisas de Amazonia, Manaus, Brazil), and Walter Hdl (Dept. of Evolutionary Biology, Inst. of Zoology, Univ. of Vienna, 1010 Vienna, Austria)

Signal detection and signal recognition are necessary for effective communication and, thus, vulnerable to the selective effects of interference by heterospecific signals. Previously, we showed correlative evidence that the frequency response curve of territorial frogs *Allobates femoralis* is narrower at places where it co-occurs with the frog *Ameerega trivittata*, which utters a spectrally overlapping call. Here we (1) conducted manipulative experiments to test the underlying assumption that *A. trivittata* calls affect the behavioral and reproductive performance of unexperienced *A. femoralis* males. The addition of *A. trivittata* calls led to larger territories and lower opportunities for mating and breeding in *A. femoralis* males. Thus, heterospecific masking reduces breeding opportunities through its effects on territorial but not calling behavior. We also (2) tested at a species-rich acoustic community whether the recognition space of five frog species was shaped in a way that minimizes overlap with the heterospecific signal spaces. The recognition space was modeled from the all-or-none phonotactic reactions of territorial males. Our data supported the initial hypothesis for three out of five species. Both data sets suggest that masking interference may shape the receiver portion of the communication systems without concomitant changes in the call.

9:10

4aAB5. Monitoring frog species recovery in secondary tropical forests using automated species identification. B. Hilje, T. M. Aide (Dept. of Biology, Univ. of Puerto Rico, P.O. Box 23360, San Juan, PR 00931-3360, bhilje@gmail.com), and C. Corrada-Bravo (Univ. of Puerto Rico, San Juan, PR 00936-8377)

Although the most important causes of the current amphibian decline and species extinction are habitat degradation and habitat loss, in some regions, forests are recovering where agriculture has been abandoned. After 30–40 years of recovery ,vegetation characteristics are often similar to those of mature forests, but little is known about the recovery of the fauna. In this study we evaluated the recovery of amphibian species in a chronosequence of secondary forests (10–30 years old) and mature forest. Each site was sampled in March, April, and May 2010 for 10 days using automated recorders which produce 2000 1 min recordings per site. Species specific algorithms were developed for 20 species of frogs and an automated species identification software was used to process the recordings. A total of 11 species were identified, and most were leaf-litter frogs. Species *Oophaga pumilio* and *Diasporus diastema* were identified in all sites. The highest species richness was observed in the 22–30-year-old forest, where there was a mix of species from both young and mature forests. These results suggest that similar to the vegetation, the majority of the original frog community has been able to recolonize these secondary forests.

Invited Papers

9:25

4aAB6. Flexibility in the advertisement call of *Hypsiboas pulchellus* (Anura: Hylidae) in response to microhabitat characteristics. Lucia Ziegler, Matías Arim (Secc. Zoologá Vertebrados, Facultad de Ciencias, Univ. de la República, Iguá 4225, Montevideo 11400, Uruguay, luciaz@fcien.edu.uy), and Peter M. Narins (Univ. of California Los Angeles, Los Angeles, CA 90095)

The structure of the environment determines patterns of signal degradation and attenuation, potentially affecting communication. The expected adjustment in call structure improving signal transmission in an environment was formalized in the acoustic adaptation hypothesis. Within this framework, most studies considered anuran calls as fixed attributes determined by local adaptations. However, intra-individual variability in vocalizations has also been reported. Evidence is presented here for phenotypic flexibility in the advertisement call of *Hypsiboas pulchellus* in response to induced changes in microhabitat structure. Males were recorded in free-field conditions. Microhabitat was then modified, placing a styrofoam enclosure around each calling male, re-recording calls therein. Student tests for independent samples (intra-individual variation) and paired samples (inter-individual variation) rendered similar results. All temporal variables were significantly longer when individuals were broadcasting inside the enclosure. Spectral attributes were also subject to modulation, although the magnitude and direction of these adjustments were inconsistent among males. Playback experiments within the enclosure yielded results which rule out treatment-induced variation, highlighting the potential rôle of call flexibility on detected call patterns. This study questions the view of fixed adaptations as the sole determinant of the match between call and environment, positing phenotypic flexibility as a main determinant of this interplay.

9:45

4aAB7. Communication through the substrate as a solution for small plant-dwelling insects. Andrej Cokl and Meta Virant-Doberlet (Dept. of Entomology, Natl. Inst. of Biology, Vecna pot 111, SI-1000 Ljubljana, Slovenia)

High-frequency airborne sound is quickly absorbed in vegetation, and outside the plant shelter such signals attract predators. Furthermore, small insects cannot emit efficiently low-frequency airborne sound when their diameter is smaller than a third of the radiated wavelength. For plant-dwelling insects, substrate-borne sound signaling remains the best solution for communication in dense vegetation typical for the tropics. Communication range in such an environment depends on tuning of vibratory signal properties with mechanical characteristics of plants and sensory abilities to extract the information from the environmental noise. *Nezara viridula* and other until now investigated species of the stink bug family *Pentatomidae* represent the model for solutions optimizing long-range communication through green plants. They communicate with vibratory signals of the dominant frequency around 100 Hz, which travel through plants with low attenuation, creating standing wave conditions in the plant's rod-like structures. Green plants act as low-pass filter and their resonant peaks fit well with the spectral peaks of stink bug vibratory emissions. The species' leg sensory organs with the underlying neural network are sensitive enough to enable communication through a plant on a distance of several meters. Comparative data on substrate-borne sound communication in burrower (*Cydnidae*) and predatory (*Asopinae*) bugs will be discussed.

10:05—10:20 Break

10:20

4aAB8. Structure and contexts in the long distance calls of *Alouatta clamitans* and *Alouatta belzebul*. Dilmar Oliveira (Centro de Fauna Silvestre, Secretaria do Meio Ambiente, 05459-010 Sao Paulo, SP, Brazil, dilmar.ago@gmail.com) and Cesar Ades (Univ. de Sao Paulo, Sao Paulo, SP, Brazil)

The structure of the long range calls of howler monkeys, primates of the genus *Alouatta*, is scarcely understood in relation to the functions they assume. Records of *Alouatta clamitans* and *Alouatta belzebul* long distance calls were obtained at different Brazilian sites. The long distance calls of both species are low-frequency, harsh, loud sounds, uttered in sequences that may last several minutes. There is wide gradation both within and between call categories. The structure of roars and barks seems to serve different roles: barks function as graded signals of alarm, roars as resource holding potential signals used in resource defense. Observations in *A. clamitans* give support to such hypotheses. Roar use and structure have the most marked differences between these species. A dawn chorus of roaring was observed only for *A. belzebul*, which probably has a more spontaneous emission of roars, while *A. clamitans* use roars mainly in intergroup encounters. The structural divergence between the long range calls of these species may reflect species–typical adaptations. The presence of both brief and continuous roars in both species, but not in the most basal *Alouatta palliata* can bring cues to loud calling evolution in howlers. [Work supported by CNPq, FAPESP, IF/SP, FURB, CEPESBI, IBAMA/PB, and Fazenda Pacatuba.]

10:40

4aAB9. Acoustic behavior in crocodilians. Adam R. C. Britton (Big Gecko Crocodilian Res., P.O. Box 925, Sanderson, NT 0813, Australia, abritton@crocodilian.com)

Sound production in reptiles is generally limited in scope, yet crocodilians have developed a modest repertoire of vocal and non-vocal acoustic signals to convey a range of different information. This ability is perhaps less surprising considering that their Archosaur heritage is shared with dinosaurs and birds, and basic similarities to avian bioacoustics are apparent. An overview is presented of sound production and reception in crocodilians, the sound characteristics and repertoire of known signals, and the important role that these signals play in crocodilian behavior. The evolution of bioacoustics in this group is touched upon, based on biological and behavioral evidence between species.

Contributed Papers

11:00

4aAB10. Tapir vocalizations: A comparison with equines and other perissodactyls. David G. Browning (139 Old North Rd., Kingston, RI 02881) and Peter D. Herstein (24 Mohegan Rd., Charlestown, RI 02813)

The perissodactyls (odd-toed ungulates) comprised of equines, rhinos, and tapirs have vocalizations during which the frequency spectra can change, in contrast to the artiodactyls (even-toed ungulates), comprised of cows, sheep, deer, etc., whose vocalizations are primarily tonals. This suggests that a simple vocal expression by perissodactyls may be possible, and a recent study of horse whinnies under different conditions indicates that this may be the case. The vocalizations of tapirs, perhaps the least studied members of this group, are compared to those of equines and rhinos.

11:15

4aAB11. The acoustic ecology and behavior of minke whales (*Balaenoptera acutorostrata*) near tropical and subtropical North Pacific Islands. Thomas Norris (Bio-Waves Inc., 517 Cornish Dr., Encinitas, CA 92024), Tina M. Yack (Southwest Fisheries Sci. Ctr., La Jolla, CA 92037-1023), Stephen Martin (SPAWAR Systems Ctr., Pacific, San Diego, CA), Julie N. Oswald (Bio-Waves Inc., Encinitas, CA 92024), Amanda J. Cummins (Scripps Inst. of Oceanogr., La Jolla, CA 92093-0205), and Len Thomas (Univ. of St. Andrews, St. Andrews, Scotland KY16 9LZ,)

Passive acoustic monitoring, acoustic localization and acoustic/visual line-transect surveys of minke whales were conducted near the Hawaiian and Marianas Islands between 2006 and 2010. Acoustic data were collected using (1) towed hydrophone arrays deployed off Kauai and the Marianas Islands, (2) seafloor hydrophones from the U.S. Navy's Pacific Missile Range Facility (PMRF) northwest of Kauai, (3) the Aloha Cabled Observatory (ACO) seafloor hydrophone northwest of Oahu, and (4) HARP autonomous recorders deployed off the Northwest Hawaiian Island Chain. Significant differences were detected in the pulse repetition rates of boings recorded in Hawaiian versus the Marianas Islands. This information is being used to assess the population characteristics of North Pacific minke whales. Analysis of ACO recordings indicates seasonal patterns, but not diurnal patterns in the number of boings detected. We are in the process of estimating the abundance of vocalizing animals in the main Hawaiian and Northern Marianas Islands study sites using towed hydrophone array data. These results will be compared to estimates made with the PMRF hydrophone data using spatially explicit capture-recapture methods. Results of these studies are providing a better understanding the acoustic ecology and behavior of minke whales in low-latitude breeding areas of the North Pacific. [Work supported by ONR.]

11:30

4aAB12. Vocal activity of tropical dolphins is inhibited by the presence of killer whales, *Orcinus orca*. Shannon Rankin, Eric Archer (Southwest Fisheries Sci. Ctr., 3333 N. Torrey Pines Ct., La Jolla, CA 92037, shannon.rankin@noaa.gov), and Julie Oswald (Oceanwide Sci. Inst., P.O. Box 61692, Honolulu, HI 96839)

Research has suggested killer whale predation may affect cetacean vocal behavior; however, few data exist to test this hypothesis. Data collected for 19 609 km of visual and acoustic shipboard surveys in the tropical Pacific Ocean were examined to determine if changes in dolphin vocal activity could be attributed to the presence of killer whales. These surveys included 346 detections of three highly vocal dolphin species (genus *Stenella*), whose whistles can be detected at ranges over 4.6 km. Random forest analysis was used to model vocal behavior based on sea state, visibility, fog/rain, thermocline temperature/depth, mixed layer depth, chlorophyll, distance-to-shore, species, group size, perpendicular distance, and presence of killer whales. Our results show that the presence of killer whales significantly inhibited vocal activity in these tropical dolphins ($p = 0.02$). Killer whales are rare in the tropics, and this disruption in communication may not have a significant impact on interactions necessary for survival. However, in temperate climates, where increased productivity supports a greater abundance of killer whales, this interruption in communication may have a greater impact. The lower incidence of whistling dolphins in temperate waters may be related to the greater abundance of killer whales in these areas.

11:45

4aAB13. Analysis of the interaction of acoustic waves with hard and soft corals in the near field of a source. Mardi C. Hastings (Woodruff School of Mech. Eng., Georgia Inst. of Technol., Atlanta, GA 30332-0405, mardi .hastings@gatech.edu)

Hard and soft corals exist in tropical and sub-tropical regions usually within 25–70 m of the surface. They are invertebrates with a mouth, tentacles, nematocysts, and simple sensory organs and nervous systems. Reef-building corals live in bottom-attached colonies and depend on moving water for transport of nutrients and removal of waste products. The physical characteristics of their bodies that promote interaction with flowing water also enhance fluid dynamic forces. Thus in the near field of a powerful acoustic source, the instantaneous drag, lift, and acceleration forces imposed on a coral reef have the potential to cause injury. An analysis of the fluid dynamic forces generated by an air gun blast was used to estimate stresses induced in stony coral skeleton and polyp tissues as a function of colony size. The stress at which injury is expected was calculated by applying a factor of safety of 2 to available values for rupture strength of stony coral skeleton and longitudinal strength of polyp connective tissue. Results indicate that polyps in colonies with characteristic length of about 5 m are likely to be injured by an air gun blast when the peak-to-peak received level exceeds 260 dB re 1 μPa.

THURSDAY MORNING, 18 NOVEMBER 2010 CORAL GALLERY 1B/2B, 7:55 A.M. TO 12:00 NOON

Session 4aBB

Biomedical Ultrasound/Bioresponse to Vibration: Image Guided Therapeutic Ultrasound

Tyrone M. Porter, Chair
Dept. of Mechanical Engineering, Boston Univ., 110 Cummington St., Boston, MA 02215

Chair's Introduction—7:55

Invited Papers

8:00

4aBB1. Optimizing parameters for ultrasound-induced disruption of the blood-brain barrier. Meaghan A. O'Reilly and Kullervo Hynynen (Sunnybrook Res. Inst., 2075 Bayview Ave., Rm. C713, Toronto, ON M4N3M5, Canada, moreilly@sri.utoronto.ca)

Microbubble-mediated ultrasound-induced blood-brain barrier disruption (BBBD) is a promising and increasingly investigated technique for the targeted delivery of therapeutics in the brain. Currently in the pre-clinical stages of research, there is a need to establish treatment parameters which will produce consistent and safe BBBD. The BBB was disrupted in rats at 1.18 MHz using modified pulses which have been shown to eliminate acoustic standing waves in the skull cavity. 10-ms bursts consisting of single excitation cycles separated by a set interval were repeated at 1 Hz for 2 min. The transducer used (10-cm aperture, FN=0.8) rang for approximately 3 μs following a single cycle at 1.18 MHz. The interval between cycles was varied using values of 6, 60, 300, 600 μs, and a single cycle every second. Enhancement levels measured via contrast-enhanced T1W MRI decreased as the time interval was increased. This did not appear proportional to the time-averaged acoustic power, and a single excitation cycle every second successfully caused BBBD. Enhancement levels following 5-min sonications with bolus injection of microbubbles versus those with slow infusion were also compared. Enhancement levels were higher following bolus; however, sonications with infusion produced more consistent enhancement levels.

8:20

4aBB2. Combined ultrasound and computed tomography for image-guided acoustic therapy. Charles F. Caskey, Mario Hlawitschka, Shengping Qin, and Katherine W. Ferrara (Dept. of Biomedical Eng., Univ. of California at Davis, 451 Health Sci. Dr., Davis, CA 95616)

Fusion of computed tomography (CT) and ultrasound (US) imaging is currently used for guiding interventional procedures, including biopsy, local drug delivery, and radiofrequency ablation. While US facilitates real-time visualization of blood flow and tissue structure, CT provides a 3-D view of an extended region of interest, can differentiate tissue types, and operate in a fluoroscopic mode for catheter placement. Our laboratory is working to extend this fusion to include US therapy and imaging co-registered with CT. Here, we present methods for semi-automated registration between tracked US and CT with the objective of imaging and guiding acoustic therapy in small animals. A calibration phantom and fiducial markers were used to generate accurate transformation between the real-time US and pre-acquired CT image stack. The mean error in the transducer calibration was 1.1 $\pm$ 0.76 mm with a maximum error of 3.1 mm. A target registration error of 1.7 $\pm$ 0.9 mm was observed in US and CT images of a dual-modality phantom. We assessed *in vivo* feasibility by imaging MET-1 hind limb tumors in a mouse model using both CT and US contrast agents to enhance vascular structures in both modalities. The ability to plan and guide US-induced hyperthermia is currently being tested. [Work supported by NIHR01CA103838.]

8:40

4aBB3. High precision histotripsy treatment using active preconditioning of cavitation nuclei. Charles Cain, Tzu-Yin Wang, and Zhen Xu (2121 Carl A. Gerstacker Bldg., 2200 Bonisteel Blvd., Ann Arbor, MI 48109-2099)

Histotripsy is an ultrasound therapy that mechanically disrupts soft tissue using cavitation bubble cloud. The histotripsy process results in a damaged zone with size and shape matching those of the focal volume. Recent studies have shown that the damaged zone can be further confined to a region substantially smaller than the focal volume using a cavitation nuclei preconditioning pulse sequence delivered to the periphery of the focus before each therapy pulse. To investigate the underlying mechanism, we studied the cavitation activity and the corresponding damage in a red blood cell-gel tissue phantom. The tissue phantom consists of an agarose hydrogel with red blood cells embedded in a thin central layer. As red blood cells were damaged, local transparency would increase in the damaged zone. This phantom allows for direct observation of both the cavitation bubble cloud and the treatment outcomes in real time. Using high speed photography, we have discovered that the nuclei preconditioning pulses appear to destabilize cavitation nuclei from the periphery within a few tens of pulses. Significant cavitation damage was, therefore, prevented in the periphery. This study demonstrates the feasibility of local cavitation suppression using cavitation nuclei preconditioning. This approach allows active protection of adjacent critical structures.

9:00

4aBB4. Passive mapping for real-time monitoring of ultrasound therapy. Constantin C. Coussios, Miklos Gyongy, Robert Ritchie, Ian Webb, Sacha Nandlall, Edward Jackson, Costas Arvanitis, Miriam Bazan-Peregrino, and Manish Arora (Inst. of Biomedical Eng., Dept. of Eng. Sci., Univ. of Oxford, Oxford OX1 3PJ, UK)

Arrays of detectors placed co-axially with a therapeutic transducer make it possible to receive the high-frequency emissions produced by tissue during therapeutic ultrasound excitation with high sensitivity and in real time, whether in the presence or absence of cavitation activity. The signals received by the array of receivers can be decomposed into broadband, harmonic, and ultraharmonic components and combined using recently developed passive mapping techniques to produce real-time maps of inertial cavitation, stable cavitation, boiling, and of regions of tissue where the high-frequency ultrasonic response changes significantly. In the context of high-intensity focused ultrasound ablation, passive mapping of inertial cavitation makes it possible to visualize the focus in real time, whilst mapping of boiling can identify regions of over-treatment. Further analysis of changes in the harmonic component of high-frequency emissions in the presence or absence of cavitation can also provide reliable detection of the change in viscoelastic tissue properties that corresponds to successful ablation. In the context of ultrasound-enhanced drug delivery, cavitation mapping can help identify the tissue regions where successful extravasation of therapeutic agents from blood vessels has occurred. It is concluded that passive mapping could provide a novel, effective, and low-cost method for real-time guidance and monitoring of ultrasound therapy.

9:20

4aBB5. Acousto-optic sensing for the real-time monitoring and feedback control of non-cavitating high-intensity focused ultrasound lesion formation in optically diffuse tissues. Ronald A. Roy, Puxiang Lai, James R. McLaughlan, Andrew B. Draudt, Robin O. Cleveland (Dept. of Mech. Eng., Boston Univ., 110 Cummington St., Boston, MA 02215), and Todd W. Murray (Univ. of Colorado Boulder, Boulder, CO 80309)

The acoustic monitoring of non-cavitating high-intensity focused ultrasound (HIFU) lesions is challenging due to the relatively weak acoustic contrast between normal and necrosed tissues. Fortunately, thermal lesions posses optical scattering and absorption characteristics that can differ significantly from surrounding tissue. Diffusive optical techniques such as photo-acoustic imaging and diffusive optical tomography have been successfully employed to image absorbing structures in organs such as breast and brain. We describe a technique in which the nonlinear interaction of diffuse laser light (1064 nm) and the HIFU field is used to sense the onset and spatial extent of lesion formation in excised chicken breast. Changes in AO response correlate with lesion volume. We show that the AO signal can be used to provide real-time feedback in order to control the duration of the HIFU exposure. The lesions that were formed with AO feedback had a more consistent volume than lesions formed with a fixed duration of HIFU [Work supported by the Center for Subsurface Sensing and Imaging Systems (NSF ERC Award No. EEC-9986821).]

Contributed Papers

9:40

4aBB6. Design and testing of an interstitial therapeutic ultrasound applicator for the thermal ablation of cerebral tumors. Michael S. Canney, Remi Souchon (INSERM U556, 151 Cours Albert Thomas, 69424 Lyon and CarThéra, 47-83 Boulevard de l'Hôpital, 75013 Paris, France, michael.canney@inserm.fr), Alexandre Carpentier (Laboratoire de Recherche en Technologies Chirurgicales Avancées, Université Paris VI and Hôpital de la Pitié-Salpêtriére, 75013 Paris, France), Flavien Girard, Jean-Yves Chapelon, and Cyril Lafon (INSERM U556, 69424 Lyon, France)

Conventional treatments for cerebral tumors involve surgical resection of the lesion in combination with chemotherapy or radiotherapy. In this work, an alternative, minimally invasive approach is presented for thermally ablating cerebral tumors using an interstitial ultrasound transducer. Initial testing and characterization of a prototype device based on a mono-element design is presented. Heating experiments are performed in tissue phantoms and in *ex vivo* bovine and sheep brain. Real-time temperature monitoring and lesion characterization are performed using magnetic resonance imaging (MRI). Furthermore, obtained temperature rise and lesion volume registered on MRI are compared with numerical modeling. The results demonstrate that the prototype interstitial probes have good MRI compatibility and are capable of ablating a volume of tissue of up to several centimeters in diameter in several minutes under real-time MRI guidance. Furthermore, there is a possibility to precisely tailor the lesion volume to the treatment zone using a rotational approach. [Work supported by the ASA Hunt Fellowship and CarThéra SAS.]

9:55—10:15 Break

10:15

4aBB7. Real time two-dimensional temperature imaging for guidance and monitoring of high-intensity focused ultrasound beams. Dalong Liu, John R. Ballard (Dept. of Elec. and Comput. Eng., Univ. of Minnesota, 200 Union St. SE, Minneapolis, MN 55455), Alyona Haritonova, Jing Jiang, John C. Bischof, and Emad S. Ebbini (Univ. of Minnesota, Minneapolis, MN 55455)

We have recently introduced a fully real time 2-D temperature imaging system using diagnostic ultrasound. A SonixRP is used to collect beamformed M2D mode data with frame rates in the 200–400 fps during the application of pulsed high-intensity focused ultrasound (pHIFU). M2D mode is a modification on the SonixRP allowing for maximizing the number of scanlines per frame for a specified frame rate. This allows for capturing the full range of tissue motions during the application of the pHIFU beams, including native motions due to breathing and pulsations, radiation forces due to pHIFU, and temperature-induced strains. In this paper, we demonstrate the use of this image-guidance mode in the control of the pHIFU exposure in real time with millisecond temporal resolution. Results from heating and

lesion formation experiments in the hindlimb of nude mice *in vivo* will be presented. Temperature imaging results during the application of subtherapeutic pHIFU beams before therapeutic pHIFU lesion formation will demonstrate the advantages of this approach in the guidance and dose estimation. In addition, temperature imaging of subtherapeutic pHIFU after lesion formation allows for the measurement of changes in tissue properties that may be used as indicators of irreversible tissue damage.

10:30

4aBB8. Monitoring temperature elevations in tissues using quantitative ultrasound. Goutam Ghoshal and Michael L. Oelze (Dept. of Elect. and Comp. Eng., Univ. of Illinois at Urbana-Champaign, 405 N Mathews, Urbana, IL 61801, gghoshal@illinois.edu)

High intensity focused ultrasound (HIFU) is a noninvasive technique that has great potential for improving thermal therapies but requires accurate monitoring of lesion formation. Quantitative ultrasound (QUS) is a novel imaging technique that can improve monitoring of HIFU treatment by quantifying tissue changes. Ultrasonic backscatter experiments were performed on tissue-mimicking phantoms, fresh rabbit liver, and beef liver samples versus increases in temperature from 37 to 50 °C in 1 °C increments. Two parameters were estimated from the backscatter coefficient [effective scatterer diameter (ESD) and effective acoustic concentration (EAC)] and two parameters were estimated from envelope statistics (k parameter and μ parameter) of the backscatter. No significant changes in ESD were observed for the phantoms but the ESD increased by more than 10% with increasing temperature in the liver samples. Significant decreases in EAC of 20%–40% were observed in all the samples. No specific trends were observed in envelope statistics versus temperature. From the results, it was observed that some QUS parameters were more sensitive to lesion formation than others. The results suggest that QUS has the potential to be used for noninvasive monitoring of lesion formation because QUS may be sensitive to tissue microstructure changes. [Work supported by NIH R01EB008992.]

10:45

4aBB9. Real-time passive acoustic monitoring of tissue damage during thermal ablation by high-intensity focused ultrasound. Sacha D. Nandlall, Edward Jackson, and Constantin-C. Coussios (Inst. of Biomedical Eng., Dept. of Eng. Sci., Univ. of Oxford, Oxford OX3 7DQ, United Kingdom, sacha.nandlall@eng.ox.ac.uk)

This work proposes a novel method for monitoring tissue ablation by high intensity focused ultrasound (HIFU) in real time. The proposed method employs the passively acquired acoustic signal emitted from the HIFU focus throughout an exposure. A total of 161 exposures were performed in seven freshly excised ox livers using 1.067-MHz HIFU at a 95% duty cycle. Acoustic emissions were recorded using a 15-MHz passive detector aligned confocally and coaxially with the HIFU transducer. Lesion presence and size were ascertained by slicing the tissue in the transverse and axial focal planes post-exposure. Results demonstrate that the successful formation of HIFU lesions in *ex vivo* ox liver is highly correlated with the presence of pronounced dips in the magnitude of the received signal at integer harmonics of the insonation frequency. Optimization and validation of a detection algorithm based on this observation show that the detector agrees with the post-exposure lesioning assessment in 75% of cases overall, and that the error rate drops further for exposures shorter than 1 s or longer than 2 s. Such a detector could therefore provide a low-cost means of effectively monitoring clinical HIFU treatments passively and in real time.

11:00

4aBB10. Visualization of a focused ultrasound beam to guide radiation force-induced clearance of kidney stones. Neil R. Owen, Bryan W. Cunitz, Anup Shah, Francesco P. Curra, Michael R. Bailey, and Lawrence A. Crum (Ctr. for Industrial and Medical Ultrasound, APL-UW, Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, nowen@apl.washington.edu)

The incidence of kidney stones within the US population is now 10%, and rising. Many patients present with small stones, primary or recurrent, do not indicate interventional stone removal. We previously described a new stone removal method employing selective application of acoustic radiation force, at diagnostic output levels, to reposition stones for passive clearance. In this method, an imaging array transducer transmits pulses for image guidance and focused pulses to reposition the stone. Here we propose a new flash imaging modality to visualize the focused pulse to confirm targeting on the stone. To visualize the focused beam, short pulses were phase-delayed across the transducer aperture to transmit a focused wave, from which echo data were collected, beamformed, and overlaid on a B-mode image. The beam profile is visible because echo amplitude is higher within the convergent, focal, and divergent regions. During experiment, a stone was placed within a tissue phantom simulating the kidney lower pole and ureter. Once identified with B-mode imaging, focal delays were calculated, targeting was confirmed by the beam visualization modality, and the stone was repositioned. Flash imaging visualization of the focused beam could be similarly applied to high-intensity focused ultrasound therapy. [Work supported by NIH DK43881 NSBRI-SMST01601.]

11:15

4aBB11. Acoustic stone localization during lithotripsy. Jonathan M. Kracht, Nicholas J. Manzi, Gonzalo R. Feijo'o, Paul E. Barbone, and Robin O. Cleveland (Dept. of Mech. Eng., Boston Univ., 110 Cummington St., Boston, MA 02215, jkracht1@bu.edu)

It is desirable to mitigate damage to kidney tissue induced by shock waves administered during lithotripsy. Stone movement due to patient respiration causes a fraction of the shock waves to miss the stone. The goal here is to ensure that shock waves are delivered to the kidney only when the kidney stone is in the focal region of the lithotripter. We present the design of a collar with an array of ultrasound transducers that can be retrofitted to a clinical lithotripter. The transducers are used in a pulse-echo mode and a hybrid technique combining numeric time-reversal and MUSIC (MUltiple-SIgnal-Classification) is employed to determine the location of a kidney stone relative to the focus of the lithotripter. A model of the acoustic collar was used to select system parameters including the transducer count, center frequency, and transducer placement. The performance of the collar was assessed by translating artificial kidney stones in water and using porcine body wall as an aberrating layer. Performance will be reported of the system including resolution limit, ability to target multiple stone fragments, and the effect of an aberrating layer. [Work supported by NIH.]

11:30

4aBB12. Perfluorocarbon nanoemulsions as contrast agents for imaging and ultrasound-mediated targeted chemotherapy. Natalia Rapoport (Dept. of Bioengineering, Univ. of Utah, 72 S. Central Campus Dr., Rm. 2624, Salt Lake City, UT 84112, natasha.rapoport@utah.edu)

During the last decade, achievements of nanomedicine have made possible image-guided drug targeting to disease sites. In oncology, tumor targeting of chemotherapeutic agents can be realized by drug encapsulation in nanocarriers such as liposomes, polymeric micelles, nanoemulsions, etc. A degree of tumor targeting may be increased by developing stimuli responsive nanocarriers that release their drug load in response to external stimuli such as heat, light, or ultrasound. Using ultrasound allows combining imaging and therapy. However, many mechanistic aspects of the ultrasound-mediated drug delivery remain elusive. Which mode of ultrasound, mechanical or thermal, is optimal for drug delivery? Which is the optimal timing of ultrasound application in relation to drug injection? What would be the best target for molecularly targeted carriers? These problems will be discussed for paclitaxel-loaded perfluorocarbon nanoemulsions that manifest a unique combination of properties (acoustic droplet vaporization, cavitation-induced drug release, ultrasound and 19F MR contrast, excellent therapeutic effectiveness). Drug carrier biodistribution was monitored by ultrasound and 19F imaging. Focused ultrasound treatment of red fluorescence protein (RFP) transfected orthotopic pancreatic tumors proceeded under the control of MR thermometry. RFP imaging allowed visualization of ultrasound beam traces and treatment results. [Work supported by NIH R01 EB001033.]

11:45

4aBB13. Exploitation of cavitation-enhanced heating for release of doxorubicin from thermosensitve liposomes by therapeutic ultrasound. Eleonora Mylonopoulou, Miriam Bazan-Peregrino, Costas D. Arvanitis, and Constantin C. Coussios (Inst. of Biomedical Eng., Eng. Dept., Oxford Univ., Old Rd. Campus Res. Bldg., Headington, Oxford OX3 7DQ, UK, eleonora.mylonopoulou@seh.ox.ac.uk)

The systemic toxicity of current chemotherapeutic treatments motivates the investigation of targeted delivery techniques to restrict the action of drugs to the tumor region. ThermoDox© copyright (Celsion Corporation) consists of doxorubicin, a common anticancer agent, encapsulated within thermally sensitive liposomes designed to release their contents above 39 °C. Activation of such an agent with the use of HIFU, which can generate localized heating non-invasively, potentially combines the benefits of targeted chemotherapy and hyperthermia while minimizing undesirable systemic side-effects. ThermoDox© release was investigated and optimized in a novel cell-embedding, tissue-mimicking material. The material was exposed to 1.1 MHz HIFU using a range of clinically relevant pressure amplitudes, duty cycles, and exposure durations to identify optimal insonation conditions for complete doxorubicin release, resulting in cell death that is solely due to drug activity. The corresponding temperature profiles were mapped throughout the focal region using embedded needle thermocouples. Drug release and cell viability were quantified using fluorescent microscopy and cell luminescence, respectively. Optimal conditions for drug release only due to the thermal effects of HIFU, with and without cavitation, were identified and cell death only due to drug release was achieved, demonstrating the potential for safe and effective targeted drug release by HIFU-induced hyperthermia.

THURSDAY MORNING, 18 NOVEMBER 2010 — CORAL SEA 1/2, 11:00 A.M. TO 12:00 NOON

Session 4aED

Education in Acoustics: Take 5’s

Andrew C. H. Morrison, Chair
Physics Dept., DePaul University, Chicago, IL 60614

For a Take-Five session no abstract is required. We invite you to bring your favorite acoustics teaching ideas. Choose from the following: short demonstrations, teaching devices, or videos. The intent is to share teaching ideas with your colleagues. If possible, bring a brief, descriptive handout with enough copies for distribution. Spontaneous inspirations are also welcome. You sign up at the door for a five-minute slot before the session starts. If you have more than one demo, sign up for non-consecutive slots.

THURSDAY MORNING, 18 NOVEMBER 2010 — CORAL SEA 1/2, 8:30 TO 10:35 A.M.

Session 4aMU

Musical Acoustics: Effects of Wall Vibrations in Musical Instruments

Thomas R. Moore, Chair
Dept. of Physics, Rollins College, Winter Park, FL 32789

Chair’s Introduction—8:30

Invited Papers

8:35

4aMU1. Bamboo pipe wall vibrations in Asian free reed instruments. James P. Cottingham (Phys. Dept., Coe College, 1220 First Ave. NE, Cedar Rapids, IA 52402, jcotting@coe.edu)

Wall vibrations in the bamboo pipes of the Asian free reed mouth organs have been investigated. The pipes studied include quasi-cylindrical examples from a khaen built in northeastern Thailand and some pipes from a Southeast Asian free reed gourd pipe. Modal frequencies and mode shapes of a number of the pipes were measured using mechanical excitation. The modes of vibration of these pipes were also simulated using measured physical properties of the bamboo, which has a relatively low density and elastic modulus, and a significant difference between the elastic moduli parallel and perpendicular to the grain. Vibration of the pipe walls was also studied for the mechanically blown reed-pipe combination. Measurements of pipe input impedance have been made, showing changes occurring as a result of damping the pipe vibrations. Possible effects of wall vibrations on the radiated sound have also been explored.

8:55

4aMU2. Vibro-acoustics of organ pipes: Coincidence effect between acoustical and mechanical modes. Francois Gautier (Laboratoire d'Acoustique de l'Universit du Maine, UMR CNRS, Av. O. Messiaen, 72085 Le Mans, France, Francois.Gautier@univ-lemans.fr)

How does the wall vibration of a wind instrument affect its sound? This question has engendered a long-lasting debate among scientists, musicians, and instrument makers. In fact, the wall vibrations are easily felt or measured on most wind instruments. However, their influence on the produced sound is more difficult to bring to light, because the fluid-structure couplings involved are weak. This leads to the conclusion that the wall vibration effect does not induce any significant or audible contribution except when particular conditions are gathered. The conditions correspond to coincidence relationships between acoustical and structural modes of the instrument. A century ago, Miller published an experimental study about vibrations of organ pipes. [D. Miller, Science **29**, 161–171 (1909)]. He used a double walled organ pipe, for which the space between the two walls could be filled with water while the pipe was sounded, presumably damping the wall vibrations. Miller then observed, without explaining it, that the filling led to the unusual behavior of the pipe, clearly audible. Some heights of the water jacket produced pitch changes or unstable tones. A copy of this historical experiment has been made recently and is now interpreted using the coincidence concept.

9:15

4aMU3. Coupling between air column and structural vibrations in the flaring bell of brass wind instruments. Wilfried Kausel (Inst. of Musical Acoust., Univ. of Music, Anton von Webernpl. 1, A-1030 Vienna, Austria, kausel@mdw.ac.at) and Thomas R. Moore (Rollins College, Orlando, FL 32789)

The effect of wall vibrations on the radiated sound of wind instruments has been under debate for quite a long period of time now. Experiments with brass wind instruments like trumpets and horns have shown that this effect is much more pronounced in the steeply flaring bell than in a cylindrical tube. A theory for the coupling between air column and structural oscillations is presented and applied to the flaring bell of brass wind instruments. It will be shown that the effect of a finite Young's modulus of the wall material on the radiated sound becomes more significant when the flare of the bell is steep. Theoretical predictions for input impedance and sound pressure transfer function are in agreement with measurements made on real instruments. The theory is also extended to the case of randomly broken axial symmetry, a realistic assumption if hand-made thin walled brass instrument bells are taken into consideration. Under the assumption of 0.5% radial deviations from a perfectly circular cross-sectional shape, breathing modes become one order of magnitude more effective than without pre-deformations. It is also shown that purely elliptical pre-deformations are much less effective in modifying air column characteristics than randomly distributed deformations.

9:35

4aMU4. Influence of wall vibrations on the transient sound of flue organ pipes. Malte Kob (Erich Thienhaus Inst., Univ. of Music Detmold, 32756 Detmold, Germany, kob@hfm-detmold.de)

The question of how the vibrations of a metal flue organ pipe contribute to the perceived sound has been investigated intensively in the last decades. Numerical, experimental, and theoretical works have shown that the influence can modify fundamental frequency, amplitude, and stability of the sound. However, these changes are small in the stationary sound of a pipe and vary with boundary conditions. Fewer investigations have concentrated on the transient sound of a pipe. As one result of the research done in a project within the Gteborg Organ Art center, it was shown that due to the casting process, baroque flue organ pipes often have considerably thinner upper ends than modern pipes. To prove the influence of the wall vibration on the perceived sound, recordings of a blown pipe with and without damping of the wall have been carried out. Furthermore, investigations of eigenmodes of the pipe under test using finite element calculations and pressure and velocity measurements with force excitation at the labium with subsequent modal analysis were made. The results confirm that some structural modes are excited when the pipe is blown and some components of the transient spectra change when the pipe is damped.

9:55

4aMU5. Player-to-player variability of the dependence of brass-instrument timbre on bell alloy. Robert Pyle (S. E. Shires Co., 11 Holworthy Pl., Cambridge, MA, 02138, rpyle@post.harvard.edu)

Previous measurements showed a substantial difference between two trombone players, call them player A and player B, in the degree to which the radiated spectrum changed with changes of bell alloy. [J. Acoust. Soc. Am. **125**, 2597 (2009).] Player A maintained a much more consistent timbre across changes of bell alloy than did player B. It is hypothesized that player A altered his embouchure to offset the effects of changes in the instrument to a much greater extent than did player B. This will be tested by simultaneous measurement of sound pressure internally within the mouthpiece and externally on the bell axis. If the hypothesis is true, changing the bell alloy should be reflected by a greater change in the signal within the mouthpiece cup for player A than for player B. Additional tests are planned with instrument-to-player feedback disturbed by shielding the player's ears and introducing masking noise.

10:15

4aMU6. Vibrations in brass instrument bodies: A review. Peter L. Hoekje (Dept. of Phys. and Astronomy, Baldwin-Wallace College, 275 Eastland Rd., Berea, OH 44017, phoekje@bw.edu)

The bodies of brass instruments vibrate when played, as evident to any player. These vibrations are driven mechanically at the player's lips and by coupling with the internal acoustic field throughout the instrument. The coupling can equally well allow energy flow in the opposite direction, from the body to the player's lips and to the internal acoustic field. The vibrating shell can also radiate directly into the room. Studies involving human or machine players suggest that the effect of the wall vibrations on the acoustic spectral components is of the order of a few decibels. However, for the most part, the direct radiation of the wall vibrations is approximately 20 dB

below the level of the radiated internal acoustic field. The wall vibrations have a slightly different time envelope, though, and also have a formant-like region around a few kilohertz. Some of the pieces of this complex puzzle that need further investigation include the interaction between player and instrument, the effect of sound level on the coupling, the perceptibility of the radiated vibrations by auditors, and the variations within the brass family.

THURSDAY MORNING, 18 NOVEMBER 2010 CORAL KINGDOM 1, 8:00 TO 10:00 A.M.

Session 4aNSa

Noise: Environmental Noise in Cities II: Measurements, Health, and Legislation

Martha G. Orozco Medina, Cochair
Univ. de Guadalajara, Km. 15.5 Carretera Nogales, Las Agujas, Zapopan Jal. CP 45110, Mexico

Sergio Beristain, Cochair
Mexican Inst,. of Acoustics, P.O. Box 12-1022, Narvarte 03001 DF Mexico City, Mexico

Contributed Papers

8:00

4aNSa1. Measurement of the contribution of each individual vehicle to the road traffic noise. David Ibarra, Pedro Cobo, and Teresa Bravo (CAEND, CSIC-UPM, Serrano 144, 28006 Madrid, Spain, david.ibarra@caend.upm-csic.es)

Road traffic is the main noise source in urban environments. While it is feasible to measure the noise emitted by road vehicles in standard conditions [ISO 11819-1;2, ISO 362], nowadays it is not possible to detect the contribution of each vehicle to the overall road traffic noise in real driving conditions. However, some authors have demonstrated previously that there is a high correlation between maximum noise levels and mean annoyance. Therefore, an efficient control procedure to reduce road traffic noise annoyance should be to identify those vehicles contributing with the maximum noise levels. The main goal of this paper is to describe a boarded acoustic system to measure the contribution of each vehicle to the overall road traffic noise. The system consists of two microphones, one close to the tire furthest from the exhaust duct, and the other close to the engine air intake. The signals picked up by these microphones are processed to calculate the evolution of the Leq,1s and the levels histogram along the driving path. The tests carried out in an urban circuit in Madrid (Spain), with two cars (gasoline and diesel) and five drivers, demonstrate that it is possible to detect those especially noisy vehicles.

8:15

4aNSa2. Train noise: Peruvian case study to legislate its emission. Elena Gushiquen and Walter Montano (Arquicust SRL, Av. Javier Prado Oeste 304, Lima 17, Peru, montano_walter@yahoo.com.ar)

This work presents the results of train noise EIS in order to write down legislation for allowed maximum limits for Peruvian gas, electric, and diesel convoy.

8:30

4aNSa3. Results and methodologies of airport noise studies. Max J. Glisser and Christian E. Gerard (Control Acustico, Villaseca #21, Of. 505, Nunoa, Santiago, Chile)

Some 7 years ago, Control Acustico Company is working on surveys to get information on noise levels generated by airplane operation at airports, both in Chile and abroad. These surveys included Chilean airports such as Arturo Merino Benítez in Santiago, Maquehue in Temuco, Carriel Sur in Concepción, El Tepual at Puerto Montt, Mataveri at Eastern Island, and Espanol in Barcelona, Spain. This paper displays some results of these surveys using the software Integrated Noise Model from the Federal Aviation Administration, discusses methods used to obtain curves of equal sonority of YLDN, among them measuring of 24 h, or longer, continuous sound pressure levels, and shows an analysis allowing to calibrate the models, fleet structure, approaching and take-off pathways, and finally results evaluation according to the North American Rules, FAR, Chap. 150, "Airports Noise Compatibility Plan."

8:45

4aNSa4. Comparison of noise emissions from *in situ* measurements construction projects in chile with those from British Standard BS 5228-1:2009. Max J. Glisser and Christian E. Gerard (Control Acustico, Villaseca #21, Of. 505, Nunoa, Santiago, Chile, info@controlacustico.cl)

A large number of noise impact studies for projects of different types are developed annually as part of the System of Environmental Impact Assessment in Chile. One of the most important variables to predict the emission of noise is the sound power of noise sources involved. In this study, we compare sound power levels (Lw) calculated according to measurements of sound pressure levels by the Acoustic Control Company in different construction projects and the levels calculated from the British Standard BS 5228-1:2009 "Code of Practice for Noise and Vibration Control on Construction and Open Sites Noise". The study considers the most frequently involved machines in construction tasks, such as loader trucks, cement mixer trucks, wheeled loaders, rollers, angle grinders, cranes, circular saws, and backhoes.

9:00

4aNSa5. Development and calibration of an acoustic model for a mixed residential-industrial zone. Víctor Cortínez, Martín Sequeira, Adrián Azzurro (CIMTA, Univ. Tecn. Nac., Bahía Blanca, 8000 Buenos Aires, Argentina, vcortine@frbb.utn.edu.ar), and Facundo Ponds (CTE, Municip. de Bahía Blanca, 8000 Buenos Aires, Argentina)

This paper deals with the noise characterization of a residential mixed area, originating from stationary industrial sources of a petrochemical establishment in the city of Bahía Blanca. The sound power levels are estimated from a set of acoustics measurements and a subsequent implementation of a method similar to that proposed in the ISO 8297 standard. Later, an outdoor sound propagation model is developed and calibrated by using measurement data located near the urbanized sector. The objective of this work is to provide a useful prediction tool of the noise impact on the nearby residential area and to study possible acoustic treatments in order to plan a suitable urban scenario according to local legislation.

9:15

4aNSa6. Regulatory protocol for Colombia's national regulations on environmental noise and noise emission Act 0627. Carlos Echeverri Londoño (Programa de Ingeniería Ambiental, U. de Medellín, Carrera 87 No. 30-65, Apartado Aéro 1983, Medellín, Colombia, cecheverri@udem.edu.co), Alice Elizabeth González (F. de Ingeniería UdelaR, CP 11.300, Montevideo, Uruguay), Alejandro Acosta, and Juan Fernando Lenis (U. de Medellín, Apartado Aéreo 1983, Medellín, Colombia)

Act 0627 of April 7, 2006, was an important progress regarding national regulations on environmental noise and noise emissions. However, during the first years of its validity, doubts about its interpretation and application have been raised. In response to this issue, the Ministry of Environment, Housing and Territorial Development has signed a contract with the University of Medellin for the development of a detailed protocol that would allow the unambiguous and rigorous implementation of the Act. The protocol was submitted to the Ministry in January 2010. It includes detailed guidelines for organizing and carrying out measurements of noise levels (including the selection of points to consider), processing and interpretation of measurement results, minimum content of the reports delivered as a result of the measurements, performing acoustic maps, and guidelines for the preparation of plans for noise abatement.

9:30

4aNSa7. Discussing the effects of environmental noise upon the immune system. M. Orozco Medina, A. Orozco Barocio, A. Figueroa Montaño, and N. Ochoa Ramos (IMACH, Universsidad de Guadalajara, Km. 15.5 carretera a nogales, Las Agujas, Zapopan, Jal., C.P. 45110, Mexico, morozco@cucba.udg.mx)

Human health relates either positively or negatively to environmental stimuli. It is well documented in science literature, how toxic substances disturb lung function and how ionizing radiation damage genetic material. In modern societies, noise is an agent entering the list of the diverse environmental stressors of urban population. Noise in the urban environment affects quality of life, as it alters interpersonal communication, sleep, and is a potential risk for hearing loss due to continuous and chronic exposure, as well as some other documented effects. Recently the response of the immune system to the effects of noise is taking importance in the scientific community. Researchers are studying and applying unconventional approaches in order to find any relationship between the amount of noise and the response of the immune system. This paper discusses evidence of the relationship among environmental noise and the immune system, and also questions the methods and techniques applied to finally propose issues to be considered when studying relationships among environmental noise and the immune system.

9:45

4aNSa8. Comparative study of levels of care in children exposed to environmental noise on three campuses in the city of Guadalajara, 2009. N. Preciado Caballero, M. Orozco Medina, A. Figueroa Montaño, and M. Ruvalcaba (IMACH, Universidad de Guadalajara, Km. 15.5 Carretera a Nogales, Las Agujas, Zapopan, Jal., C.P. 45110, Mexico, morozco@cucba.udg.mx)

Noise is a pollutant that significantly affects the quality of life and health of individuals. Individuals exposed to noise may experience stress, discomfort, sleep disturbances, impaired immune system, lack of alertness, hearing loss (to varying degrees), cardiovascular disease, increased aggressiveness of individuals, low productivity, and traffic and workplace accidents. [WHO (1999); Berglund and Lindvall (1995); Clark *et al.* (2006); Haines *et al.* (2001)]. In the city of Guadalajara, a large number of schools are located on heavy traffic roads or busy streets, or in hot spots of environmental noise caused by motor vehicles, which exposes children and teachers who attend them to unfavorable acoustic conditions to develop their school tasks. Thus having a negative effect on their attention span, memory, and performance, as well as damaging physical and emotional health. This study aims to contribute to the study of sound quality by means of the analysis of noise conditions at elementary schools, which were located at critical points in terms of heavy vehicular flow. Besides, it analyzes the perception of environmental noise within the same facilities, and how this compares with results from schools located in quiet streets.

THURSDAY MORNING, 18 NOVEMBER 2010 CORAL KINGDOM 1, 10:15 A.M. TO 12:00 NOON

Session 4aNSb

Noise: Culture and Noise

Fernando J. Elizondo-Garza, Chair

Mechanical and Electrical Engineering School, Univ. Autonoma de Neuvo Leon, P.O. Box 28, San Nicolas 66450, N.L. Mexico

Chair's Introduction—10:15

Invited Papers

10:20

4aNSb1. Ritual and noise. Roberto Rebolloso (Sociology Faculty, Philosophy and Lang. Arts School, Universidad Autónoma de Nuevo León, 66450 NL, Mexico, rfrebolloso@gmail.com) and Fernando J. Elizondo-Garza (Universidad Autónoma de Nuevo León, Cd. Universitaria, San Nicolás, 66450 NL, Mexico)

Starting from an ethnographic analysis of primitive rituals, the evolution of the production of noise in different kinds of rites is analyzed up to the contemporary society. The definition of rite and noise, and the concatenation of stimuli on people produced by rites that induce habits—that in a certain sense are addictive—are discussed. The use of noise in tribal, religious, sport, political, and festive rituals is described.

10:40

4aNSb2. Environmental noise culture. Pablo Lizana (Acoust. Lab,. ESIME, IPN, Antonia 15 15, San Jeronimo Lidice, DF, Mexico, plizana@ipn.mx)

Noise generation and perception are highly related to given societies, so in order to improve the acoustic quality of life for communities, it is necessary to include noise items in formal education. The American States Organization is doing research on the relation of science, technology, society, and values to help implement programs to set these principles for the new generation. Some of these relationships are discussed.

11:00

4aNSb3. Toward an engineering of noise design in rituals. Fernando J. Elizondo-Garza (Acoust. Lab., Mech. and Elec. Eng. School, Universidad Autónoma de Nuevo León. P.O. Box 28 F, Cd. Universitaria, San Nicolás, 66450, NL, Mexico, fjelizon@hotmail.com)

The design of rites is not something new since shamans, tribal heads, and leaders from very ancient times knew clearly that the conjunction of physical–chemical–biological stimuli produces very useful states of euphoria to reinforce the sense of unity and thus to obtain the favors of multitudes. These rites, or of tribal, religious, sport, or festive character, in many cases, use the noise like one of their central elements. In this paper a structure and tools to the noise design in rituals are proposed.

11:20

4aNSb4. Music and noise. Ricardo Martinez Leal (Electroacoustic Music Lab., School of Music, Universidad Autónoma de Nuevo León, Privada Neil Armstrong #115, Col. Mirasierra, San Pedro Garza García, Nuevo León, CP 66240, Mexico, rimarle@hotmail.com) and Fernando J. Elizondo-Garza (Universidad Autónoma de Nuevo León, Cd. Universitaria, San Nicolás, 66450 NL, Mexico)

Although the noise takes part in sonorous rites from antiquity, the development of the concept of harmony, which allowed the development of occidental music as art, initially left the noise outside music. From conceptual reviews and the modern music theory, the functions of noise in the sonorous arts were rediscovered and its use received a strong impulse—through electroacoustic music and other artistic tendencies intended to produce reactions on a wide sense on listeners—when intentionally included as noise in the "musical" production. In this paper a historical review of the relation between music and noise with emphasis on the conceptual progresses on the last two centuries is done.

11:40—12:00 Panel Discussion

THURSDAY MORNING, 18 NOVEMBER 2010 CORAL GALLERY 1A/2A, 7:55 A.M. TO 12:00 NOON

Session 4aPA

Physical Acoustics: Thirty Years of Resonant Ultrasound I

Albert Migliori, Cochair
Los Alamos National Lab., Los Alamos, NM 87545

Veerle M. Keppens, Cochair
Dept. of Materials Science and Engineering, Univ. of Tennessee, Knoxville, TN 37996

Chair's Introduction—7:55

Invited Papers

8:00

4aPA1. Twenty years of fun with resonant ultrasound. Robert G. Leisure (Dept. of Phys., Colorado State Univ., Fort Collins, CO 80523-1875, leisure@lamar.colostate.edu)

Resonant ultrasound (RUS) is an elegant method for determining the elastic and anelastic properties of materials. The development of computational methods, robust computer codes, and sensitive instrumentation by a few key players has resulted in a simple, yet powerful, tool. The speaker was introduced to RUS 20 years ago. During the past two decades he, along with numerous students and in collaboration with many colleagues, has used this technique to study various problems: phase transitions, texture in alloys, and the elastic and anelastic properties of several materials—hydrogen absorbing compounds and alloys, nanocrystalline materials, and quasicrystalline materials—over an extended temperature range. All of these problems would have been difficult, if not impossible, to address with conventional ultrasonic methods. A few examples—phase transitions and internal friction due to hydrogen hopping, in particular—will be used to illustrate the power of RUS.

8:30

4aPA2. Resonance ultrasound spectroscopy: Highlights of the mathematics. Paul Heyliger (Civil Eng., Colorado State Univ., Fort Collins, CO 80523) and Hassel Ledbetter (Mech. Eng., Univ. of Colorado, Boulder, CO 80309)

We highlight the key mathematics associated with resonance ultrasound spectroscopy (RUS). From the physics viewpoint, RUS is part of the general problem of elastic vibrations in continuous media. A solid can transmit three waves: one quasilongitudinal and two quasitransverse. At resonance, we see standing waves, which we can represent by an eight-summation Fourier-series wave function. We begin by reviewing briefly the 1-D vibrating-string problem that arose in the music of many early civilizations (about 4000 BC). Pythagoras of Samos (582–507 BC) first treated this problem mathematically. In the early beginnings of modern science, both Galileo (1564–1642) and Mersenne (1588–1648) considered vibrating strings, which have a perfect analog in axial-bar vibration. Lamb and Lamé contributed in the 1800s. Probably the first inverse RUS measurement came from Cole and Frazer (1964), who studied the sphere-vibration problem using graphical methods. Most of our talk will highlight the remarkable mathematical contributions by Demarest (then at Reed College), Ohno (Ehime University), and Visscher (Los Alamos National Laboratory) in considering anisotropic vibrations of unusual-shape solids.

9:00

4aPA3. Resonant ultrasound spectroscopy for small or arbitrarily shaped samples and thin films. J. D. Maynard (Dept. of Phys., The Penn State Univ., Univ. Park, PA 16802)

When new materials are developed, among the first properties which should be measured are the elastic constants, and resonant ultrasound spectroscopy (RUS) has been found to be an appropriate method for accomplishing such measurements. However, novel materials to which RUS might be applied are often available only as small samples, only a few hundred micrometers in size, or as thin films deposited on a substrate, with film thicknesses of only a few hundred nanometers. On some occasions samples are fragile or chemically reactive so that they cannot be polished into the shapes required by conventional RUS; for such cases a finite element method is required. In this talk the development of RUS for small or arbitrarily shaped samples and thin films will be presented. [Research supported by NSF DMR 0804105.]

Contributed Papers

9:30

4aPA4. On complications associated with measuring the piezoelectric properties of relaxor single crystals with high electromechanical coupling using resonant bar samples and comparison with a lumped parameter method. David A. Brown, Corey L. Bachand, and Boris S. Aronov (BTech Acoust. and Univ. of Massachusetts Dartmouth, 151 Martine St., Fall River, MA 02723)

The development of relaxor single crystal piezoelectric materials with exceptionally high electromechanical coupling (k above 0.85) and high elastic and piezoelectric anisotropy presents challenges and questions to the applicability of the use of traditional resonant bar methods (IEEE Standard on Piezoelectricity, 176-1987, Sec. 6.4) for determining the constituent piezoelectric properties. Standard bar techniques for determining coupling coefficient rely on the single degree of freedom electromechanical circuit models for sufficiently long samples whereby measurements of isolated resonances and antiresonance frequencies are uncorrupted by lateral resonances. The limited applicability of the standard method for these piezoelectrics has motivated the development of an alternative method that avoids many of the previous short comings. A lumped parameter dumb-bell resonator approach was implemented by employing a small single crystal piezoelectric sample sandwiched between reactive terminations whereby the piezoelectric and elastic properties may be determined. Coupling coefficients k-33 and k-31 as high as 0.88 were measured for single crystal CPSC160-95 and CPSC180-120 by Ceracomp.

9:45

4aPA5. Resonant acoustic spectroscopy of the fluid saturation effects in a carbonate rock. Vyacheslav Averbach, Vladimir Bredikhin, Andrey Lebedev, and Sergey Manakov (Inst. of Appl. Phys., Russian Acad. Sci., 46 Ulyanov Str., Nizhny Novgorod 603950, Russia)

We present the results of the acoustic spectroscopy of a carbonate sedimentary rock under various saturation degrees. The data for the same sample were obtained both for elasticity and dissipation tensors within the 2-octaves frequency range. High accuracy of acoustic measurements allowed distinguishing all three stages in the saturation process: condensation, meniscus creation, and filling the pore space with fluid. Both dry rock parameters and the variations observed are in a good agreement with standard granulometry and chemical analysis. Simple estimations of the relative changes in elastic moduli enable clear explanations of these changes observed. The alteration in the frequency dependence of attenuation coefficient allowed proposing reasonable dissipation mechanisms in porous media that can also be useful in the acoustics of ocean sediments.

10:00—10:15 Break

Invited Papers

10:15

4aPA6. Resonant ultrasound and high temperature superconductivity. Greg Boebinger (Natl. High Magnetic Field Lab., Florida State Univ., 1800 E Paul Dirac Dr., Tallahassee, FL 32310, gsb@magnet.fsu.edu)

The first experimental test of the 1957 Bardeen–Cooper–Schreiffer theory of superconductivity was a measurement of ultrasonic attenuation. Since then, ultrasound has been sporadically used as a probe of the fundamental physics of superconductivity. One primary difficulty in the application of ultrasonic techniques to high temperature superconductivity has been dissipation due to twin grain boundaries in the crystals. With the availability of detwinned yttrium-barium-copper-oxide superconducting single crystals, resonant ultrasound can now make precision measurements that use the fundamental connection between sound speeds and thermodynamics to explore questions relating to the nature of the so-called pseudogap state, the precursor state to superconductivity, in which preformed

Cooper pairs and a strong superconducting interaction have been proposed. Because the simplistic connection from superconductivity to the bulk modulus is of the order of the square of the ratio of the superconducting transition temperature to the Fermi temperature, only high precision techniques such as resonant ultrasound spectroscopy are capable of observing the expected—but small—signatures.

10:45

4aPA7. Resonant ultrasound spectroscopy of complex vanadium spinel oxides. Veerle Keppens (Dept. Mater. Sci. and Eng., The Univ. of Tennessee, Knoxville, TN 37996)

Complex behavior in spinel oxides is primarily due to the geometry of the pyrochlore sublattice. Spinel vanadates AV_2O_4 are of particular interest, as they exhibit an additional degree of freedom due to orbital degeneracy, and the physics of these materials is expected to involve an interplay between spin frustration, collective Jahn–Teller effects, and the relativistic spin-orbit interaction. We report here a systematic study of the elastic response of vanadium spinel oxides, where the *A* cation is Zn^{2+}, Fe^{2+}, or Mn^{2+}. ZnV_2O_4 is characteristic of the entire series and has received extensive theoretical attention. All V-spinels are cubic at room temperature, but as the materials are cooled, they undergo structural and magnetic phase transitions that are currently the subject of intense experimental and theoretical interest. Resonant ultrasound spectroscopy was used from 2–400 K and in magnetic fields up to 9 T to obtain the elastic moduli. The temperature-dependence of the moduli is found to strongly reflect the structural transition, as large softening is observed when the transition temperature is approached. In the case of MnV_2O_4, measurements in magnetic field reveal an additional feature near 50 K, which could represent a striking manifestation of direct spin-orbit coupling. [Work supported by NSF-DMR-0804719.]

11:15

4aPA8. Electronics for resonant ultrasound. Albert Migliori (Los Alamos Natl. Lab., MS E536, Los Alamos, NM 87545)

An essential advantage of resonant ultrasound spectroscopy (RUS) is that there is no need to bond the transducers to the specimen. Needed to preserve the free-surface boundary conditions so important to RUS, the lack of bonding also makes signals weak. Exacerbating the signal-level problem is a requirement of RUS to measure over a broad frequency range, precluding the use of gain-enhancing resonant transducers. So with broadband transducers and weak contact, the demands on RUS electronics, from preamps to computers, are substantial. The evolution of electronics for RUS has paralleled the availability of high-performance analog-to-digital converters, FET-based electronic components, and digital signal processing. We describe here all the essential features for state-of-the-art RUS signal acquisition. [Work at Los Alamos National Laboratory (LANL) was supported by NSF-DMR-0654118, DOE, the State of Florida. NSF-DMR 0602859 (FSU), and LANL LDRD-DR20070013. LANL is operated by LANS LLC.]

Contributed Paper

11:45

4aPA9. On the correlation between the acoustic anisotropy and the magnetic susceptibility anisotropy of the rocks. Andrey Lebedev, Vladimir Bredikhin (Inst. of Appl. Phys., Russian Acad. Sci., 46 Ulyanov St., Nizhny Novgorod 603950, Russia), and Yuri Bretshtein (Inst. of Tectonics and Geophys., Far East Div. of Russian Acad. Sci., Khabarovsk, 680000, Russia)

Using resonant acoustic spectroscopy technique, we studied several rock samples. Acoustic data were compared with the data for magnetic susceptibility of the same samples. The correlation between the acoustic anisotropy and the magnetic susceptibility anisotropy has been revealed for samples of metamorphic and sedimentary rocks. The existence of this correlation points to the same origin for both magnetic and acoustic anisotropy. The origin is the rock texture created during long tectonic history of the rock samples analyzed. The value of acoustic anisotropy allows evaluating the rock fracturing or the amplitude of tectonic stresses. Under additional calibration, it makes possible the use of the magnetic susceptibility value as a measure of tectonic processes intensity. The latter can be very useful to create express diagnostics of tectonic history in the field conditions.

Session 4aPP

Psychological and Physiological Acoustics: Hearing Impairment, Hearing Aids, and Cochlear Implants

Robert P. Carlyon, Chair
MRC Cognition and Brain Science Unit, 15 Chaucer Rd., Cambridge CB2 2EF, UK

Contributed Papers

9:00

4aPP1. Pitch comparisons between cochlear-implant stimulation and sounds played to a normal-hearing contralateral ear. Robert P. Carlyon, Olivier Macherey (MRC Cognition & Brain Sci. Unit, 15 Chaucer Rd., Cambridge CB2 7EF, England, bob.carlyon@mrc-cbu.cam.ac.uk), Johan H. M. Frijns (Univ. Leiden, The Netherlands), Patrick Axon (Addenbrookes NHS Trust, Cambridge, England), Randy K. Kalkman (Univ. Leiden, The Netherlands), Patrick Boyle (Adv. Bionics, Great Shelford, Cambridge, England), David M. Baguley (Addenbrookes NHS Trust, Cambridge, England), John Briggs (Addenbrookes NHS Trust, Cambridge, England), John Deeks (MRC Cognition & Brain Sci. Unit, 15 Chaucer Rd., Cambridge CB2 7EF, England), Jeroen J. Briaire (Univ. Leiden, The Netherlands), Xavier Barreau, and Rene Dauman (Hopital Pelegrin, Bordeaux, France)

A small group of cochlear implant users, having normal hearing in the unimplanted ear, compared the pitches of electrical and acoustic stimuli presented to the two ears. Comparisons were between 1031-pps pulse trains and pure tones or between 12- or 25-pps electric pulse trains and bandpass filtered acoustic pulse trains of the same rate. Three methods (pitch adjustment, constant stimuli, and interleaved adaptive procedures) were used. For all methods, we showed that the results can be strongly influenced by nonsensory biases arising from the range of acoustic stimuli presented, and proposed a series of checks that should be made to alert the experimenter to those biases. We then showed that the results of comparisons that survived these checks do not deviate consistently from the predictions of a widely used cochlear frequency-to-place formula or of a computational cochlear model. In one case, the matches were reliable enough to successfully reveal the movement (slippage) of the electrode array. We also demonstrate that substantial range effects occur with other widely used experimental methods, even for normal-hearing listeners.

9:15

4aPP2. Under what conditions can non-uniformly spaced stimulation enhance auditory sensitivity? An investigation using time-resetting maps. David E. O'Gorman and H. Steven Colburn (Hearing Res. Ctr., Dept. of Biomedical Eng., Boston Univ., 44 Cummington St., MA 02115, ogorman@bu.edu)

Most cochlear implants excite auditory nerve fibers with electric current pulses that are uniformly spaced in time. Yet, studies of the vesicle release properties of the ribbon synapses present at the inner hair cell show that temporal spacing of vesicular release is not uniform in time. It is possible that this non-uniform excitation contributes to the high sensitivity and temporal responsiveness of the healthy auditory system and that restoring this feature to the deafened nerve via electrical stimulation would be beneficial. This study uses time-resetting maps, computed from Hodgkin–Huxley-type models, to analyze the conditions under which a non-uniform periodic pulse train, consisting of an alternating sequence of short and long intervals, produces higher modulation sensitivity than a uniform pulse train of the same average rate.

9:30

4aPP3. Acquisition of spatial hearing abilities in two-year-old children: Role of auditory experience and bilateral cochlear implantation. Ruth Litovsky, Samantha Harris, and Melissa Born (Waisman Ctr., 1500 Highland Ave., Univ. of Wisconsin, Madison, WI 53705)

Bilateral cochlear implants (CIs) are being provided to a growing number of young children to improve spatial hearing and speech understanding in noise. While outcomes and benefits in older bilateral CI users have been documented, little is known about spatial hearing in 2–3 year olds. We studied sound localization abilities in two groups of children: normal hearing children and deaf children, who are fitted with bilateral cochlear implants. A novel task was implemented, whereby children were trained to reach for sounding objects that are hidden behind acoustically transparent curtains as a means of indicating where they perceived the sound to be coming from. Loudspeakers were positioned at ±60, ±45, ±30, ±15, and 0 deg in the horizontal plane. Two tasks were used: (1) left versus right discrimination for each pair and (2) sound localization using the 9-loudspeaker array. Results will be presented for conditions in which stimulus levels were either fixed at 60 dB sound pressure level or roved to minimize monaural level cues. In addition, the effect of auditory experience on performance will be shown for the CI users. Results consist of sound localization accuracy, quantified by the deviation of the reach from the actual source location.

9:45

4aPP4. Source segregation in noisy environments by children with normal hearing and bilateral cochlear implants. Ruth Litovsky, Sara Misurelli, and Shelly Godar (Waisman Ctr., Univ. of Wisconsin, 1500 Highland Ave., Madison, WI 53705, litovsky@waisman.wisc.edu)

Spatial release from making (SRM) refers to the improvement in speech intelligibility measured when target and maskers are spatially separated, as opposed to when they are co-located. SRM is affected by numerous factors that are of interest in this study. Asymmetrical placement of maskers to one side of the head leads to SRM as large as 12 dB in normal-hearing (NH) adults, due to a combination of monaural head-shadow and binaural interaction effects. In addition, SRM is affected by the content of the masker and its similarity to the talker. To better study these effects in children, we recently measured SRM when maskers were either symmetrically or asymmetrically distributed in the horizontal plane to the right and left. In addition, the sex of the talker and thus its similarity to the masker were varied. In the NH groups, children between the ages of 4–6 and 7–9 years and adults were tested. In addition, children who are deaf and use bilateral cochlear implants, also 4–6 and 7–9 years of age, participated in many of the conditions. Results will be discussed in the context of developmentally emerging and lacking abilities to utilize binaural cues for spatial unmasking.

10:00

4aPP5. Effect of acoustic-phonetic and semantic enhancements on speech recognition for children with cochlear implants. Rajka Smiljanic (Linguist., Univ. of Texas at Austin, Calhoun Hall 407, 1 University Station B5100, Austin, TX 78712-0198) and Douglas Sladen (Commun. Sci. and Disord., Univ. of Texas at Austin, Austin, TX 78712)

With early implantation and intense rehabilitation, children with cochlear implants are able to develop high levels of speech understanding in quiet. However, speech recognition in noise remains a challenging task for this listener group. In this study, we examine how signal clarity interacts with the use of compensatory information at higher levels in determining speech recognition levels for children with cochlear implants and children with normal hearing. Specifically, we explore the effect of clear speech, a distinct, intelligibility-enhancing speaking style, and of contextual-semantic

information on speech recognition in noise. One hundred and twenty sentences in which final word varied in predictability, i.e., high versus low semantic context, and which were produced in conversational and clear speech by one female and one male talker were embedded in noise. Twenty children between 5 and 12 years of age participated in a sentence-in-noise perception test. The goal was to examine the level of signal clarity needed to take advantage of the contextual information for children with cochlear implants compared to children with normal hearing. This research will allow us to explore the interaction between lower-level sensory and higher-level cognitive factors that affect speech processing in these two target groups of children.

10:15—10:30 Break

10:30

4aPP6. Duration of point vowels in four- and five year olds with hearing aids and cochlear implants. Mark VanDam, Dana Ide-Helvie, and Mary Pat Moeller (Ctr. for Childhood Deafness, Boys Town Natl. Res. Hosp., 555 N 30th St., Omaha, NE 68131)

Advances in hearing science have dramatically reduced the age of identification of hearing loss (HL) and improved intervention strategies, especially amplification methods. As a result, children with HL have earlier, better auditory access. How this early auditory experience affects language and speech abilities is not fully understood. This study addresses one aspect of speech production looking closely at point vowel duration. Although typically longer in children with HL and younger children, vowel duration has not been examined in early identified children with HL, and possible influence of device type (cochlear implant or hearing aids) is unclear. This study examines point vowel duration at four- and five-year-olds among children with normal hearing, cochlear implants, and hearing aids. Children produced /æ, ɑ, i, u/ in words modeled by the experimenter in a live-voice, listen-and-repeat format. As expected, results indicate that children with HL and younger children produced longer vowels. Results also indicate that effects are likely driven by high vowels in all children, and children with cochlear implants and hearing aids perform similarly, despite less auditory experience in the group with cochlear implants. Results have implications for developmental models and possible clinical applications. [Work supported by NIH /NIDCD Grant Nos. T32-DC00013 and R01-DC006681.]

10:45

4aPP7. A large-scale study on meaning-oriented auditory training with single versus multiple talkers. Nancy Tye-Murray (CID at Washington Univ. School of Medicine, 825 S. Taylor Ave., St. Louis, MO 63110, MurrayN@ent.wustl.edu), Mitchell Sommers, and Joe Barcroft (Washington Univ., St. Louis, MO 63130)

In their research review, Sweetow and Palmer (2002) concluded that the question of whether auditory training can be effective remains largely unanswered, primarily due to numerous methodological flaws in previous investigations. In light of this situation, we initiated a large-scale study to assess the benefits of auditory training while controlling for many of the methodological limitations pointed out by Sweetow and Palmer. The study also was designed to assess the effectiveness of using activities that are meaning-oriented while comparing the impact of two versions of the program—single-talker and multiple-talker. Participants include both hearing-aid users and cochlear implant recipients. Participants complete 12 1-h lessons during training. Each lesson includes spoken input at a variety of levels of linguistic analysis, including word, sentence, and discourse. Data will be presented for the two test groups. Analyses of both traditional (e.g., phoneme and word discrimination) and novel (perceptual effort) assessments indicate that the auditory training program is effective. Initial comparison of the single- and multiple-talker training conditions suggest that for the multiple-talker testing conditions, which most closely simulate real-world speech perception demands, training with multiple talkers is generally better than single-talker training.

11:00

4aPP8. Effects of reducing speech audibility on signal-to-noise-ratio loss for hearing-impaired listeners. Peggy B. Nelson, Yingjiu Nie, Elizabeth Crump Anderson, and Bhagyashree Katare (Dept. of Speech-Lang.-Hearing Sci., Univ. of Minnesota, 164 Pillsbury Dr. SE, Minneapolis, MN 55455, peggynelson@umn.edu)

Listeners with sensorineural hearing loss show reduced benefit from fluctuating compared to stationary maskers and experience apparent signal-to-noise-ratio loss when compared to listeners with normal hearing. Previously [Nelson *et al.* ASA Baltimore (2010)] we tested normal-hearing and hearing-impaired listeners at similar reduced audibility levels. Listeners with normal hearing and hearing loss were presented IEEE sentences at a range of overall levels (from 30 to 80 dB sound pressure level), signal-to-noise ratios, and low-pass filter settings, resulting in a range of signal AIs that varied from 0.1 to 0.95. For the normal-hearing listeners, there was a very systematic relationship between AI and performance; the relationship between AI and performance was less systematic for listeners with hearing loss. In the current project we tested each listener over a wide range of AIs and report the change in performance with increasing AI for individual listeners with normal hearing and hearing loss. Implications for hearing aids will be described. [Work supported by NIDCD R01-DC008306.]

11:15

4aPP9. A lumped parameter mechanical model of tensor tympani muscle contraction of the middle ear. Philip P. Garland, Fawaz M. Makki, Ross W. Deas, Robert B. Adamson, Jeremy A. Brown, and Manohar L. Bance (SENSE Lab., Dalhousie Univ., 1276 South Park St., Rm. 3189, Halifax, NS, Canada)

The role of the tensor tympani (TT) muscle of the middle ear is not well understood, and there is a long history of the implied, but unproven, part it plays in various inner ear disorders, particularly Meniere's disease. In order to gain an improved understanding of the effect of TT contraction, a lumped parameter mechanical model of the middle ear including the TT has been developed. This model uses a previously developed lumped parameter model of the middle ear ossicular chain along with experimentally obtained visco-elastic material model for the TT in order to predict the changes in the acoustic impedance of the middle ear experienced when the TT is contracted. Qualitatively, the results of the computer model agree quite well to similar laser Doppler vibrometer measurements from various cadaveric temporal bones where TT contraction has been simulated using force loading.

11:30

4aPP10. Middle ear reflectance in various middle disease states. Manohar Bance, Phillip Garland, Adamson Robert, and Brown Jeremy (3184 Dickson Bldng., VGH Site, QEII HSC, 1278 Tower Rd., Halifax, NS B3H 2Y9, Canada)

Many middle ear conditions are difficult to diagnose based on inspection of the eardrum, tympanometry, and audiometry. Most appear as a simple conductive hearing loss. We report findings using wideband middle ear reflectance measurements in a range of middle ear and inner ear conditions, particularly looking for evidence of middle ear involvement in some traditionally inner ear conditions such as Meniere's disease.

Session 4aSA

Structural Acoustics and Vibration, Engineering Acoustics and Underwater Acoustics: Acoustic Metamaterials I

Michael R. Haberman, Cochair
Applied Research Labs., Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758

Stephen D. O'Regan, Cochair
NSWC Carderock Div., 9500 MacArthur Blvd., West Bethesda, MD 20874

Thomas R. Howarth, Cochair
NAVSEA Newport, 1176 Howell St., Newport, RI 02841

Chair's Introduction—7:55

Invited Papers

8:00

4aSA1. Transformation acoustics. Steven A. Cummer (Dept. of Elec. and Comput. Eng., Duke Univ., P.O. Box 90291, Durham, NC 27708, cummer@ee.duke.edu)

It is now well understood how coordinate transformations of the Maxwell equations can be interpreted in terms of an electromagnetic material in the original coordinates with transformed values of permittivity and permeability. Through this transformation optics approach, the bending and stretching of electromagnetic fields specified by coordinate transformations can be implemented with electromagnetic materials, enabling unexpected and interesting solutions such as electromagnetic cloaking. In this presentation we describe work by us and others that show how this same concept can be extended to realize arbitrary manipulations of acoustic waves, including cloaking, using materials with very specific properties. While the concept of transformation acoustics was first demonstrated by analogy with electromagnetics, here we derive it from first principles through an analysis of the divergence and gradient operators under coordinate transformations. We will also discuss several approaches for engineering composite materials with the acoustic properties needed to realize transformation acoustics devices and demonstrate their performance through full wave simulations.

8:30

4aSA2. Transformation acoustics and elastodynamics. Andrew Norris (Mech. and Aerosp. Engng., Rutgers Univ., Piscataway, NJ 08854, norris@rutgers.edu)

The properties associated with acoustic metamaterials are usually related to the idea of spatial transformation. The acoustic magnifying hyperlens [Li *et al.* (2009)] and the acoustic cloak are two notable examples. The basic premise is that spatial deformation can be accommodated by changes in the material properties. This is fine as long as the material properties are mapped into new ones that are physically realistic. Often, however, the transformed material no longer lies within the original class of constitutive relations of linear acoustics. This talk will describe how acoustics and elastodynamics transform to produce new material behavior. The transformed acoustic material is not unique and can be, at one extreme, an anisotropic inertial fluid and, at the other, a pentamode material. Transformed elastic materials are even more general in their constitutive behavior. However, for *SH* waves in elasticity, the transformation implies another elastic material—in this case, just an orthotropic material. These ideas will be illustrated by application to cloaking of acoustic and elastic waves.

9:00

4aSA3. Acoustic metamaterials based on phononic crystals. J. Sanchez-Dehesa and D. Torrent (Dept. of Electron. Eng., Univ. Politécnica de Valencia, Camino de vera s.n., ES-46022 Valencia, Spain, jsdehesa@upvnet.upv.es)

This talk will review recent results on the topic of sound propagation through periodic arrays of sonic scatterers embedded in a fluid or a gas. These structures, which are called phononic crystals, define in the homogenization limit (i.e., for lattice separations much lower than the sound wavelength) a class of acoustic metamaterials with extraordinary properties. For example, they are solid structures that are used to engineer effective fluid-like systems with anisotropic dynamical mass density, a property that can be tailored by changing the parameters of the structure and the material composition of scatterers. A homogenization method based on multiple scattering theory has been developed and semianalytical expressions for the effective acoustic parameters will be given. By using these expressions it is easy to design novel refractive devices like gradient index sonic lens with a perfect match of impedance with the background. Results for a gradient index lens made of aluminum cylinders in air will be presented. Moreover, metamaterials with mass anisotropic have also been proposed to build broadband acoustic cloaks and a new type of crystals named radial sonic crystals, which are radially periodic structures that verify the Bloch theorem. [Work supported by ONR and by MICIIN of Spain.]

9:30

4aSA4. Active acoustic metamaterials. Amr Baz (Mech. Eng. Dept., Univ. of Maryland, College Park, MD 20742)

Extensive efforts are being exerted to develop various types of acoustic metamaterials to effectively control the flow of acoustical energy through these materials. However, all these efforts are focused on passive metamaterials with fixed material properties. The emphasis is placed here on the development of a class of acoustic metamaterials with tunable effective densities and bulk modulus in an attempt to enable the control of wave propagation. More importantly, the active metamaterials can be tailored to have increasing or decreasing variation of the material properties along and across the material volume. With such unique capabilities, physically realizable acoustic cloaks can be achieved and objects treated with these active metamaterials can become acoustically invisible. The theoretical analysis of this class of active acoustic metamaterials will be presented and the theoretical predictions are determined for an array of fluid cavities separated by piezoelectric boundaries. These boundaries control the stiffness of the individual cavity and in turn its dynamical density and bulk modulus. Various configurations will be considered to achieve different spectral and spatial control of the density and bulk modulus of this class of acoustic metamaterials. Extensive efforts are now exerted to build and test modules of these active acoustic metamaterials in order to build practical configurations of acoustic cloaks.

10:00—10:15 Break

Contributed Papers

10:15

4aSA5. Transformation acoustics: Theory, ray-tracing, and finite element simulations. Nachiket Gokhale, Jeffrey Cipolla (Weidlinger Assoc., Inc., 375 Hudson St., New York, NY), and Andrew Norris (Rutgers Univ.)

Norris' theory of transformation acoustics enables the realization of pentamode acoustic materials having anisotropic density and finite mass. Two additional extensions of the theory are considered, with a view toward demonstrating feasibility for practical applications. First, specializations of the theory developed by Norris ["Acoustic cloaking theory," Proc. R. Soc. London, Ser. A **464**, 2411–2434 (2008)] are considered in order to develop transformations corresponding to pentamode acoustic metamaterials with specified functional forms (constant or powerlaw) for density and stiffness. Ray-tracing and finite element simulations of wave propagation through such pentamode acoustic media are presented. Next, the design of acoustic materials for objects composed of simple geometric shapes (e.g., a cylinder with spherical endcaps) is considered. It is shown that certain classes of transformations are well-suited for the design of such shapes and validate the concept with finite element simulations.

10:30

4aSA6. Constrained negative stiffness metamaterials. Lia B. Kashdan, Michael R. Haberman, Preston S. Wilson, and Carolyn Conner Seepersad (Appl. Res. Labs. and Dept. of Mech. Eng., The Univ. of Texas at Austin, Austin, TX 78712)

Acoustics-oriented metamaterial research has focused primarily on the use of subwavelength resonant structures to elicit unconventional macroscale properties such as negative mass, anisotropic density, and negative stiffness. Unfortunately, reliance on resonant behavior limits the usefulness of these types of metamaterials to a narrow band of frequencies. One intriguing manifestation of negative stiffness, however, does not require resonant behavior and is therefore useful over a broader range of frequencies. This type of negative stiffness relies on constrained bistable structures and is known as constrained negative stiffness (CNS). The present work investigates the use of CNS mechanisms for vibration isolation and acoustic wave attenuation. A linearized analytical model of a constrained buckled beam structure is presented to predict system behavior and stability, and investigate the fundamental physical behavior leading to increased energy absorption. Experimental studies are then shown confirming model predictions. Finally, a 1-D candidate acoustical metamaterial containing CNS inclusions is modeled and analyzed.

10:45

4aSA7. Design and experimental evaluation of particulate composite materials for use as acoustic scattering cancellation layers. Matthew D. Guild, Michael R. Haberman (Appl. Res. Labs. and Dept. of Mech. Eng., The Univ. of Texas at Austin, Austin, TX 78712), and Andrea Alù (Dept. of Elec. and Comput. Eng., The Univ. of Texas at Austin, Austin, TX 78712)

The possibility of creating acoustic cloaks has recently received significant attention in the scientific community. Many authors have suggested the use of resonant metamaterial layers to achieve functionally graded, anisotropic, and extreme values of density and compressibility to effectively redirect acoustic waves around an object. The need for materials with these exotic properties stems from the use of the coordinate transformation method to design the cloaking layer. An alternative approach to acoustic cloaking, known as the scattering cancellation technique, can be taken in which an elastic cloaking layer is designed to significantly reduce the scattered field over a given bandwidth. Previous work [M. D. Guild, A. Alù, and M. R. Haberman, J. Acoust. Soc. Am. **127**, 1952 (2010)] has demonstrated the feasibility of this approach when applied to the case of an isotropic sphere coated with a single isotropic elastic cloaking layer. The current work presents the use of micromechanical models to design multiscale particulate composite materials consisting of micro- and mesoscale inclusions embedded in a continuous matrix. Polymer matrix samples containing varying inclusion volume fractions were constructed and tested to determine the bulk elastic properties using compressional and shear wave time-of-flight measurements. Experimental results validate the use of micromechanical modeling to design acoustic cloaking layers.

11:00

4aSA8. Anisotropic inertial acoustic metamaterials for use in aqueous environments. Theodore Martin, Michael Nicholas, Gregory Orris (Naval Res. Lab., Washington, DC), Daniel Torrent, and José Sanchez-Dehesa (Universidad Politéchnica de Valencia, C/Camino de vera s.n., E-46022 Valencia, Spain)

The concept of using multiple scattering to create an effective anisotropic mass density/bulk moduli for a composite inertial metamaterial has been around for a few years. Designs for use in aqueous environments face several distinct challenges in comparison to designs useful for airborne applications. The main difficulties lie in that the types of sturdy material components with which an inertial metamaterial can be created are effectively limited to relative impedances of less than a few tens of times that of water. Simply put, the density of water and sound speed comprising the background wave propagation media are much larger than their gaseous counterparts, creating an upper bound on such systems. Further complications can arise from the fact that approximations typically used to describe the scattering elements in homogenized fluid-like systems for airborne applications can become inaccurate for aqueous environments. Within this context, we present the current state of our experimental verification and validation of a multiple scattering based gradient index lens constructed for frequencies lower than 20 kHz, as well as theoretical/numerical investigation into this and related phenomenon for an orthotropic metamaterial system in an aqueous environment and consider the implications on more complex 2- and 3-D inertial metamaterial systems for use therein. [Work supported by the Office of Naval Research.]

11:15

4aSA9. Experimental evaluation of membrane type acoustic metamaterial arrays. Christina J. Naify (Dept. of Mater. Sci., Univ. of Southern California, 3651 Watt Way VHE 416, Los Angeles, CA 90089), Chia-Ming Chang, Geoffery McKnight (HRL Labs., Malibu, CA 90265-4797), and Steve Nutt (Univ. of Southern California, Los Angeles, CA 90089)

Traditional acoustic treatments such as foam and fiber panels used for sound insulation in transportation applications typically perform poorly at low frequencies without costly addition of mass. Locally resonant acoustic metamaterials have demonstrated a single transmission loss (TL) peak of up to 500% over the acoustic mass law at frequencies as low as 100 Hz with minimal weight penalty. The frequency of the TL peak was tuned by adjusting various geometric properties of the structure including membrane material, mass magnitude, and membrane size. Limits to the practical applications of the single-cell structure resulted in the development of panels comprised of multiple mass-weighted membranes. Arrays of membrane-type metamaterials were fabricated and evaluated to examine their transmission loss and dynamic response under acoustic loading. Large membranes were divided using support frames to create individual cells of the metamaterial. Multiple TL peaks were achieved for the arrays by varying the mass magnitude attached to each cell of the array. In addition, membrane material variation kept the TL peak at low frequencies with reduction in the size of each cell in the array. Finally, dynamic analysis using a laser vibrometer was performed to understand the interaction between cells under acoustic loading.

THURSDAY MORNING, 18 NOVEMBER 2010 GRAND CORAL 1A, 9:00 TO 11:50 A.M.

Session 4aSC

Speech Communication: Production and Perception of Spontaneous Speech I

Ann R. Bradlow, Cochair
Dept. of Linguistics, Northwestern Univ., 2016 Sheridan Rd., Evanston, IL 60208

Valerie L. Hazan, Cochair
Dept. of Phonetics and Linguistics, University College London, Gower St., London WC1N 1PF, U.K.

Chair's Introduction—9:00

Invited Papers

9:05

4aSC1. Prosody production in spontaneous speech: Phonological encoding, phonetic variability, and the prosodic signature of individual speakers. Jennifer Cole (Dept. of Linguist., Univ. of Illinois, 707 South Mathews, Urbana, IL 61801), Stefanie Shattuck-Hufnagel (MIT, Cambridge, MA 02139), and Yoonsook Mo (Univ. of Illinois, Urbana, IL 61801)

We examine the prosodic variation in production with corpus and experimental speech materials. Prosodic transcription of spontaneous speech from the Buckeye corpus of American English (38 speakers, 54 excerpts, 11–55-s duration each) reveals striking inter-speaker variation in the frequency and distribution of prosodic prominences and boundaries, and in acoustic correlates (pitch, intensity, duration, and spectral measures). Inter-transcriber agreement rates also vary systematically across speakers, suggesting inter-speaker differences in the clarity/consistency of prosodic cues. To further explore variability in the phonological and phonetic expression of prosody while holding lexico-syntactic content constant across speakers, we conducted an auditory repetition experiment. Ten American English speakers listened to 32 excerpts (8–15 words each, 4 speakers) from the American English Map Task corpus and reproduced each utterance with the exact words and "in the way the speaker said them", without text prompts. Preliminary results from prosodic transcription and acoustic analysis show reliable replication of the phonological structures locating prosodic prominences and phrase boundaries, but with variation in the pitch melody and other phonetic details of the prosodic features. These findings shed light on the mapping between the phonological encoding and the acoustic expression of prosodic features, and highlight those acoustic parameters that identify the prosodic signature of individual speakers.

9:35

4aSC2. The Edinburgh Speech Production Facility's articulatory corpus of spontaneous dialogue. Alice Turk (Ling. & Eng. Lang., Univ. of Edinburgh, Dugald Stewart Bldg., 3 Charles St., Edinburgh EH8 9AD, Scotland, UK, turk@ling.ed.ac.uk), James Scobbie (Queen Margaret Univ., Edinburgh, Scotland, UK), Christian Geng, Cedric Macmartin, Ellen Bard, Barry Campbell, Catherine Dickie, Eddie Dubourg (Univ. of Edinburgh, Edinburgh, Scotland, UK), Bill Hardcastle (Queen Margaret Univ., Edinburgh, Scotland, UK), Phil Hoole (Ludwig-Maximilians-Univ., Munich, Germany), Evia Kanaida (Univ. of Edinburgh, Edinburgh, Scotland, UK), Robin Lickley (Queen Margaret Univ., Edinburgh, Scotland, UK), Satsuki Nakai (Univ. of Edinburgh, Edinburgh, Scotland, UK), Marianne Pouplier (Ludwig-Maximilians-Univ., Munich, Germany), Simon King, Steve Renals, Korin Richmond (Univ. of Edinburgh, Edinburgh, Scotland, UK), Sonja Schaeffler (Queen Margaret Univ., Edinburgh, Scotland, UK), Ronnie Wiegand, Kevin White (Univ. of Edinburgh, Edinburgh, Scotland, UK), and Alan Wrench (Articulate Instruments Ltd., Edinburgh, Scotland, UK)

The EPSRC-funded Edinburgh Speech Production is built around two synchronized Carstens AG500 electromagnetic articulographs (EMAs) in order to capture articulatory/acoustic data from spontaneous dialogue. An initial articulatory corpus was designed with two aims. The first was to elicit a range of speech styles/registers from speakers, and therefore provide an alternative to fully scripted corpora. The second was to extend the corpus beyond monologue, by using tasks that promote natural discourse and interaction. A subsidiary driver was to use dialects from outwith North America: dialogues paired up a Scottish English and a Southern British English speaker. Tasks. Monologue: Story reading of "Comma Gets a Cure" [Honorof *et al.* (2000)], lexical sets [Wells (1982)], spontaneous story telling, diadochokinetic tasks. Dialogue: Map tasks [Anderson *et al.* (1991)], "Spot the Difference" picture tasks [Bradlow *et al.* (2007)], story-recall. Shadowing of the spontaneous story telling by the second participant. Each dialogue session includes approximately 30 min of speech, and there are acoustics-only baseline materials. We will introduce the corpus and highlight the role of articulatory production data in helping provide a fuller understanding of various spontaneous speech phenomena by presenting examples of naturally occurring covert speech errors, accent accommodation, turn taking negotiation, and shadowing.

10:05—10:20 Break

10:20

4aSC3. Perception of reduced pronunciation variants in conversational speech. Mirjam Ernestus (Radboud Univ. Nijmegen, P.O. Box 310, NL-6500 AH Nijmegen, The Netherlands)

Spontaneous conversations are characterized by reduced pronunciation variants that, relative to the corresponding citation variants, provide less acoustic information (e.g., the English word "hilarious" may be pronounced like [hlErEs]). This paper discusses three series of experiments investigating how listeners process reduced variants within their natural contexts. In two series of experiments, native speakers of Northern Dutch listened to extracts from spontaneous conversations in Northern Dutch and either predicted masked reduced words or repeated stretches of speech. These experiments show that even though reduced variants contain little acoustic information, listeners heavily rely on this information. Furthermore, listeners use acoustic cues in the context, which is often reduced itself, and these cues may override semantic/syntactic cues. In the third series of experiments, native speakers of Northern and Southern Dutch listened to conversations in Northern Dutch and indicated whether words appearing on the screen had just occurred in the conversation. The results show that acoustic cues are less informative for listeners who are less familiar with the language variant. These listeners rely more on semantic/syntactic information. In conclusion, acoustic cues, even if highly reduced, play a major role in the processing of conversational speech, especially in native listeners.

10:50

4aSC4. Clarifications in spontaneous speech under three different adverse communicative situations. Rachel Baker and Valerie Hazan (UCL Speech Hearing and Phonetic Sci., Chandler House, 2 Wakefield St., London WC1N 1PF, rachel.baker@ucl.ac.uk)

Difficult communicative situations requiring clear speech are common. This study investigated whether (a) clear speech elicited by instructing talkers to read sentences clearly reflects clarifications made by talkers in spontaneous speech with communicative intent and (b) talkers modulate their clear speech according to the type of adversity under which communication takes place. Casual and clear speech from 40 Southern British English talkers was elicited in dialogues recorded while two talkers engaged in "spot the difference" picture tasks, based on the Diapix task of Bradlow and collaborators. Three types of "communication barrier" were used to elicit clear speech: in the VOC condition, one talker heard the other via a three-channel vocoder; in the NOISE condition, one person heard the other with simultaneous babble noise; in the L2 condition one talker was a low-proficiency non-native speaker. Read clear speech showed more extreme changes in median F0, F0 range and speaking rate than spontaneous clear speech. Greater changes were made to speaking rate and vowel space in the VOC than the L2 condition, while greater changes in F0 and mean energy were made in the NOISE than the VOC condition, suggesting that talkers clarify their speech according to listeners' needs. [Work funded by ESRC project no. RES-062-23-0681.]

11:20

4aSC5. Considering the storage, management, and processing of spontaneous speech corpora: New methods and new findings. Tyler Kendall (Linguist. Dept., Northwestern Univ., 2016 Sheridan Rd., Evanston, IL 60208, t-kendall@northwestern.edu)

Recordings of speech, both spontaneous and nonspontaneous, comprise the backbone of acoustic phonetic, speech science, and much general linguistic research. The creation of these speech recordings (e.g., recording equipment and its use) and their analysis (e.g., appropriate statistical methods) tend to be thoroughly treated in methodology sections of papers and in specialized methodology textbooks. However, the storage, management, and preservation of these resources are rarely discussed in the academic literature, though these practices influence both the short-term and long-term usability of these resources. This talk addresses the related questions of how we might best manage large and growing collections of audio data and how we can leverage new technologies so that our data archives are not just usable, but maximally useful. Examples are provided from two web-based archiving projects, the Sociolinguistic Archive and Analysis Project (SLAAP) and the Online Speech/Corpora Archive and Analysis Resource, which feature organization and analysis tools that enhance the overall usefulness of the archived recordings. To exemplify some substantive outcomes of these projects, findings from a corpus sociophonetic study of speech rate and pause variation in American English, made possible by SLAAP's software, are also discussed.

Session 4aUW

Underwater Acoustics and Signal Processing in Acoustics: Physics-Based Signal Characterization, Classification, and Processing I

Timothy K. Stanton, Cochair
Dept. of Applied Ocean Physics and Engineering, Woods Hole Oceanographic Inst., Woods Hole, MA 02543

R. Lee Culver, Cochair
Applied Research Lab., Pennsylvania State Univ., University Park, PA 16802

Invited Paper

8:00

4aUW1. An overview of physics-based signal and information processing in sonar. Douglas Abraham (CausaSci LLC, P.O. Box 5892, Arlington, VA 22205, abrahad@ieee.org)

Sonar systems are usually designed for the purposes of detection, classification, localization, and tracking (DCLT) objects under water. DCLT is accomplished through signal and information processing applied to the sonar's sensor data. Physics exploitable in sonar DCLT enter either through the object of interest's sound generation or scattering or through propagation from and to the sonar. In general, the former is exploited in classification and the latter in localization and tracking. In order to highlight key principles and issues in physics-based signal and information processing, examples of DCLT algorithms exploiting physics will be presented from each of the major processing tasks. There are a plethora of examples, ranging from those coincident to the unadulterated signal-processing approach to those enabling performance unachievable without exploitation of the specific physics. A metric of equal importance to performance is that of robustness to assumed models and information. Ideally, algorithms are designed to maximize both performance and robustness to uncertainty or inaccuracy. However, it is more common to see performance and robustness inversely related to each other. The talk will conclude with observations on potential open areas for exploiting physics in sonar DCLT. [Work sponsored by ONR under Contract no. N00014-09-C-0318.]

Contributed Papers

8:20

4aUW2. Propagation equations for non-stationary noise in waveguides. Leon Cohen (Dept. of Phys., City Univ.-Hunter, 695 Park Ave., New York, NY 10065)

We discuss how to transform wave equations into phase-space equations and apply the method to the time-varying autocorrelation function and to time-varying spectral distributions. We give a number of examples and argue that the phase-space equation is often more revealing than the original wave equation. We derive a successive approximation scheme for propagation in media that has dispersion and attenuation. The methods developed are applied to the propagation of noise fields in wave guides wherein we obtain relations for how the non-stationary autocorrelation function changes in position and time. We also show that the evolution of the statistics toward equilibrium can be seen in a much clearer way when the problem is formulated in phase-space. In addition, we show that one can obtain propagation equations for the amplitude and phase separately.

8:35

4aUW3. Potential for extraction of scatterer and waveguide characteristics from reverberation spectrograms. Kevin D. LePage (NATO Undersea Res. Ctr., Vle. San Bartolomeo 400, 19126 La Spezia, SP, Italy)

Coherent structure in the spectrograms of broadband reverberation in the mid frequency regime has been measured at various sites by various researchers. Recently, very broadband data collected over three and a half octaves on the Malta Plateau during the Clutter 09 experiment conducted by the NATO Undersea Research Centre show very clearly the striation structure of interest. In this paper, the theory of why the striations appear in the spectrograms of the reverberation is briefly reviewed, and the extraction of channel and scatterer information from the time-frequency content of the signal is explored using holomorphic processing. To constrain these studies, we exploit the availability of broadband reverberation models which contain all the features of the data, are noise free and have zero uncertainty, and have known waveguide and scatterer characteristics to compare with the extracted information. The Clutter 09 data are also evaluated using the method, and the resolution and robustness of the approach are discussed.

Invited Paper

8:50

4aUW4. Exploring the limits of matched-field processing. Claire Debever and W. A. Kuperman (Scripps Inst. of Oceanogr., UCSD, 9500 Gilman Dr., Mail Code 0238, La Jolla, CA 92093-0238, cdebever@ucsd.edu)

Though matched-field processing (MFP) has been shown to successfully localize acoustic sources in an assortment of ocean environments, it has a questionable overall reliability that is still not completely understood. The combination of factors such as signal-to-noise ratio, environment and propagation model accuracy, incoherent versus coherent frequency processing, and the time scales of ocean

processes and source/receiver/interferer motion all contribute to MFP performance. Examples of MFP experiments that include results that transition from successful to failed localization are presented and further examined using simulation. Conventional and adaptive methods are included in this study as well as both narrowband and broadband processing.

Contributed Papers

9:10

4aUW5. Sonar transmit and receiver design for detection of underwater objects in nonstationary environments. Brandon Hamschin and Patrick Loughlin (Dept. of Elec. and Comput. Eng., Univ. of Pittsburgh, 348 Benedum Hall, Pittsburgh, PA 15261, bmh161@gmail.com)

In undersea environments, particularly shallow water, the sonar backscatter from objects of interest can be subject to propagation effects such as dispersion, attenuation, and multi-path, which can confound detection and classification. Detection and classification of buried objects are further complicated by dramatic changes in the backscatter due to the sediment layer. These propagation and environmental effects can induce time-varying (or nonstationary) characteristics in the received sonar signal. In addition, the object itself can induce nonstationarities, such as the inherent dispersion characteristics of some elastic objects. The processing and analysis of such signals for detection and classification can be enhanced by applying time-varying methods to the received signal, such as time-frequency analysis and linear time-varying filters. Detection can also be enhanced by designing a transmit waveform to optimize some metric, such as received signal-to-interference/noise power. In this talk, we explore the use and benefits of optimum transmit waveform design, and time-varying processing at the receiver, for the detection of elastic objects from their sonar backscatter. [Work supported by ONR, code 321US.]

9:25

4aUW6. Physics-based performance prediction model for a coherent communications algorithm. Daniel Rouseff and Darrell R. Jackson (Appl. Phys. Lab., Univ. of Washington, Seattle, WA 98105)

A demodulation algorithm for coherent underwater acoustic communications may start by applying a finite impulse response (FIR) filter matched to the response of the channel. Relevant design parameters are the length of the FIR filter and the rate at which it must be updated. The performance of the algorithm as a function of these design parameters can then be quantified in terms of the mean-squared error (MSE) in the soft demodulation output. In terms of propagation physics, the length of the FIR filter is related to the number of acoustic paths retained as usable signal, and the required update rate is related to the time variation of the channel. In the present study, scattering by the sea surface is the assumed source of time variation. A model is developed for the MSE as a function of the design parameters. The Kirchhoff approximation is used to model acoustic paths that may have undergone multiple reflections by the sea surface. Required model inputs include the grazing angle, the significant wave height, and the dominant period for the surface waves. Model predictions are compared to published experimental results. [Work supported by ONR.]

9:40

4aUW7. Relating acoustic communication performance in different shallow water environments. T. C. Yang (Naval Res. Lab., Washington, DC 20375)

Relating the performance of an underwater communication system with the signal/environmental characteristics under different environmental conditions is a subject of great interest. Consider a very-shallow water channel, for example, with different wind speeds. The higher the wind speed, the more random the channel (due to sound-scattering from the rough surface), and hence the less the channel coherence-time and the higher the channel-estimation error. For such a channel, one expects the output signal-to-noise (SNR) to decrease inversely with the channel-estimation error; the latter reflects how much of the channel energy is random and treated like noise. On the other hand, the output SNR is expected to be unrelated between different environments where propagation conditions (the multipath arrivals) are significantly different. This paper shows, based on experimental data, that the output SNR (given the same input SNR) is correlated with the channel-estimation error with the same functional relationship despite the environmental differences. The reason is the universal property of the so-called Q function, the auto-correlation of the received impulse responses summed over all receiver channels. The channel-estimation error can thus serve as a metric for performance assessment in different shallow water environments. [Work supported by the Office of Naval Research.]

9:55

4aUW8. Source range and depth estimation using passive sonar, horizontal arrays, and knowledge of the environment. Brett E. Bissinger (Appl. Res. Lab and Dept. of Elec. Eng., Penn State Univ., State College, PA 16804), Colin W. Jemmott (Appl. Res. Lab and Grad. Program in Acoust., Penn State Univ., State College, PA 16804), and David J. Miller (Dept. of Elec. Eng., Penn State Univ., State College, PA 16804)

Target detection, classification, localization, and tracking (DCLT) using horizontal arrays and passive sonar is a well-studied problem. Good detection results can be obtained using narrow beams that maximize signal to noise ratio. Beamforming also provides azimuth angle, but is generally not reliable for range and depth estimation. Matched field processing (MFP) makes use of environmental knowledge to predict the received signal at the array, and correlates predicted and received signals to estimate target range and depth. However, poor or inaccurate environmental knowledge degrades MFP performance, and in general, MFP suffers from high side lobes or ambiguity in the range/depth probability surface. Received signal amplitude statistics have been used to estimate source depth, but the method has not been studied extensively to date. Recently several methods have been developed involving received signal amplitude statistics predicted using environmental parameter statistics to construct statistically valid descriptions of the environment, which were then fed to an ocean acoustic propagation model (i.e., Monte Carlo simulation). Performance of the algorithms has been evaluated using data from the Swellex-96 experiment. [Work sponsored by ONR Undersea Signal Processing.]

10:10—10:25 Break

10:25

4aUW9. Passive sonar classification using time-frequency domain "features". Alexander W. Sell and R. Lee Culver (Grad. Prog. in Acoust., Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, aws164@psu.edu)

Success in classifying passive sonar signals depends heavily on understanding the source characteristics and the physics of propagation and scattering in the ocean environment. Passive sonar signals are typically analyzed using bearing versus time records (BTRs) and time versus frequency displays (lofargrams). Much research has been devoted to studying the relationship between features in the time-frequency domain (i.e., lofargrams) and the range, depth, speed, and other aspects of an acoustic source. For example, striations in time-frequency domain are known to be related to broadband excitation of resonances in the shallow water waveguide. The slope of the striations can be predicted using the so-called waveguide invariant. Other waveguide-dependent resonance features can be predicted with some knowledge of the environment. While studying data from a September 2007 shallow water acoustic transmission test, performed along the continental shelf off the coast of southeast Florida, we noted a "bathtub phenomenon" in lofargrams encompassing the closest point of approach of two surface ships. The relationship between these features and the characteristics of the surface ship source is the subject of this talk. [Work supported by ONR Undersea Signal Processing.]

10:40

4aUW10. Choosing array processor weights for observing waveguide-invariant intensity striations. Kevin L. Cockrell, Henrik Schmidt (Dept. of Mech. Eng., MIT, Cambridge, MA), and James C. Preisig (Woods Hole Oceanograph. Inst., Woods Hole, MA 02543)

It is well known that a horizontal line array (HLA) can be used to observe a planewave signal from one source while rejecting noise from other sources. This talk discusses the selection of HLA processor weights when the goal is to observe waveguide invariant striations from one source while rejecting noise from other sources. It is shown that the array weights commonly used for planewave beamforming, such as uniform weights, can have the unintended effect of suppressing some the desired striations. Examples will be given demonstrating this effect. [Work supported by ONR N00014-08-1-0013]

Invited Papers

10:55

4aUW11. Low-frequency normal mode propagation in shallow water and its application to the acoustic inverse problem. Ying-Tsong Lin and James F. Lynch (Woods Hole Oceanograph. Inst., Woods Hole, MA 02543)

Acoustic normal mode theory is useful to describe low-frequency sound propagation in shallow water. In this talk, normal-mode propagation effects will be reviewed, as well as acoustic inverse approaches based on modal propagation theory. The normal modes of low frequency sound in shallow water are geometrically dispersive due to the presence of surface and bottom boundaries. One can exploit this modal dispersion to solve some acoustic inverse problems, such as passive acoustic localization and geoacoustic inversion. The approaches of matched field processing and backpropagation are commonly used. In addition, the modal wavenumber spectrum contains information on seabed properties that can be used to infer the geoacoustic parameters. Underwater sound propagation is strongly affected by water-column fluctuations, and internal waves are one of the significant effects in shallow water environment. An overview of internal wave effects on normal mode propagation will be presented here, along with their consequence on acoustic inversions.

11:15

4aUW12. Environmentally tolerant waveguide-invariant target depth classification for active sonar. Ryan Goldhahn, Peter Nielson, and Jeffrey Krolik (Dept. of Elec. and Comput. Eng., Duke Univ., Durham, NC 27708, jk@ee.duke.edu)

Shallow-water environments produce active sonar returns with many target-like returns from bottom clutter. Scatterer depth classification methods which can discriminate bottom clutter from water column targets are thus critical for controlling false alarms. In recent work, the waveguide invariant (WI) property of shallow-water channels has been used to obtain multiple snapshots of frequency-domain target return data from a single active sonar ping. These snapshots are, in turn, used to estimate a waveguide invariant spectral density matrix (WI-SDM), which can serve as a basis for depth classification. One method employing the WI-SDM performs minimum variance filtering (MVF) matched to depth-dependent signal replicas derived from a normal mode model. While MVF is capable of depth classification when the environment is known, it is sensitive to mismatch when the channel parameters are uncertain. In this paper, robustness of the WI MVF to mismatch is achieved by using environmental perturbation constraints derived from a WI-SDM for the signal averaged over the uncertain channel parameters. Simulated and real data results are presented from the CLUTTER-09 experiment in the Mediterranean Sea. [Work sponsored by ONR.]

11:35—12:00 Panel Discussion

Session 4pAAa

Architectural Acoustics and Noise: Sound Attenuation Through Ceiling Plenums

Kenneth P. Roy, Cochair
Armstrong World Industries, 2500 Columbia Ave., Lancaster, PA 17604

Sergio Beristain, Cochair
Mexican Inst. of Acoustics, P. O. Box 12-1022, Navarte 03001 DF Mexico City, Mexico

Chair's Introduction—1:00

Invited Papers

1:05

4pAAa1. Acoustic design of wall, ceiling, and plenum for college buildings. Kenneth P. Roy and Kenneth W. Good (Armstrong World Industries, 2500 Columbia Ave., Lancaster, PA 17604)

What can you do when an owner wants to design and build flexible learning spaces that can be reconfigured as the needs of the school teaching program changes? Well, we have done a research program on this issue and will share both laboratory and field testing data. Demountable wall systems, raised floors, deep ceiling plenums, stuff you stuff into the plenum, and high CAC ceiling tile. Does it make a difference if the teaching is live voice, recorded voice program, or how about music? Of course, it does. So let us talk sound transmission through the ceiling plenum and what to do about it.

1:25

4pAAa2. Ceiling plenum design: Laboratory testing of ceiling/plenum systems for ceiling attenuation class. Kenneth P. Roy and Kenneth W. Good (Armstrong, 2500 Columbia Ave., Lancaster, PA 17552)

Any acoustic testing laboratory that performs the ASTM E1414 testing of ceiling attenuation class ceiling system will have experimented with a few odd cases and learned a few interesting "rules of thumb." The Armstrong World Industries Acoustic Testing Laboratory in Lancaster, PA, has certainly had its share of those, and has learned a few things about materials and systems along the way. We would like to share these understanding about ceiling materials, grid systems, plenum stuff, etc. Test data…that is what it is all about.

1:45

4pAAa3. Sound transmission between rooms with a common plenum: Better predictions from laboratory data. Noral D. Stewart (Stewart Acoust. Consultants, 7330 Chapel Hill Rd., Ste. 101, Raleigh, NC 27607, asacancun@sacnc.com)

ASTM E1414 provides a method of measuring one-third-octave noise reduction between two prescribed laboratory spaces sharing a common ceiling plenum and calculating a single number rating based on those results normalized for a specific amount of receiving-room absorption. However, the noise reduction between two such spaces is dependent on many variables, and E1414 provides no way of relating the measured results to rooms of different geometries. This contrasts with the situation for sound passing through a wall between spaces where the noise reduction (without flanking) depends only on the wall transmission loss, wall area, and absorption in the receiving room. Mariner devised a theoretical equation for the noise reduction through the plenum, but it included the actual transmission loss of the ceiling panels. Starting with that work, a relationship is developed between the normalized noise reduction as measured in E1414 and the actual noise reduction through the ceiling plenum that also depends on the area of the opening between spaces in the plenum, the height and width of that opening, and the absorption in the plenum.

2:05

4pAAa4. Use of air pressure dampening in impact noise control. James Keene (Keene Bldg. Products, 5910 Landerbrook Dr., Ste 210, Mayfield Heights, OH 44124) and John LoVerde (Veneklasen Assoc., Santa Monica, CA 90404)

Since the 1980s, entangled net resilient mats have been used to effectively control impact noise in multi-family dwellings. The thickness of the resilient mat was modified to provide higher levels of acoustical isolation performance, and structural underlayments were used to instal over the resilient matting to provide sufficient support for finished flooring. Other products have been developed that work similarly, including rubber, cardboard, and insulation devices, varying the resilience, thickness, and airspace. Recently through engineering of what is commonly called a "high loft" fabric integrated into an entangled net resilient mat, a composite flooring product has been developed that, to date, changes the acoustical performance of standard thickness entangled net resilient mat devices to an enhanced level. This highly compressible fabric makes a 0.25-in. device resilient mat perform at a greater level than a typical 0.375-

in.-thick resilient mat device. The addition of this fabric allows for the consistent, luxury performance in one product with typical 1.0-in. underlayment systems. This presentation is an explanation of the physics development behind the design and provides the group with some preliminary testing of different materials objective comparative acoustical tests to show the differences between conventional resilient mattings and the new composite material.

2:25

4pAAa5. Improvements in impact noise insulation from suspended ceilings. John LoVerde and Wayland Dong (Veneklasen Assoc., 1711 16th St., Santa Monica, CA 90404, jloverde@veneklasen.com)

In high-rise multifamily projects, one of the primary building design decisions from the acoustical perspective is to determine whether or not to install a suspended ceiling at some distance below the structural slab. The acoustical benefits of a suspended ceiling include an improvement of the impact noise insulation of the assembly and reduced flanking transmission; negative aspects of the construction include lower ceiling height, greater building height, and increased initial building cost. The authors analyze field acoustical impact test results in high-rise projects, quantifying the improvements in impact insulation due to top-side resilient matting options and suspended ceiling designs. A summary is presented of the acoustical benefits of suspended ceiling systems compared with the exposed concrete slab condition. The goal of the analysis is to quantify the acoustical benefits of these constructions based on historic field testing to allow for more informed primary building design direction.

2:45

4pAAa6. Plenums in apartment buildings. Sergio Beristain (Lab. Acoust., ESIME, IPN, IMA, P.O. Box 12-1022, Narvarte, 03001 Mexico City, Mexico)

A construction company decided to build an apartment 10-story building with very little investment. There, dozens of small flats were built, and it was expected that solid floors were to be installed over most of the living dining and sleeping areas (with only a few carpeting in some bedroom areas); flats were delivered with a concrete flooring of 5 cm below final level, so owners could decide the type of floor. In order to reduce the high sound levels from furniture movement and women shoe steps in the above apartment, which were considered the biggest noise problems, the space between the concrete roof surface and the false ceiling was constructed as a plenum, as there was no other way to reduce those sounds. Some sound results and comments are presented.

THURSDAY AFTERNOON, 18 NOVEMBER 2010 GRAND CORAL 1B, 3:10 TO 4:30 P.M.

Session 4pAAb

Architectural Acoustics, Noise, and Engineering Acoustics: Mechanical Engineering and Acoustics

Scott D. Pfeiffer, Chair
Threshold Acoustics LLC, 53 West Jackson Blvd., Suite 815, Chicago, IL 60604

Chair's Introduction—3:10

Contributed Papers

3:15

4pAAb1. Predicting vibrational isolation efficiency using mobilities. Peter D'Antonio (RPG Diffusor Systems, Inc., 651-C Commerce Dr., Upper Marlboro, MD 20774, pdantonio@rpginc.com) and Michael Vanstraelen (CDM, Overijse B-3090, Belgium)

To solve complex vibrational problems, we have developed a predictive program using mobility, which is the inverse of the impedance. The analogy with electrical circuits, in which mobility corresponds to resistance, is used to solve mass-spring-damper systems. To determine the vibration isolation between two points, one must first determine the mass and complex stiffness mobilities and use Kirchoff's rules to define the equivalent mobility in each node. The force is then determined everywhere in the circuit, starting from the excitation force, and used to calculate the speed potential, using the mobility definition. A derivation of the isolation efficiency will be presented. Examples of a 1 degree of freedom system, in which you have a motor isolated from a rigid floor, and a 2 degree of freedom system, in which you have a motor isolated from a resilient floor, will also be given.

3:30

4pAAb2. Predicting isolation efficiency using the prognosis method. Peter D'Antonio (RPG Diffusor Systems, Inc., 651-C Commerce Dr., Upper Marlboro, MD 20774, pdantonio@rpginc.com) and Michael Vanstraelen (CDM, Overijse B-3090, Belgium)

A method will be described to estimate the transmission loss (TL) of a floating floor separated by an elastomer from a structural concrete floor, using the prognosis method. The system resonance is determined by the two masses and the combined stiffness of the elastomer and the constrained air. At low frequencies, the TL curve has a slope of 6 dB/octave based on the mass of the double system. At the resonant frequency (fr) of the system, the TL is strongly reduced. Above fr, the isolation increases rapidly at a rate of 18 dB/octave until the upper limit of 12 dB/octave is reached, due to the sum of the isolation of the two masses. Several examples of different floating floor options, using MS Excel software program, will be provided to illustrate the effects of the floor masses, the fr of the isolator, the stiffness of the constrained air layer, and flanking.

3:45

4pAAb3. An analytical study on active control and mechanisms of sound transmission through double panel-cavity system. Guoyong Jin, Zhigang Liu, Yuehua Chen, and Tiejun Yang (Dept. of Power and Energy Eng., Harbin Eng. Univ., Harbin 150001, China, jgy1822@yahoo.com.cn)

Combing the subsystem modal coupling theory and the impedance-mobility approach, a general analytical model is developed for the analysis and active control of structural acoustic transmission through the double panel-cavity structure into the free field, in which control methods including incident panel piezoelectric (PZT) actuator, acoustic actuator in cavity, and radiating panel PZT actuator are considered. The matrix formulae for the modal response of the coupled structural acoustic system and optimal secondary source strengths are obtained. On the basis, active control of sound transmission is studied analytically and numerically in detail. Simulations and analyzes are conducted to compare the control effects between different control strategies and examine the dominant control mechanisms involved in each control case, and the effect of system parameters on control effect is also examined. The results demonstrate that the incident panel actuator can obtain sound attenuation by modal suppression or rearrangement of three subsystems so many complex control mechanisms are involved and it is effective for incident panel modes, cavity modes, and radiating panel modes. Using the secondary acoustic source in the cavity is a better control strategy compared to the other control strategies due to the dominance of the acoustic mode (0,0,0) in the modal couplings and energy transmission in low frequency range.

4:00

4pAAb4. Control strategies and mechanisms for active control of sound transmission into a cabin enclosure. Guoyong Jin, Shuangxia Shi, and Zhigang Liu (Dept. of Power and Energy Eng., Harbin Eng. Univ., Harbin 150001, China, jgy1822@yahoo.com.cn)

An analytical study of active control of sound transmission into a cabin enclosure is presented. A cabin enclosure with four acoustically rigid walls and two flexible plates is considered as the analytical model. Two types of actuators are used, i.e., acoustic actuators and distributed lead zirconate titanate piezoelectric (PZT) actuators instead of point force actuators. Using the modal acoustic transfer impedance-mobility matrices, the excitation and interaction in the coupled sound transmission system can be described with clear physical significance. With the control system designed to globally reduce the sound field, different control configurations are considered, including the structural actuator on the incident panel, actuator on the receiving panel, acoustic actuator on the cavity, and their combinations. The effectiveness and performance of the control system corresponding to each configuration are compared and discussed. The roles and attenuation mechanisms of each type of actuator are of particular interest, and desirable placements of structural actuators in terms of total potential energy reduction are also discussed.

4:15

4pAAb5. Intelligent baffle with variable labyrinth. Carlos Barroeta, Floriberto R. Ortiz, and Juan Francisco Novoa Colin (ESIME, IPN, cbarroet @hotmail.com)

This is a sound system baffle with a variable window size. The system has an inner electronic circuit working for the acoustic compliance of the box. In low frequencies, the width of the window is the maximum and will change for upper frequencies, thus increasing the acoustic compliance in order to give a better quality in the sounds. The system begins to respond from 100 mA to 5 A. When fully open, the system is closed is at its maximum power.

THURSDAY AFTERNOON, 18 NOVEMBER 2010 — CORAL GARDEN 2/3, 1:00 TO 5:00 P.M.

Session 4pAB

Animal Bioacoustics: Animal Bioacoustics Metadata Workshop

Maria A. Roch, Chair
Dept. of Computer Science, San Diego State Univ., San Diego, CA 92182-7720

Chair's Introduction—1:00

Invited Papers

1:05

4pAB1. History of the Acoustical Society of America's standard on "Underwater Passive Acoustic Monitoring for Bioacoustic Applications": Metadata issues. Aaron Thode (Marine Physical Lab., Scripps Inst. of Oceanogr., UCSD, 9500 Gilman Dr., La Jolla, CA 92093-0238) and David K. Mellinger (Oregon State Univ., Newport, OR 97365)

In 2005 an ASA working group was formed to examine whether a standard should exist for "Passive Acoustic Monitoring for Marine Mammal Mitigation for Seismic Surveys". Public discussions at subsequent meetings quickly showed no consensus existed for specifying hardware requirements for passive acoustic measurements, but consensus did seem possible for specifying "minimum requirements for recording and reporting bioacoustic data." The proposed standard was renamed "Underwater Passive Acoustic Monitoring for Bioacoustic Applications" with three defined goals: (1) providing a set of requirements for information to be documented while recording acoustic data at sea (metadata requirements); (2) detailing the minimum information about acoustic hardware and software to be included when reporting results in gray or peer-reviewed literature; and (3) specifying metrics to be used when summarizing the features of an acoustic signal. After languishing for several years, an attempt is being made to develop the standard further. This presentation describes what ASA standards are, outlines the development process, and sketches the current status of the standard, with an emphasis on metadata issues.

1:25

4pAB2. Software tools for visual and acoustic real time tracking of marine mammals. Angela D'Amico, Christopher Kyburg, and Rowena Carlson (SPAWAR Systems Ctr. Pacific, San Diego, CA 92152, angela.damico@navy.mil)

The Whale Identification, Logging, and Display (WILD) software system was developed to provide researchers tools to integrate sightings of marine mammals by trained visual observers, detections by passive acoustic arrays and other hydrophone systems, and positions of research vessels on a graphical display in real time onboard a research vessel. The WILD system was used successfully on the MED 09 sea trial in July–September 2009 aboard the NATO Research Vessel Alliance. During the cruise, all observations were transmitted over the ship's network using custom data sentences developed by the researchers based on the National Marine Electronics Association 183 standard. The custom software extended ArcMap by allowing it to read and parse these sentences and update the maps in real time. This allowed scientists, biologists, and the ship's crew to tightly coordinate their efforts, permitting the best possible data collection and observation of the animals. The research provides scientists with better estimates of whale populations and their behavior in the Western Mediterranean Sea. [Work supported by the Office of Naval Research.]

1:45

4pAB3. Integrating computer-assistance and human-review to build richly annotated sound libraries. Harold Figueroa (Cornell Lab. of Ornithology, 159 Sapsucker Woods Rd., Ithaca, NY 14850, harold.figueroa@cornell.edu)

A current challenge to the effective application of bioacoustic survey methods is the curation and sharing of large collections of sound and metadata, and their integration into a data-analysis workflow that includes computer-assistance and human-review. Computer-assistance affords the annotation of ever larger amounts of data and human-review provides for sufficient quality of annotation and creates a feedback mechanism that supports continuous learning through an ever-expanding collection of training data. I will present some solutions to the above challenges that are being implemented as part of the Bioacoustic Resource Network (BARN) project. BARN provides a web-based interface to annotated sound collections with a back-end computational engine based on XBAT. The solutions provided by the BARN platform contain both technical and social elements. In the technical arena, we consider problems of data-modeling for extensible annotation; strategies for the unique identification of data, derived annotations, and computational resources; the definition of workflow-oriented programming interfaces; and web-service approaches to the larger-scale usability of these annotation libraries. These technical approaches are complemented by network-supported social strategies such as public repositories, open-source licensing, and a developer network. Integration of these technical and social components promises to overcome current challenges and realize the potential of bioacoustic methods.

2:05

4pAB4. Large scale passive acoustic data management. Carrie C. Wall (College of Marine Sci., Univ. of South Florida, 140 7th Ave. S, St. Petersburg, FL 33701, cwall@mail.usf.edu) and David A. Mann (Univ. of South Florida, St. Petersburg, FL 33701)

Passive acoustic recording systems can generate large amounts of data, especially given the increasing capability of inexpensive flash memory. DSGLab, an open-source database and data analysis system implemented with MATLAB and a portal to MYSQL, was created to handle large amounts of raw data collected by dozens of acoustic recorders over periods up to a year. Each recorded data file is tagged with header information including latitude, longitude, depth or altitude, timestamp, sample rate, and calibration, among other metadata variables, and uploaded to the database. This information allows all files in the database to be queried based on, for example, location (latitude, longitude, depth) and time (date, hour). Then the data analysis section processes the desired selection of files according to a user-specified signal processing chain, which does not require prior programming knowledge, and returns the results to a database and to individual files. The signal processing results can then be quickly browsed with the data viewer, providing a time-series of the processed data results, a map of recording locations, and facilitating retrieval of individual file timestamps and original raw data for quality control. The system is designed to allow inter-laboratory collaboration and data sharing. Examples using glider data are provided.

2:25

4pAB5. Management of acoustic detections via the Tethys database system. Marie A. Roch (Dept. of Comput. Sci., San Diego State Univ., 5500 Campanile Dr., San Diego, CA 92182-7720) and John A. Hildebrand (Scripps Inst. of Oceanogr., Univ. of California, San Diego, La Jolla, CA 92093-0205)

Driven by advances in consumer electronics and pattern recognition algorithms, passive acoustic monitoring has become a common tool for monitoring the vocal activity of animals. The increased acoustic effort to detect and categorize animal vocalizations has resulted in a need to organize acoustic detections and provide the ability to integrate other data sources. This paper presents Tethys, a prototype hierarchical database designed to provide flexible data handling in order to meet the needs of the bioacoustics community. Tethys provides mechanisms to easily adapt to changes in metadata structure resulting from equipment and algorithm evolution. It also provides an extension mechanism to permit the integration of other databases in a transparent manner, making querying alternative data sources (e.g., environmental data) appear like native queries. Tethys has the ability to communicate with multiple languages such as Matlab, Java, C++, and Python over secured network connections, and we show example applications where Tethys interfaces with the Horizons ephemeris service (NASA Jet Propulsion Laboratory) and Google Earth. [Work sponsored by ONR.]

4p THU. PM

2:45

4pAB6. Tools and methods for acoustic data storage and retrieval.
Gianni Pavan (CIBRA, Dept. of Animal Biology, Univ. of Pavia, Via Taramelli 24, 27100 Pavia, Italy, gianni.pavan@unipv.it)

Technologies available today allow to record a large amount of acoustic data that require to be stored and catalogued properly to facilitate retrieval, analysis, and comparison. The multisensor/multichannel recording system developed at CIBRA includes tools to georeference recordings and to collect context data (e.g., navigation data when used in research cruises). Specific file naming rules allow an easy retrieval of recordings and an easy connection to GIS maps. The analysis of large amounts of files (e.g., weeks of continuous 24/24-h recordings or months of scheduled recordings made by unattended equipment) is further facilitated by an automatic classification software able to discriminate simple acoustic categories and to produce a summary report about the contents of the recordings.

3:00—3:15 Break

3:15

4pAB7. Construction, calibration, and field test of a home-made low-cost hydrophone system for cetacean acoustic research.
Eduardo Romero Vivas (Centro de Investigaciones Biológicas del Noroeste, S.C., CIBNOR, Mar Bermejo 195 Col Playa Palo de Santa Rita, La Paz, BCS, Mexico, evivas@cibnor.mx) and Braulio J. León Lpez (Universidad Autónoma de Baja California Sur, UABCS, Mexico)

Marine mammals are reliable bioindicators of aquatic ecosystems health. Since cetacean highly rely on the use of sound for conspecifics interaction, feeding, and navigation, research in bioacoustics becomes fundamental to unravel the influence of anthropogenic activities on their environment and vocal behavior. Unfortunately, the widespread studies in this area are often limited due to lack of affordable equipment. This paper first describes how to build a low-cost hydrophone-amplifier system suitable for cetacean acoustic research and then shows how to perform hydrostatic pressure tests and acoustic calibrations using easily available tools. Finally, field recordings of individuals of two dolphin species: common dolphin (*Delphinus* spp.) and bottlenose dolphins (*Tursiops truncatus*) in La Paz Bay, Baja California Sur, Mexico using the proposed system and a professional hydrophone system [AQ-1s and ITC-1042 transducers (10 Hz–100 kHz)] are compared.

3:30

4pAB8. A review of fixed autonomous underwater recorders.
Renata S. Sousa-Lima (Bioacoustics Res. Program, Cornell Lab. of Ornithology, 159 Sapsucker Woods Rd., Ithaca, NY 14850, RSL32@cornell.edu), Thomas F. Norris (Bio-Waves Inc., Encinitas, CA 92024), and Julie N. Oswald (Oceanwide Sci. Inst., Honolulu, HI 96839)

Autonomous underwater recorders (ARs) are fixed passive acoustic electronic systems that acquire and store acoustic data internally (i.e., without a cable or radio link to a receiving station). They are deployed semi-permanently underwater (via a mooring, buoy, or resting on the sea-floor) and must be retrieved after the deployment period to access the data. ARs are capable of monitoring and recording underwater sounds over a wide range of spatial and temporal scales. As part of a Joint Oil & Gas Industry Program on Sound and Marine Life (JIP) sponsored effort, we reviewed over 30 ARs that are available for recording marine mammal sounds. They varied greatly in price and capabilities, from small hand-deployable units for detecting dolphin and porpoise clicks in shallow water to large units that can be deployed in deep water and record wide frequency bands for long durations. Considerations to weigh when selecting which device to use include price, longevity and depth of deployment, area to be monitored, and, most importantly, the bandwidth and the characteristics of sounds to be monitored (i.e., marine mammal call types and noise sources).

3:45

4pAB9. A review and inventory of cabled and radio-linked hydrophones for passive acoustic monitoring of marine mammals.
Thomas Norris, Julie N. Oswald (Bio-Waves Inc., 517 Cornish Dr., Encinitas, CA 92024, thomas.f.norris@bio-waves.net), and Renata Sousa-Lima (Cornell Lab. of Ornithology, Ithaca, NY 14850)

Fixed cabled hydrophone (FCH) and radio-linked hydrophone (RLH) systems are permanently, or semi-permanently, installed acoustic monitoring systems that are located on or moored to the seafloor. These systems have the capability to passively monitor bio-acoustic signals from marine mammals and, therefore, have great potential for monitoring and mitigation of potential impacts caused by anthropogenic activities. As part of a Joint Industries Programme sponsored effort, we reviewed past and present FCH and RLH systems with respect to their capabilities, advantages /disadvantages, and effectiveness for monitoring marine mammals in relation to oil and gas exploration/production activities. Based on this review, we provide examples and applications of these technologies. FCHs are typically powered by an external source and send data continuously to a receiving station that is usually located on shore. RLHs are moored or fixed to the seafloor, and transmit acoustic signals via radio-waves to a receiving station on shore. Both these systems allow acoustic data to be remotely monitored and processed in (or near) real-time. Hybrid systems can offer a good compromise between cost and capability by providing near real-time data transmission/processing with greater flexibility in deployment possibilities, but are usually limited in longevity and bandwidth of monitoring.

4:00

4pAB10. A review of computer-based methods for the automated detection, extraction, and classification of marine mammal sounds.
Julie N. Oswald (Oceanwide Sci. Inst., P.O. Box 61692, Honolulu, HI 96839), Thomas F. Norris (Bio-waves Inc., Encinitas, CA 92024), and Renata S. Sousa-Lima (Cornell Lab. of Ornithology, Ithaca, NY 14850)

Passive acoustic systems used to study and monitor marine mammals generate enormous datasets which are costly and time-consuming to analyze. As part of a Joint Industry Programme sponsored effort, we reviewed automated and semi-automated methods and software packages available to detect, extract, and classify marine mammal sounds; identified gaps in capabilities and knowledge; and suggested ways forward. Because of the variability in marine mammal sounds, no single method is effective for all species. While spectrogram correlation works well for stereotyped calls, more general methods like band-limited threshold detection are more effective for variable sounds. Feature extraction is a rapidly evolving field, but a reliable, automated method has yet to be successfully implemented into existing software. A major gap in our capabilities is the ability to reliably detect and classify the highly variable signals produced by some species. The development of effective, efficient, and standardized methods applicable to many species will require large, validated datasets. The acquisition, maintenance, and availability of such datasets will entail concerted, collaborative efforts. Development of common datasets and organization of workshops that focus on furthering detection, extraction, and classification methods are two ways to address these important issues in the automated analysis of marine mammal sounds.

4:15

4pAB11. A new method for recognition of non-voice sounds that is suitable for searching very large records. Boucher Neil (SoundID, 55 Murer Dr. Maleny, Queensland 4552, Australia, neil@soundid.net), Michihiro Jinnai (Kagawa Natl. College of Technol., Takamatsu, Japan), and Gianni Pavan (President, Centro Interdisciplinare di Bioacustica e Ricerche Ambientali, Univ. of Pavia)

An innovative non-voice sound recognition system based on a new similarity scale called the geometric distance has been developed. It has been purpose-designed for searching terabyte-sized files for wildlife recordists. Initially this method has been used for analyzing continuous recordings of rare parrots in Australia. More recent work has extended its applications to other birds, bats, frogs, and marine animals. The software can search for multiple reference sounds and multiple species in a single pass. It has been demonstrated to achieve recognition accuracy comparable to a human expert.

4:30—5:00 Panel Discussion

THURSDAY AFTERNOON, 18 NOVEMBER 2010 CORAL GALLERY 1B/2B, 1:00 TO 5:20 P.M.

Session 4pBB

Biomedical Ultrasound/Bioresponse to Vibration and Physical Acoustics: Ultrasound Contrast Agents for Molecular Imaging and Therapy

Azzdine Y. Ammi, Cochair
Cardiovascular Medicine, Oregon Health and Science Univ., 3181 S.W. Sam Jackson Park Rd., Portland, OR 97239

Saurabh Datta, Cochair
Siemens Ultrasound Medical Solutions USA Inc., 1230 Shorebird Way, Mountain View, CA 94043

Chair's Introduction—1:00

Invited Papers

1:05

4pBB1. Experimental and theoretical studies of microbubble interaction with tissue help optimize drug delivery and imaging. Charles F. Caskey, Shengping Qin, Xiaowen Hu, Azadeh Kheirolomoom, and Katherine W. Ferrara (Dept. of Biomedical Eng., Univ. of California at Davis, 451 Health Sci. Dr., Davis, CA 95616, cfcaskey@ucdavis.edu)

A multi-disciplinary approach is required to engineer ultrasound contrast agents for drug delivery and imaging. These micrometer-sized bubbles oscillate in response to acoustic energy, emitting non-linear echoes that can be differentiated from the tissue when imaged. If large enough, the oscillations generate perturbations capable of affecting the region surrounding the bubble. Here, microbubble activity is observed in scenarios that mimic those found *in vivo* to provide intuition about the mechanisms for contrast-enhanced imaging and drug delivery. High-speed imaging of microbubble oscillation at 1 MHz within *ex vivo* vessels reveals that microbubble expansion is significantly reduced, and the constrained oscillation imparts a force capable of displacing the surrounding vessel wall. [Caskey *et al.*, **122**, 1191-1200.] Bubble oscillation near a tissue-mimicking gel boundary shows fluid jets impinging on and disrupting the gel surface at 1, 2.25, and 5 MHz. [Caskey *et al.*, **125**, EL183–EL189.] The bubble concentration and acoustic parameters associated with gel disruption and vessel displacement are examined with respect to theoretical predictions, suggesting the ratio of pressure to frequency may be a good indicator for these phenomena. Finally, we also report on extensions of this work to the interaction of targeted contrast agents with tissue.

1:25

4pBB2. Correlating microvessel permeability directly with ultrasound-activated microbubble dynamics. Hong Chen, Andrew A. Brayman (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., Univ. of Washington, Seattle, WA), Andrew P. Evan (Indiana Univ. School of Medicine, Indianapolis, IN), and Thomas J. Matula (Univ. of Washington, Seattle, WA)

Ultrasonically activated microbubbles are promising candidates for targeted therapeutic applications such as drug and gene delivery. Understanding vascular permeability is thus important. In this study, ultra high-speed microphotography was used to directly visualize the dynamics of microbubbles in microvessels, under the excitation of ultrasound pulses with a duration time of about 2 μs, a center frequency of 1 MHz, and peak negative pressures (PNPs) between 0.8 and 7.2 MPa. Correlated vascular permeability was examined by imaging extravasation of bubbles from the vessels and by looking at the treated region histologically and under transmission electron microscopy. Local vessel distention and invagination caused by microbubble oscillations and liquid jet impinging on the vessel wall all appear to contribute to enhanced permeability for microvessels with diameters < 20 μm and insonation PNP > 4 MPa. Vascular permeation was also observed in larger vessels. However, in these cases vessel invagination was greater than distention in more than 80% of the cases and liquid jets always directed *away* from the vessel wall. We thus hypothesize that the dominant mechanisms for microbubble-induced permeation of smaller microvessels are different from larger microvessels. [Work supported by NIH EB000350 and AR053652.]

4p THU. PM

1:45

4pBB3. Models for the dynamical interaction of bubbles and blood vessels. Mark F. Hamilton, Todd A. Hay, Yurii A. Ilinskii, Evgenia A. Zabolotskaya (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX 78713-8029), Hong Chen, Wayne Kreider, Michael R. Bailey, and Thomas J. Matula (Univ. of Washington, Seattle, WA 98105-6698)

An understanding of ultrasound contrast agent (UCA) microbubble dynamics in blood vessels is important for ultrasound imaging and therapy. Recent photographs of UCAs excited by high-intensity ultrasound pulses in *ex vivo* vessels reveal unexpected phenomena. Strong bubble pulsations near the vessel wall induce contraction of the vessel that remains after bubble motion has subsided. Also, the bubble often translates toward the center of the vessel while forming a jet in the direction of translation. Three models describing the dynamics of a bubble in channels formed by elastic walls have been developed in an attempt to understand these phenomena. Two of the models describe a bubble between parallel elastic layers immersed in a viscous fluid. One incorporates bubble translation and aspherical bubble shape, which accounts for the onset of jetting, but is limited to incompressible fluids and layers with small shear moduli. The other accounts for fluid compressibility and arbitrary layer viscoelasticity, but is limited to stationary spherical bubbles. The third model is for a spherical bubble at an arbitrary but fixed location in a cylindrical viscoelastic tube. Comparisons between observed behavior and theoretical predictions will be presented and the relative merits of each model discussed. [Work supported by NIH DK070618.]

2:05

4pBB4. Biocompatible solid particles for controlled instigation of cavitation during therapeutic ultrasound. Manish Arora, Heiko Schiffter, A, Varun Manhas, Sarah Wagstaffe, and Constantin Coussios, C (Dept. of Eng. Sci., Inst. for Biomedical Eng., Univ. of Oxford, Oxford OX1 3PJ, UK, manish.arora@eng.ox.ac.uk)

Having been traditionally perceived as an unstable phenomenon to be avoided, cavitation is increasingly found to mediate many beneficial bioeffects, ranging from increased heat deposition during high-intensity focused ultrasound ablation to enhanced drug activity in the context of ultrasound-mediated drug delivery for cancer and stroke. However, due to its stochastic nature, reliable cavitation instigation *in vivo* is notoriously difficult. Ultrasound contrast agents have been previously exploited to lower the cavitation threshold, but lack stability and are generally too large to readily extravasate into tumors and other target tissues. In the present work, solid biocompatible micro/nanoparticles are proposed as novel cavitation nucleation agents that can overcome the limitations presented by shelled microbubbles. Such solid agents are manufactured from biodegradable polymers by solvent evaporation or by spray freeze drying, in order to achieve high surface roughness and hydrophobicity that harbors cavitation nucleation sites. Changing process parameters makes it possible to create particles of diameters spanning three orders of magnitude (0.15–250 μm) and different surface morphologies. These solid nucleating agents are found to lower the cavitation threshold significantly in water, tissue-mimicking gels, and porcine blood, and to enable reproducible instigation of cavitation activity during repeated exposure of the same target volume.

2:25

4pBB5. Ultrasound-enhanced thrombolysis in an *ex vivo* porcine carotid artery model. Kathryn E. Hitchcock, Nikolas M. Ivancevich, Kevin J. Haworth, Danielle N. Caudell Stamper (Biomedical Eng., ML 0586, 231 Albert Sabin Way, Cincinnati, OH 45267-0586), Deborah Vela (Texas Heart Inst., Houston, TX 77030), Jonathan T. Sutton, Gail J. Pyne-Geithman, and Christy K. Holland (Univ. of Cincinnati, Cincinnati, OH 45221)

Ultrasound enhances recombinant tissue plasminogen activator (rt-PA) thrombolysis via a cavitational mechanism. An *ex vivo* porcine carotid arterial model incorporating physiologic flow and pressure was developed and stable cavitation promoted for thrombolysis. Aged, retracted whole blood clots were exposed to plasma alone, plasma containing rt-PA (3.15 μg/ml), or plasma with rt-PA and the Definity ultrasound contrast agent (0.31 μl/ml), with and without 120-kHz continuous wave ultrasound at a peak-to-peak pressure amplitude of 0.44 MPa. An insonation scheme was formulated to promote and maximize stable cavitation activity by incorporating ultrasound quiescent periods that allowed for the inflow of Definity-rich plasma. Cavitation was measured with a passive acoustic detector throughout thrombolytic treatment. Thrombolytic efficacy was measured by comparing clot mass before and after treatment. Average mass loss for clots exposed to rt-PA and Definity without ultrasound was 34%, and with ultrasound was 83%, which constituted a significant difference ($n = 6$, $p<0.0001$). Without Definity there was no thrombolytic enhancement by ultrasound exposure alone at this pressure amplitude ($n = 6$, $p<0.0001$). Acoustic stable cavitation nucleated by an infusion of Definity enhances rt-PA thrombolysis without apparent treatment-related damage in this *ex vivo* porcine carotid artery model. [Work supported by Grant Nos. NIH R01-NS047603 and NIH T32GM063483, the University of Cincinnati Neuroscience Institute, the Albert J. Ryan Foundation Fellowship, and the Rindsberg Memorial Fellowship.]

Contributed Papers

2:45

4pBB6. Determination of optimal ultrasound parameters for *ex vivo* sonothrombolysis. Nikolas M. Ivancevich, Kevin J. Haworth, Kathryn E. Hitchcock, and Christy K. Holland (Biomedical Eng., Univ. of Cincinnati, Cincinnati, OH 45221)

Stable cavitation is correlated with thrombolytic efficacy for recombinant tissue plasminogen activator (rt-PA) mediated thrombolysis. An infusion of Definity® nucleated, promoted, and sustained stable cavitation in an *ex vivo* porcine carotid model exposed to 120-kHz continuous wave ultrasound. The optimal ultrasound pressure amplitude and the durations of active and quiescent periods were determined to maximize stable cavitation. Oxygenated porcine plasma with Definity® (0.31 μL/mL) and rt-PA (3.15 μL/mL) flowed through excised carotids loaded with whole-blood clots exposed to 120-kHz ultrasound and monitored with a 2.25-MHz passive cavitation detector. Using ultraharmonic and broadband emissions to quantify stable and inertial cavitation, respectively, intraluminal ultrasound-induced bubble activity was monitored in three vessels at four acoustic pressures. Thirty-minute treatment trials were simulated to determine the optimal acoustic pressure amplitude (0.44 MPa peak-to-peak) and ultrasound duration (8.5 s) to promote stable cavitation. A fixed quiescent period (19.5 s) allowed fresh Definity® to nucleate episodes of bubble activity. Clots exposed to this ultrasound scheme, Definity®, and rt-PA showed an average

ultraharmonic energy of 1.0×10^{-1} V^2 and an average broadband energy of 3.7×10^{-4} V^2, compared to 4.5×10^{-5} and 1.8×10^{-5} V^2, respectively, with ultrasound without Definity® and 4.3×10^{-6} and 2.4×10^{-6} V^2 without ultrasound or Definity®. [Work supported by NIH 2RO1 NS047603 and a grant from the University of Cincinnati Neuroscience Institute.]

3:00

4pBB7. The stresses induced by an ultrasound contrast agent oscillation on a micro-vessel wall. Nazanin Hossein khah and Kullervo Hynynen (Dept. of Med. Biophys., Univ. of Toronto, Toronto, ON M4N 3M5, Canada)

Non-inertial oscillating micro-bubbles inside micro-vessels induce some stresses on the vessel walls. It is important to use the bubbles safely and avoid rupturing the vessels. A 3-D model is made of an oscillating micro-bubble inside blood within a micro-vessel. The bubble's oscillation is implemented for a confined environment. The resulting micro-streaming makes the flexible vessel wall dilate. The vessel wall deformation is simultaneously coupled back to the fluid to make a more realistic model. The vessel walls were modeled as visco-elastic, elastic, and rigid materials. This model is solved using finite element method. Fluid shear stress and circumferential stress on the vessel wall are investigated when the vessel wall rigidity, bubble size, and vessel size are varied. When the size of the bubble increases or the radius of the vessel decreases, both the fluid shear stress and circumferential stress increase. In an elastic vessel (E= 5 MPa, Pa= 250 kPa, f=1 MHz, rv=5 μm), the circumferential stress can exceed the vascular strength (above 800 kPa) when the bubble's initial size is 3.5 μm or above. The same trend of increasing the stresses is observed when the vessel wall rigidity increases.

3:15—3:35 Break

3:35

4pBB8. Dynamical equations for deformation of bubble wall and neighboring tissue interface under acoustical excitation. Yurii A. Ilinskii, Todd A. Hay, Evgenia A. Zabolotskaya, and Mark F. Hamilton (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX 78713-8029)

Recent photographs of ultrasound contrast agent microbubbles excited by a high-intensity acoustical pulse in *ex vivo* blood vessels reveal that bubbles often translate toward the center of the vessel while forming jets in the direction of translation. Vessels have also been observed to first distend but then settle in a contracted state after bubble motion subsides. A model is developed for a translating aspherical bubble at an arbitrary position in an incompressible viscous fluid confined within a channel formed by parallel elastic layers with low shear moduli. Translation of the bubble and deformation of its surface are taken into account by including spherical harmonics for the dipole, quadrupole, and octupole modes in the boundary condition on the bubble wall. Lagrange's equations yield a system of four coupled second-order ordinary differential equations for the volume, position, and shape of the bubble. Numerical integration of the dynamical equations predicts behavior consistent with experimental observations, i.e., an initially spherical bubble located near one side of the channel and excited acoustically translates toward the center of the channel and initiates a jet in the direction of translation. Predictions of channel contraction are also consistent with observations. [Work supported by NIH DK070618.]

3:50

4pBB9. Green's function approach to modeling spherical bubble pulsation between viscoelastic layers. Todd A. Hay, Yurii A. Ilinskii, Evgenia A. Zabolotskaya, and Mark F. Hamilton (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX 78713-8029)

Experiments have shown that bubbles oscillating near surfaces of elastic media experience a translational force whose direction, toward or away from the surface, depends on elastic moduli and standoff distance. [Brujan *et al.*, J. Fluid. Mech. **433**, 251–314 (2001).] Bubble translation has also been observed in *ex vivo* blood vessels following acoustic excitation. [Chen *et al.*, J. Acoust. Soc. Am. **127**, 1977–(2010).] A model is developed for a stationary spherical bubble pulsating between two parallel viscoelastic layers. A Green's function for particle displacement is derived via angular spectrum decomposition and is used to calculate the translational force on a bubble due to pressure reflections from the layers. The predicted dependence of the direction of the translational force on elastic properties and standoff distance is in qualitative agreement with the reported measurements. The reflected pressure is also incorporated in a Rayleigh–Plesset equation for the nonlinear pulsation of the bubble. Evaluation of the Green's function on the surfaces of the elastic layers yields their displacements due to the bubble pulsations. Predicted surface displacements exhibit characteristics in agreement with wall displacements observed in blood vessels by Chen *et al.* [J. Acoust. Soc. Am. **125**, 2680–(2009).] [Work supported by NIH DK070618.]

4:05

4pBB10. Green's function for a volume source in a viscoelastic cylindrical tube. Evgenia A. Zabolotskaya, Yurii A. Ilinskii, Todd A. Hay, and Mark F. Hamilton (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX 78713-8029)

A model based on Green's functions that has been reported at previous meetings describes the field produced by a volumetric source in a bounded viscoelastic medium. Proposed applications of this model include scattering of sound in biological media and by objects buried in soil. The model was also adapted to study acoustically driven bubble pulsations near the surface of a viscoelastic medium, including calculation of the stress and strain in the medium. The methodology is generalized here to include cylindrical tubes. The Green's function is obtained for a volumetric source at an arbitrary location in a viscoelastic or viscous fluid-filled cylindrical tube. All modes of propagation in the elastic tube are taken into account as well as waves reflected from the tube wall. The Green's function is expanded both as a Fourier integral with respect to the axial coordinate and as a Fourier series with respect to the polar angle. Amplitudes of the Fourier coefficients are determined by boundary conditions for displacements and stresses at the tube wall and by matching conditions on the cylindrical surface containing the source. The Green's function is obtained in closed form and expressed through Hankel functions. [Work supported by NIH DK070618.]

4:20

4pBB11. Response of polymer-shelled ultrasound contrast agents to static overpressure. Parag V. Chitnis, Paul Lee, Jonathan Mamou (Riverside Res. Inst., F. L. Lizzi Ctr. for Biomedical Eng., 156 William St., 9th Fl., New York, NY 10038), John S. Allen, III (Univ. of Hawaii at Manoa, Honolulu, HI), and Jeffrey A. Ketterling (Riverside Res. Inst., New York, NY 10038)

The static-pressure threshold for rupturing polymer-shelled ultrasound contrast agents (UCAs) manufactured by POINT Biomedical (mean diameter 3 μm, shell-thickness-to-radius ratio (STRR) 7.5 nm/μm, count 961) and Philips Research (mean diameter 2 μm, STRR 40 nm/μm, count 1048) was investigated. The UCAs were subjected to static overpressure in a glycerol-filled test chamber with a microscope reticule lid at the top. UCAs were reconstituted in 0.1 ml of water and added on top of the glycerol surface in contact with the reticule. A video-microscope captured UCA images while glycerol was introduced at 5 ml/h, resulting in overpressure ranging from 2–180 kPa. UCAs were sized using semi-automated post-processing algorithm. The static pressure corresponding to each frame was recorded. Neither UCA population responded to static pressure until the rupture threshold was exceeded, which resulted in an abrupt destruction. The rupture pressure of both UCAs was uncorrelated to diameter. A k-means algorithm divided the pressure data for both UCA populations into three groups: primary, residual, and intact. The three groups exhibited different rupture behaviors although their mean diameters were not statistically different. The results indicated that the Philips UCAs were three-times more resilient to rupture than the Point UCAs. [Work supported by NIH EB006372.]

4:35

4pBB12. Subharmonic behavior of targeted and untargeted lipid encapsulated microbubbles at high ultrasound frequencies. Brandon Helfield, Emmanuel Cherin, and David Goertz (Sunnybrook Health Sci. Ctr., 2075 Bayview Ave., Toronto, ON M4N 3M5, Canada brandon.helfield @utoronto.ca)

Molecular imaging with ultrasound contrast agents (microbubbles) has recently gained interest as a feasible technique for disease-specific imaging, with applications ranging from intravascular ultrasound to small animal imaging. The attachment of targeting ligands to the microbubble shell enables a selective accumulation of bound microbubbles around a target site. The ability, however, to differentiate between the nonlinear signal from bound microbubbles and from unbound, circulating agent still remains a challenge. This study conducts a size-per-size comparison of the acoustic nonlinear response of individual streptavidin-coated MicroMarker microbubbles either bound (BMM) or adjacent (UBMM) to a compliant agarose/biotin gel surface. Bubbles were optically sized and insonified at 25 MHz over a range of pressures and pulse bandwidths. The subharmonic (nonlinear) response between unbound (n = 24) and bound (n = 29) bubbles was found to differ significantly, with UBMM bubbles having a higher propensity to initiate non-destructive subharmonics, in addition to lower onset threshold pressures and a smaller preferentially active diameter than BMM bubbles. In summary, this variability in the nonlinear response of the same bubble type between targeted and untargeted states can have implications for detection strategies, agent fabrication, and contrast imaging quantification for high frequency molecular imaging applications.

4:50

4pBB13. Are electrocardiogram premature complexes induced by ultrasonic cavitation biomarkers for cardiomyocyte necrosis? Douglas L. Miller, Chunyan Dou (Dept. of Radiology, 1301 Catherine St., Univ. of Michigan, Ann Arbor, MI 48109), and Benedict R. Lucchesi (Dept. of Pharmacology, Univ. of Michigan, Ann Arbor, MI 48109)

The objective of this study was to explore the relationship between lethal cardiomyocyte injury and arrhythmia induced by ultrasonic cavitation in the myocardium. Anesthesized rats were exposed in a heated water bath to 1.55-MHz focused ultrasound aimed at the heart with 2-ms bursts triggered at end systole during Definity (Lantheus Medical Imaging, Inc., N. Billerica, MA) contrast agent infusion. Premature complexes (PCs) where detected in electrocardiograph recordings, and Evans blue stained cardiomyocytes were scored in samples taken the next day. With 0.1 µl/kg/min infusion for 5 min, both effects increased strongly with increasing peak rarefactional pressure amplitude. At 8 MPa, statistically significant effects were found for no agent infusion relative to sham tests, indicating nucleation by naturally occurring nuclei. At 2 MPa, significant bioeffects were seen for 10, 1, and 0.1 µl/kg /min infusion but not for 0.01 µl/kg/min, which indicated a problematic nucleation. The numbers of PCs showed an exponential rise to maximum (r^2=0.88) for increasing numbers of stained cells with 4.5 stained cells per PC in the linear range. These results suggest a casual relationship between lethal cardiomyocyte injury and PCs induced by cavitation nucleation during cardiac ultrasound exposure. [Work supported by NIH Grant EB000338.]

5:05

4pBB14. Drug uptake by endothelial cells through targeted microbubble sonoporation. Nicolaas DeJong, Klazina Kooiman, Miranda Harteveld, and Ton Vandersteen (ErasmusMC, Rotterdam, The Netherlands, n.dejong @erasmusmc.nl)

Molecular ultrasound imaging uses targeted contrast agents consisting of microbubbles. Drug uptake using microbubbles can be induced by sonoporation, a method that induces transient cell membrane pores by oscillating or jetting microbubbles so that therapeutics can enter the cell. This study focuses on inducing sonoporation with CD31-targeted microbubbles in endothelial cells at low acoustic pressures. Biotinylated lipid coated microbubbles with a C4F10 gas core were made by sonication including CD31 antibody conjugation. Microbubble-cell behavior upon insonification at 1 MHz (6 × 10 cycle sine-wave bursts) at 80, 120, and 200 kPa peak negative acoustic pressure was studied with the Brandaris 128 high-speed camera at a frame rate of 12 Mfs. The cell-impermeable propidium iodide was used as indicator for increased cell membrane permeability due to sonoporation and was detected using fluorescence imaging. A total of 31 cells were studied, wherein all had one microbubble attached per cell. After insonifying the targeted microbubbles at 80, 120, and 200 kPa, only 6 burst of each 10 cycles, 12 cells PI uptake. This study reveals that targeted microbubbles can induce sonoporation. This feature may now be used in molecular imaging using ultrasound, thereby combining imaging and drug delivery.

THURSDAY AFTERNOON, 18 NOVEMBER 2010 — CORAL GARDEN 1, 1:00 TO 4:30 P.M.

Session 4pEA

Engineering Acoustics and Signal Processing in Acoustics: Physical Array Microphone Development and Applications for Sound Source Identification

Samir N. Y. Gerges, Chair

Mechanical Engineering, UFSC, Campa Universitario Trindade, Florianopolis, Sc, 88040-900, Brazil

Contributed Papers

1:00

4pEA1. Acoustic scene analysis using microphone arrays. Dmitry Terez (Solarium, 264 Eleanor Terrace, Cherry Hill, NJ, dmitryterez@gmail.com)

A spherical microphone array is built to perform acoustic scene analysis in a realistic reverberant environment with several speakers talking at the same time. Speech sound sources and their significant reflections can be localized and grouped together using a newly developed algorithm. The algorithm uses pitch cue to achieve speech sound source segregation and has significantly better time resolution compared to other known methods. The time-delays of arrivals are also precisely computed and match geometric calculations. Attempts are made to mimic some well-known auditory effects in humans such as the Haas effect as an essential first step toward usable computational auditory scene analysis. The proposed methods can significantly improve signal-to-noise ratio of speech input for robust speech recognition by combining sounds from speech sources of interest with their reflections and rejecting all other sources. Real-time algorithm implementation on commodity hardware is enabled by using a graphics processor.

1:15

4pEA2. Electronic sonar system. Carlos Barroeta, Floriberto R. Ortiz, Juan Francisco Novoa Colin, and Rodrigo C. Lopez (ESIME, IPN, Mexico City FDM 07038, Mexico, cbarroet@hotmail.com)

This is about a sonar system with specific characteristics. This sonar is able to measure up to 3 m and work is being done to get the best frequency and time period. The gate should be closed if the time is larger than the size of the sonar capacity. T1 is the time for the ongoing signal and T2, the total time is equal to 2T1. The system will be tested for many different frequencies beginning with a frequency of 15 000 kHz.

1:30

4pEA3. Structural health monitoring of plate by using ultrasonic Lamb wave and time reversal, multiple signal classification beam forming. Je-Heon Han and Yong-Joe Kim (Dept. of Mech. Eng., Texas A&M Univ., Collge Station, TX 77843-3123, jeep2000@tamu.edu)

The objective of this study is to detect structural defects such as delaminations and notches in a plate by exciting it at a frequency to selectively generate a specific Lamb wave and recording reflected wave signals with a piezoelectric wafer sensor/actuator array. Here, the lowest symmetric (S0) and anti-symmetry (A0) modes are excited by adjusting excitation frequency to increase signal-to-noise-ratio. Note that the measured wave signals consist of the wave signals reflected from the defects as well as the boundaries of the plate. For the purpose of reducing the effects of the reflective waves from the boundaries and pinpointing out the structural defect locations in a high spatial resolution, it is proposed to apply a time reversal (TR), multiple signal classification (MUSIC) algorithm to the measured signals. Through an experimental measurement made with an array installed on a plate, it is demonstrated that the TR MUSIC beam forming procedure can be used to effectively identify the structural defect locations, their relative sizes, and their defect types without significant degradation induced by the boundaries of the plate on the resulting beam forming image.

1:45

4pEA4. Subsurface characterization of microelectronics using high-frequency contact ultrasound. Eric A. Dieckman, Mark Hinders, and Jonathan Stevens (Dept. of Appl. Sci., College of William and Mary, P.O. Box 8795, Williamsburg, VA 23187, eric.dieckman@gmail.com)

The goal of this project was to develop and demonstrate a prototype system for characterizing subsurface features in microelectronic components using high-frequency pulse-echo ultrasound. Of particular interest are suspected counterfeit router components, which need to be inspected quickly and nondestructively with portable equipment that does not require extensive training to operate. Although ultrasound in the 100 MHz frequency range routinely images the relevant subsurface features in microelectronic components, scanning acoustic microscopes are neither portable nor inexpensive, and they require the component to be submerged in a bath of coupling water. Our alternative approach was a custom-designed probe which uses direct contact with the component surface and does not employ water for coupling. Also, rather than high-precision computer-controlled scanning to produce images, the pulse-echo waveforms are processed using our discrete wavelet fingerprinting algorithm. This maps out the subsurface layers and interfaces and finds differences from nominal values that indicate that the part is flawed or counterfeit.

2:00

4pEA5. Image enhancement of the temperature field by the inverse estimation using the acoustic pyrometry. Tae-Kyoon Kim and Jeong-Guon Ih (Ctr. for Noise and Vib. Control, KAIST, Sci. Town, Daejeon 305-701, Rep. of Korea, thaad@kaist.ac.kr)

In the acoustic pyrometry for the measurement of temperature distribution of a cross section in a furnace, the temperature can be reconstructed from the measured time-of-flights between sensor-actuator array systems. The calculated transfer matrix is composed of the approximated slowness function and its coefficient vector. In this study, the least-squares method is used to inversely reconstruct the temperature. If the transfer matrix is highly singular, reconstruction result will be deteriorated in the presence of the measurement error. To reduce the effect of the singularity, the redundancy of the sensor data can be removed and/or the regularization of the restored field can be employed. As a simulation example, a thermal field within a rectangular cross section of a 1-D infinite duct with asymmetrical temperature distribution was considered. Condition number of the transfer matrix revealed a highly singular state of the system, regardless of the number of sensors. Sensor/actuator positions were selected by using the effective independence method. A modified Tikhonov technique was adopted for the regularization. Although the resultant precision depended on the number of sensors, the reconstruction error was drastically reduced, from 214% to 14% in the case of using 36 sensors, in particular by applying the regularization. [Work supported by BK21.]

2:15

4pEA6. Identification of noise sources in notebook systems using acoustic holography. Jose Cordova (Systems Res. Ctr. Mexico, Intel, Periferico Sur 7980 Edificio 4E, Tlaquepaque, Jalisco 45600, Mexico, jose.a.cordova@intel.com)

The acoustic noise generated by notebook computers is an important quality aspect for consumers. Cooling fans, hard disk drives, and optical disk drives are the main sources of noise in these types of systems. Standard measurement techniques such as sound power or sound pressure are used for measuring the overall acoustic energy generated by a device or the sound pressure at a specific point in space respectively, but give us little information of the location of the noise sources and the transmission paths such as noise leaks. This paper outlines the use of acoustic holography for the identification of noise sources and transmission paths in notebook systems. The setup consists of static reference microphones and a roving array of microphones used to scan the surface above a notebook system. With acoustic holography software, the noise field is constructed and back propagated to the surface where the location of the main noise sources is identified. 3-D isosurface sound pressure maps and sound intensity maps (passive and reactive) are shown for notebook computers with different airflow system impedances. The technique is validated by reconstructing the radiation characteristics and obtaining the location of elementary monopole and dipole sources.

2:30

4pEA7. Experimental analysis of laptop fan noise radiation by acoustic source decomposition and inverse boundary element methods. Flor López Rodríguez, Felipe Orduña Bustamante, and Antonio Pérez López (Centro de Ciencias Aplicadas y Desarrollo Tecnológico, Universidad Autónoma de México, Circuito Exterior s/n, C.P. 04510, Ciudad Universitaria, DF, Mexico, florlopezr@yahoo.com.mx)

Measurements are presented of fan noise radiation of a complete laptop system in a hemi-anechoic chamber using different microphone arrays. The analysis of sound radiation is performed by means of acoustic source decomposition and inverse boundary element methods. These methods allow determination of equivalent source distributions and radiation patterns of the complete laptop environment. Results are extended and validated by comparison with sound intensity mappings and near-field acoustic holography.

2:45—3:00 Break

3:00

4pEA8. A directional dogbone flextensional transducer. Stephen C. Butler (Naval Undersea Warfare Ctr., 1176 Howell st., Newport, RI 02841)

Sonar flextensional transducers, in order to transmit energy in one direction, are combined into arrays of elements that are spaced 1/4 wavelength apart. This spacing is dependent on the sound speed of the water; thus, directionality is reduced as the sound speed changes. Here a single projector that is 1/3 wavelength in size that is capable of developing unidirectional beams and independent of sound speed is described. The directional class VII dogbone flextensional is a modified version of an oval shaped class IV flextensional having a concave shell rather than a convex one. The piezoelectric ceramic stack is a trilaminar bar with two active sections separated by an inactive center section. By driving both sections of the stack in-phase, the shell is driven into an omnidirectional radiation pattern. Driving each stack section 180 deg out-of-phase causes a bending mode in the stack, resulting in a dipole radiation pattern. Driving both sections using complex coefficients determined from the omnidirectional and dipole mode patterns, a cardioid directional radiation pattern is developed. Cardioid patterns are developed over an octave frequency band with a front/back pressure ratio of 50 dB. COMSOL/acoustics module FEA code is used to predict in-water electroacoustic performance and compared with experimental data.

3:15

4pEA9. A hydrophone based on the piezoelectric gate of field effect transistor for deep-sea application in low-frequency range. Min Sung, Kumjae Shin, and Wonkyu Moon (Dept. of Mech. Eng., Pohang Univ. of Sci. and Tech., PIRO, 416(VATrans Lab.), San 31, Hyoja, Namgu, Pohang, Kyungbuk 790-784, South Korea, smmath2@postech.ac.kr)

A hydrophone based on the piezoelectric gate of field effect transistor (FET) for deep-sea application in the low-frequency range (10 Hz–20 kHz) was designed. The basic structure was piezoelectric thickness mode, which was directly applied on the gate area of a field effect transistor. Several types of sensors which used PVDF film integrated with FET have been reported. They have sensitivity limit in the low-frequency range due to stray capacitance by extended electrodes and low coupling coefficient of PVDF film. To achieve high sensitivity, a piezoelectric ceramic was directly applied to the gate area of the FET, which resulted in decreased effect of stray capacitance. Depletion mode FET with large gate area was adopted to realize high transconductance in the operation voltage range of the piezoelectric ceramic. The head mass structure, which was motivated by the Tonpilz structure, for both sound pressure amplification and prestress was placed between the clamped diaphragm and the piezoelectric ceramic. To develop the endurance for the high hydrostatic pressure in deep-sea, oil backing on the backside of the diaphragm was proposed. The parametric study based on the traditional FET model was performed in the design procedure. The comparison of simulation and experiment will be presented. [Research supported by MRCnd.]

3:30

4pEA10. Fabrication and characterization of a sapphire based fiber optic microphone for harsh environments. Benjamin A. Griffin, David A. Mills, Tony Schmitz, and Mark Sheplak (Dept. of Mech. & Aero. Engr., Univ. of Florida, 231 MAE-A, Gainesville, FL 32611, griffo@ufl.edu)

This paper describes the fabrication and characterization of a sapphire based fiber optic microphone designed for the harsh environment of gas turbine engines. The performance requirements are driven by turbine inlet temperatures that have risen in excess of 1500 °C. The harsh environment makes conventional instrumentation unsuitable for time-accurate, continuous, direct measurements. The use of commercially available sapphire substrates and optical fibers allows for performance in extremely high temperature environments due to the high melting point and matched coefficient of thermal expansion. The sensor is based on a fiber optic lever transduction mechanism with a remote photodiode optical readout allowing for isolation of the electronics from the harsh environment. The microphone's diaphragm consists of a thin sapphire substrate with a sputtered titanium/platinum reflective surface. The back cavity is formed by macro-machining of a 1-mm-thick sapphire substrate. The diaphragm and back cavity are joined using high temperature, alumina based epoxy. The paper focuses on the fabrication and packaging of the sensor with characterization results given using silica based optical fibers at room temperature. The future work is aimed at the implementation of sapphire optical fibers and high temperature characterization. [Work supported by the Air Force.]

3:45

4pEA11. Commercial packaging of an optical microelectromechanical systems microphone. Michael L. Kuntzman (Dept. of Elec. and Comput. Eng., Univ. of Texas, 2501 Speedway Blvd., Austin, TX 78712, mkuntzman@mail.utexas.edu), Karen D. Kirk (Univ. of Texas, Austin, TX 78712), Caesar T. Garcia, Guclu A. Onaran (Silicon Audio, LLC, Austin, TX 78702), and Neal A. Hall (Univ. of Texas, Austin, TX 78712)

A microelectromechanical systems (MEMS) optical microphone has been presented that measures the interference of light resulting from its passage through a diffraction grating and reflection from a vibrating diaphragm. [J. Acoust. Soc. Am. **122**, (2007).] In this embodiment, both the diffractive optical element and the sensing diaphragm are micromachined on silicon. Additional system components include a semiconductor laser, photodiodes, and required readout electronics. In our efforts to commercialize this technology for hearing-aids and other applications, a goal has been set to achieve a microphone contained in a small surface mount package (occupying $2 \times 2 \times 1$ mm^3 volume), with ultra-low noise (15 dBA) and broad frequency response (20 Hz–20 kHz). Such a microphone would be consistent in size with the smallest MEMS microphones available today, but would have the noise performance characteristics of professional-audio microphones at least 10× larger in size and 10× more expensive to produce. This paper will present several unique challenges in our effort to develop the first surface mount packaged optical MEMS microphone, including the optical and acoustic design of the package. Dynamic models used for simulating frequency response and noise spectra of complete capsules will be presented and compared with measurements performed on recent prototypes.

4:00

4pEA12. A microelectromechanical systems-based piezoelectric microphone for aeroacoustic measurements. Matthew D. Williams (Dept. of Mech. & Aero. Engr., Univ. FL, 231 MAE-A, Gainesville, FL 32611, mdwilli@ufl.edu), Benjamin A. Griffin (Dept. of Mech. & Aero. Engr., Univ. FL, Gainesville, FL 32611), Jessica Meloy (Dept. of Elec. & Comp. Engr.., Univ. FL, Gainesville, FL 32611), and Mark Sheplak (Dept. of Mech. & Aero. Engr., Univ. FL, Gainesville, FL 32611)

This paper describes the packaging and characterization of micromachined piezoelectric microphones designed as aircraft fuselage instrumentation for full-scale noise characterization flight tests. An important requirement for such microphones is a high maximum sound pressure level (SPL) >150 dB coupled with noise floor <45 dB SPL (narrow bin) and flat frequency response in the audio band (20 Hz–20 kHz). In contrast to many past micromachined piezoelectric microphones that use PZT and ZnO for piezoelectric transduction, aluminum nitride (AlN) was used due to inherent advantages in dielectric loss and signal-to-noise ratio. The microphone structure includes a 500–900-μ m diameter circular diaphragm composed of a structural layer and an annular AlN/metal film stack on a silicon substrate. Promising microphone die were selected via laser vibrometer measurements followed by packaging of the microphones and associated electronics in individual 1/4 in. tubular housings. Characterization of the packaged microphones included frequency response, noise floor, and total harmonic distortion measurements. Early results for a packaged microphone reveals a sensitivity of 27 μV/Pa, dynamic range of 36–157 dB, and resonant frequency of 176 kHz.

4:15

4pEA13. A micro-machined piezoelectric sensor for monitoring viscosity-variation. Sungjoon Choi, Geunbae Lim, and Wonkyu Moon (Dept. of Mech. Eng., POSTECH san 31, Hyojadong, Namgu, Pohang, Korea csjangel@postech.ac.kr)

A micro-machined piezoelectric sensor is developed for monitoring the viscosity-variation in small amounts of liquid in real time. The device is initially devised for monitoring the state of polymerized chain reaction (PCR) of or ribonucleic acid and being integrated into a Lab-on-a-chip (LOC) since the viscosity may increase with an increase in the amount of DNA with PCR. The sample fluid is assumed to be homogeneous so that the wave propagation characteristics in a micro-channel can be used for measurement. The developed measuring technique is devised to be performed with a small amount of sample liquid and to be able to monitor the viscosity variation in real time. The developed device is composed of two chambers connected by multi-micro-channels and was fabricated using micro-machining technology so that it can be fabricated with other components when it is integrated in LOC. Each chamber has a unimorphic piezoelectric diaphragm; one for generating and the other for sensing sound waves. From the feasibility test performed with test samples (alcohol-water mixtures), it can be shown that the viscosity variation can be measured via a single-frequency driving with the amount of about 13 nl sample. Either continuous harmonic or ton-burst inputs at 4.84 MHz can be used for operating the sensor. [Research supported by R0A-2007-000-20042-0 and 10024719.]

Session 4pED

Education in Acoustics and ASA Student Council: Project Listen Up

Preston S. Wilson, Chair
Dept. of Mechanical Engineering, Univ. of Texas at Austin, 1 University Station, Austin, TX 78712

Chair's Introduction—1:00

Invited Papers

1:05

4pED1. Expo Acustica. José Ángel Tafolla Mrquez and Montserrat Guadalupe Victoria (Acoust. Acad. of the School of Mech. and Elec. Eng. of the Natl. Polytechnic Inst., Mexico City, Mexico)

During the Expo Austica in 2002, the acoustics academy of the previously mentioned Institute noted the relevance of acoustics study in the country's development in the industrial, cultural, and intellectual fields, taking into account the areas of application of the mentioned specialty: audio, architectural acoustics, musical acoustics, noise control, bioacoustics, psychoacoustics, vibration analysis, acoustics metrology, and acoustic signal analysis. Since then, such event has been held, each time achieving greater support from the private sector. The objective was to promote scientific and technological advances of the specialty of acoustics in the country's productive sector. Particular objectives were: (1) disseminate the activities of the specialty within the acoustic National Polytechnic Institute; (2) project the image and work of students and professors specializing in acoustics to achieve successful integration into the labor market; (3) Communicate to the visiting companies that the School of Mechanical and Electrical Engineering at the IPN is a provider of acoustic specialists in Mexico; (4). meet specific applications related to the subject through the conference with support of the team displayed, and display student projects. Location: School of Mechanical and Electrical Engineering of the National Polytechnic Institute unity Zacatenco.

1:20

4pED2. Standing wave measurements in tubes. Stanley A. Cheyne and Walter C. McDermott (Dept. of Phys. and Astronomy, Hampden-Sydney College, Hampden-Sydney, VA 23943, Australia)

Sound speed measurements have been made using a standing wave technique for possible inclusion in Project Listen Up! In a previous paper [S. A. Cheyne and W. C. McDermott, "Sound speed measurements in air using a variable sound source and tubes," J. Acoust. Soc. Am. **125**, 2625 (2009)], sound speed measurements in tubes were made using traditional standing wave techniques. A low-cost variable frequency sound source was constructed and used to produce the sound waves. Resonances in the source itself made it difficult to make reliable measurements. Also, it was difficult to detect the tube resonances by simply listening to the resonances with one's ear. In this work, we have improved the quality of the apparatus to make it easier to generate and detect tube resonances.

1:35

4pED3. Impulse excitation of the singing rod. J. R. Gladden, III (Dept. of Phys. and Astronomy, Univ. of Mississippi, University, MS 38677, jgladden@olemiss.edu)

A classic demonstration in acoustics is the excitation of both longitudinal normal modes of a metallic rod. Typically the fundamental mode is excited by holding the bar at the center and stroking the bar with resin coated fingers along the length to excite the first longitudinal mode to a very high amplitude. In this version, several longitudinal modes are excited at once with a broadband impulsive force such as striking the end with a hammer. A mix of the fundamental and higher order tones can be initially heard; however, the higher order modes decay rather rapidly, leaving only the fundamental to ring. Fourier analysis can be used to measure the decay constants of these modes. Further, by choosing rods of identical geometry but different materials (such as aluminum and steel), the different resulting tones can prompt a discussion of the relationship between elastic moduli and sound speed.

1:50

4pED4. The singing shoebox. Part 2: Educational opportunities in a $5 loudspeaker. Scott P. Porter, Daniel J. Domme (The Graduate Program in Acoust., The Penn. State Univ., Appl. Sci. Bldg., State College, PA 16802), and Jeffrey S. Whalen (Dept. of Mech. and Nuclear Eng., The Penn. State Univ., State College, PA 16802)

Moving-coil loudspeakers are an excellent resource for introducing students to acoustics: their widespread use makes them familiar to nearly everyone and the interdisciplinary nature of the device forms a technically rich problem. To exploit this, the authors have created a simple, low-cost loudspeaker in a shoebox to function as both a science project and demonstration. Because of the many technical concepts illustrated by this device, it is an appropriate demonstration for a wide range of academic levels. Already, the shoebox

speaker has been used as a teaching demonstration in a previous session [J. Acoust. Soc. Am. **127**, 1910]. In this talk, the design and construction of the shoebox speaker will be shown, highlighting the incorporation of everyday materials. The focus of this talk will be on the educational opportunities afforded by this demonstration and its suitability for Project Listen Up. Lastly, the shoebox speaker will be demonstrated.

Contributed Papers

2:05

4pED5. Interactive exploration of three-dimensional deflections on vibrating structures using linked multiview displays. José Luis Villarreal (Depto. Vis. Cient., DGSCA UNAM, Circuito Exterior s/n, Cd. Universitaria, Coyoacán, DF, CP04510, Mexico, josel@servidor.unam.mx), Jesús Alejandro Torres (Facultad de Ingeniería, UNAM, Cd. Universitaria, Coyoacán, DF, CP04510, Mexico), and Ramón Ramírez (DGSCA UNAM, Cd. Universitaria, Coyoacán, DF, CP04510, Mexico)

A software was developed to explore and compare structural deflections under different conditions. Modal analyzes of the same structure with slight variations were previously calculated through finite element method (FEM). Four modal analyzes of a guitar top plate (at different stages of its construction) were compared as an example using the developed visualization software. The resulting FEM meshes were imported to the developed application due to the incapability of the FEM software for real time visualization of the results of different modal analysis data. In the main screen of the developed software were shown several mode shapes. The user picks any mode shapes to compare, and their 3-D controllable versions appear in full or in divided screen versions (depending on the number of modes picked). The developed software allowed the models to be freely turned, zoomed, and moved using any standard laptop. The whole software characteristics brings a powerful technique to researchers and teachers, exploring and explaining some details in 3-D structural deflections, often limited and showing only static 2-D images.

2:20

4pED6. Fields everywhere. Michael Ermann, Mitzi Vernon, Ana Jaramillo, Barnobi Chris, and Han Joseph (School of Architecture, Virginia Tech, 201 Cowgill Hall (0205), Blacksburg, VA 24061-0205, mermann@vt.edu)

This proposed project aims to define science education as architects might define science education. There are values associated with points in space, those values can be mapped to generate a field, and those fields can have meaning. These values might be light levels, wind speeds, carpet pile heights, or sound pressure levels; when mapped, values might tell of a shadow, an approaching cold front, the former location of a piece of furniture on a carpet, or sound propagation. By allowing for a spatial and immersive science museum exhibit dedicated to sound propagation, frequency, and localization, this project aims to make plain the invisible nature of sound. Authenticity of phenomena is important to this endeavor. The points in space envisioned are not mathematical representations on a Cartesian grid, but real points in real space among and in between the museum visitors. Indeed, nothing in the proposed exhibit is simulated at all: patrons can experience the speed, propagation, frequency, and localization of sound in real time and real space. It is hoped that moving beyond the normative regime of kiosks simulating phenomena, and into a realm that is spatial, museum goers can better take ownership of the scientific content.

2:35

4pED7. Ahorra ahora para comprar un carro caro. Speech perception in teaching pronunciation: Learning Spanish liquid phonemes by Chinese, Korean, and Japanese students. Aída Espinosa (Centro de Enseñanza para Extranjeros, Av. Universidad 3002, Ciudad Universitaria, Distrito Federal, Mexico)

Chinese, Korean, and Japanese students (Asian) have trouble pronouncing Spanish liquid phonemes. The main aims of this research were to prove that speech perception is a requisite to improve the production of these sounds, and that adult Asian students can accurately produce Spanish liquid phonemes if they are guided to change their patterns of perception. This text is based mainly on speech perception studies by Flege and colleagues, [(1995); (2007)] who have found that one of the difficulties experienced by students in the oral production during a second language acquisition is the result of speech perception habits acquired with their first language. The study was carried out with subjects who study at Universidad Nacional Autónoma de México (UNAM). An experimental group was exposed to stimuli to raise their speech perception; other experimental group were given exercises of listen and repeat and given articulatory instructions; a control group was not given any kind of training. The results were assessed by Fisher Exact Probability Test, which allowed us to prove that there exists a strong relationship between learning liquid phonemes and training with speech perception exercises. It is concluded that adult Asian students can improve their pronunciation.

2:50

4pED8. Development of an educational impedance tube: Experimental vibroacoustic characterization. Noé Melo (Depto. de Engenharia Mecânica (UnB/FT/EnM), Universidade de Brasília, Campus Universitário Darcy Ribeiro, Faculdade de Tecnologia, Bloco G, 70.910-900 DF, Brazil), Marcus Vinícius Morais, Maria Alzira Nunes, and Alessandro Oliveira (Universidade de Brasília, 72.405-610 Gama, DF, Brazil)

One of the great difficulties encountered by designers who work with room acoustics is to know the absorption coefficient of the material to be employed in the project. The tube wave, also known as impedance tube, is a device which determines this coefficient and also determines the acoustic impedance of materials. These values are essential for choosing products that best fit the needs of an acoustic treatment project, either indoors, machinery, pipes, or mufflers. This paper presents the first studies to construction and characterization of an educational impedance tube (Kundt's tube) called TIGaD (demonstrative impedance tube of the Faculty at Gama). In this first phase we will present a dimensional study of the tube, its construction details, an experimental analysis of its vibroacoustic characteristics, and, finally, its limitations in frequency range. This project belongs to the educational area since this experimental apparatus is being developed to equip the laboratory of acoustics and vibration (NVH Gama). The aim is to obtain low-cost equipment which makes possible the study of the various acoustic phenomena such as absorption, reflection, impedance, and further demonstration of plane waves and the determination of the speed of sound.

Session 4pMUa

Musical Acoustics: Acoustics of Guitars

Thomas D. Rossing, Cochair
Dept. of Music, Stanford Univ., Stanford, CA 94305

Felipe Orduna-Bustamente, Cochair
Lab de Acustica y Vibraciones, CCADET-UNAM, Mexico DF, CP 04510, Mexico

Ricardo Ruiz-Boullosa, Cochair
Lab de Acustica y Vibraciones, CCADET-UNAM, Mexico DF, CP 04510, Mexico

Invited Papers

1:00

4pMUa1. Guitar acoustics research 1980–2010. Thomas D. Rossing (Dept. of Music, Stanford Univ., Stanford, CA 94022, rossing @ccrma.stanford.edu)

Although the guitar is more than 200 years old, much of our understanding of its acoustical properties has developed during the past three decades. The low-frequency acoustical behavior can be explained on the basis of the coupled vibrations of the top plate, the back plate, the ribs, and the enclosed air cavity. Prominent peaks in the frequency response can be attributed to reflex coupling between these elements, and the luthier controls the tonal response of the instrument by skillful design and tuning of the various components through adjustment of their stiffness and mass. We discuss some of the more important acoustical research on guitars during the past three decades.

1:20

4pMUa2. Subjective and physical experiments related to the tuning of classical guitars. Felipe Orduña-Bustamante, Francisco Fernández del Castillo Gómez, Erika Enedina Martínez Montejo, and Humberto Contreras Tello (Grupo de Acústica y Vibraciones, Centro de Ciencias Aplicadas y Desarrollo Tecnológico, Universidad Nacional Autónoma de México, Mexico)

The subject of tuning the classical guitar is discussed in the light of subjective experiments during which guitarists tuned a classical guitar using different tuning methods. Tones of the six open strings were recorded after each tuning session, and the perceived musical pitch was estimated using a psychoacoustically based pitch detection algorithm. Also, the input mechanical mobility was measured at the top plate bridge, at the nut bridge, and at all the frets. Frequency shifts due to string-body coupling were estimated for the frequency components of all possible (open and fretted) guitar tones, and perceived musical pitch was also estimated for these tones. Results are presented of tuning accuracy and variability for individual guitarists as well as among different guitarists. Also, estimates are presented of overall guitar tuning accuracy, depending on string-body coupling, fretted-tone over-tension, tuning method, and other factors.

1:40

4pMUa3. Research on the acoustics of the guitar at the National Autonomous University of Mexico. Ricardo Ruiz-Boullosa (Laboratorio de Acústica y Vibraciones, CCADET, UNAM, Cd. Universitaria, México DF, CP 04510, Mexico, ricardo.ruizb@gmail .com)

The research on the acoustics of the guitar has had an interesting though brief history that began more or less in the 1960s. The results spread over a broad range of subjects, but is on the main focused in the lower modes of the guitar. Work being done now includes such topics as visualization of modes of vibration, radiation, etc., by the use of FEM and BEM methods. Lines of investigation that have some bearing on the sound of the guitar remain to be made. Tentative initial work on several topics is presented that have been undertaken in the acoustics laboratory at the National Autonomous University of Mexico. These include ways to measure the problems in tuning, the contribution of the neck to the radiation, the importance of high frequencies and modes of vibration on the sound of the guitar, the influence of the bridge design on the vibration and radiation of the top plate, and the influence of factors such as humidity. Some of these experimental lines, and others, could further expand the horizon of the acoustics of the guitar.

2:00

4pMUa4. Evaluating design modifications in classical guitars. Bernard Richardson (School of Phys. and Astronomy, Cardiff Univ., 5 The Parade, Cardiff CF24 3AA, UK, RichardsonBE@cardiff.ac.uk)

The structure and musical usage of the classical guitar has changed considerably over its long history and the instrument still continues to develop. Major the changes in construction of guitars in the latter part of the nineteenth century and changes in their playing style helped establish the classical guitar as a serious solo instrument, which still holds considerable prominence today. Many makers and players strive to “improve” the instrument, but their objectives are rarely clearly defined or the overall consequences of specific changes are misunderstood. This paper will try to establish an objective framework for establishing what constitutes the sound of a

guitar and how that sound may be modified by changes in the design or construction of the instrument or through the use of different materials. It will bring together objective experimental and theoretical work on the structural vibrations of guitars and acoustical radiation with subjective psychoacoustical evaluation of real and synthetic sound signals with the aim of helping to answer the many questions posed by guitar makers.

2:20

4pMUa5. Plucking the string: The excitation mechanism of the guitar. Sandra Carral (Univ. of Music and Performing Arts, Vienna; Inst. of Musical Acoust.; Anton von Webern Platz 1, 1030 Vienna, Austria)

The excitation mechanism of any musical instrument is a key component of the sound production and resulting sound of the instrument. In the case of the guitar, the manner and object with which the strings on the guitar are plucked determine, to some extent, the sound of the instrument. Parameters such as the plucking position, initial displacement, and deformation of the string have a strong influence on the resulting sound and its spectrum. In this talk, these and other parameters, as well as how they influence the sound, will be explained in detail, reviewing the studies available on this subject.

2:40—2:50 Break

2:50

4pMUa6. What are the limits of the discrete stringed instrument model? Mark French and Charles Jackson (MET Dept., 138 Knoy Hall, 401 N. Grant St., W. Lafayette, IN 47907, rmfrench@purdue.edu)

Many different analytical models for stringed instruments have appeared in the technical literature, but one of the most widely used is a simple discrete model presented by Christensen and Vistisen in 1980. This model accounts, in a simple and approximate way, for the first two or three modes of a typical stringed instrument such as a guitar and includes coupling between the structure and the enclosed air. For the model to be used in predicting the effect of design changes, it is important to understand its limits. In order to explore the applicablity of the model, we have constructed a fixture that models the volume of an acoustic guitar with rigid side walls and flexible top and back plates, which can be changed quickly. A group of different tops was tested to show the effect of changes in mass and stiffness. Additionally, different soundhole geometries were used. In order to change the interior volume of the body, rigid plates were inserted into the body cavity. We report on the ability of tuned discrete models to account for changes to the geometry and the structure. Further, we show the experimentally determined trends in dynamic response of the guitar fixture.

3:10

4pMUa7. Four mass coupled oscillator guitar model. John E. Popp (Dept. of Phys. (retired), Moraine Valley CC, 9000 W. College Pkwy., Palos Hills, IL 60465-2478, johnepopp@cox.net)

Coupled oscillator models have been used for the low-frequency response (50–250 Hz) of a guitar [G. Caldersmith, "Guitar as a reflex enclosure," J. Acoust. Soc. Am. **63**, 1566–1575 (1978); O. Christensen, "Quantitative models for low-frequency guitar function," in 103rd ASA Spring Meeting (1982) and J. Guitar Acoustics **6**, 10–25 (1982); T. D. Rossing, J. Popp, and D. Polstein, "Acoustical response of guitars," in Proceedings of Stockholm Music Acoustics Conference, Royal Swedish Academy of Music (1985), pp. 311–332]. These 2 and 3 mass models correctly predict measured resonance frequency relationships under various laboratory boundary conditions, but did not always represent the true state of a guitar in the players hands. The model presented has improved these models in three ways: (1) a fourth oscillator includes the guitar body; (2) plate stiffnesses and other fundamental parameters were measured directly and effective areas and masses were used to calculate the responses, including resonances and phases, directly; and (3) neck vibrations were calculated and measured and shown to effect the response significantly in some guitars. The calculated and measured resonances and phases agree reasonably well.

Contributed Papers

3:30

4pMUa8. Internal visualization of finished guitars and their sound. J. A. Torres (Lab. Acust. Apl. y Vibr. CCADET UNAM, Circuito Exterior s/n CU, Coyoacan, DF, CP04510, Mexico, jesusalejandrott@yahoo.com.mx), A. Granados, G. Hernández, and A. García (Club Lauderos A. C., Paracho Michoacán, Mexico)

The internal structure of two classical guitars (using a near infrared transmittance technique) and typical sound spectra from each were compared. Both guitars were crafted by Paracho luthiers: one with carbon fiber reinforcements, and the other one with traditional Torres design. For the experiments, one fluorescent light bulb was inserted through the soundhole, and after, it was substituted with an incandescent one. The internal structure of each guitar was photographed, using a standard digital camera and a sony camcorder with infrared cut filter (nightshot). A typical song fragment was played in each single guitar, whereas the same experimental conditions were kept. Meanwhile, the corresponding mono record was captured. Sound spectra were averaged using selected fragments from the audio files. The images revealed the internal structure, which were useful in inspecting the internal guitar designs. Spectra of different song fragments played in the same guitar were very similar. However, the spectra did change when playing the same fragments using the other guitar. The procedures here explained allowed are to notice important differences in guitars using inexpensive and easy-to-implement techniques. This infrared photography technique was shown to be a significant improvement in comparison with optical photography; furthermore, it leads one to explore the whole soundbox structure. [Work supported by the Faculty of Engineering (UNAM).]

3:45

4pMUa9. Studying guitar acoustics with modular software synthesizers. Michael Steppat (Beuth Univ. of Appl. Sci., Luxemburgerstr. 10, 13353 Berlin, Germany, steppat@beuth-hochschule.de)

The contructional details of the guitar top plate have a great influence on the percepted sound quality of the instrument. Modeling the whole body with the finite element method provides a good analysis of the mechanical frequency response, with respect to the different spacings of the struts, though it is difficult to calculate it in real time. Modular software synthesizers can be used to simulate the mechanical behavior of the particular components, such as the strings, top-plate, back-plate, and the sound hole. The vibration of each coupled component is calculated real time by a numerical solution of the related differential equations. The results can be used for listening tests to determine the influence of the different individual components.

4:00

4pMUa10. Acoustical differences between treble guitar strings of different tension (i.e., gauge). Michael Faherty and Neil L. Aaronson (School of Natural Sci. and Mathematics, The Richard Stockton College of NJ, P.O. Box 195, Pomona, NJ 08240, neil.aaronson@stockton.edu)

The thickness of a guitar string has a complicated effect on the overall sound of the instrument, affecting the degree of inharmonicity in the tone. [Fletcher *et al.*, (1991).] Different players also select strings of certain tensions because of their perceived qualities such as brightness and loudness. The acoustical differences in the sound and sustain of plucked nylon guitar strings as a function of string tension were measured in an effort to quantify these differences. The open string, fifth-fret, and 12th-fret harmonic waveforms of the treble (G, B, and E) strings were studied for sets of strings with different tensions. Only treble strings, made of a monofilament pure nylon strand, were studied. The levels of the first ten harmonics relative to the level of the fundamental were measured for each sound on each string, as well as the duration of the sustain in each case. Patterns in the sustain times and spectra as a function of string gauge will be presented. [Research supported by the Richard Stockton College of New Jersey.]

4:15

4pMUa11. Acoustic guitar microphone techniques. Sam Ortallono (MediaTech Inst., 3324 Walnut Bend Ln., Houston TX 77042, samortallono @yahoo.com)

This paper will discuss acoustic properties of microphone techniques for the unamplified guitar. By varying only microphone placement and the acoustic properties of the space, we will compare the effects on frequencies captured and reproduced. On and off axis data will be presented as well as those centered on the hole versus the twelfth fret. Also, the recording environment will be tested in both the more and less reverberant states. Mainly the frequency differences will be outlined but some attention will be paid to decay times and constructive or destructive interference. Finally, some examples in context will be delivered to illustrate the strengths and weaknesses of an application for a specific situation. Only one microphone will be presented, as we are only interested in the properties of the placement and the environment rather than microphone choice. No matter the selection of equipment at hand, these techniques will provide guidance to achieve the desired outcome.

THURSDAY AFTERNOON, 18 NOVEMBER 2010 — GRAND CORAL 1A, 5:00 TO 6:00 P.M.

Session 4pMUb

Musical Acoustics: Classical Guitar Concert with Juan Carlos Laguna

Miguel Zenker, Cochair
Escuela Nacional de Musica, NAUM, Mexico City 04100, Mexico

Thomas D. Rossing, Cochair
Dept. of Music, Stanford Univ., Stanford, CA 94022

A concert featuring Juan Carlos Laguna, one of the foremost guitarists of Mexico, will include music from the Renaissance to recent works. The program will include music by Ponce, Brouwer, Villa-Lobas, Pujol and Kantor.

Session 4pNS

Noise: Noise in Latin America and Other Developing Countries

Jorge P. Arenas, Cochair
Inst. of Acoustics, Univ. Austral de Chile, P.O. Box 567, Valdivia, Chile

Sergio Beristain, Cochair
Mexican Inst. of Acoustics, P.O. Box 12-1022, Narvarte 03001 DF Mexico City, Mexico

Chair's Introduction—1:25

Invited Papers

1:30

4pNS1. Noise in the largest Mexican city. Sergio Beristain (Acoust. Lab., ESIME, IPN, IMA, P.O. Box 12-1022, Narvarte, Mexico City, Mexico, sberista@hotmail.com)

Mexico City is one of the largest cities in the whole world, and so it faces many of the contamination problems such as ozone, garbage, dust, and vibration, among others. Noise has spread along the city far beyond the industrial areas, the airport area which is now within the city limits, or the main traffic streets and fast roads. It can be found in commercial areas trying to attract customers passing by the streets, and also inside large commercial centers, living areas from most economical status, through popular parks and squares, which are supposed to be used for relaxation, but many of them have become commercial and exhibit areas, not to mention in most recreational sites, where people gather to listen high-level sounds.

1:50

4pNS2. Noise pollution problem in Mexico City. Felipe Rolando Menchaca Garcia (IPN, P.O. Box 12-1022, Narvarte 03001, Mexico City, Mexico, fmenchac@ipn.mx)

Noise pollution in Mexico City is not a solved problem yet. In 1976 a national regulation was issued to prevent and reduce noise emitted by fixed and mobile sources, but it has not been implemented. This paper analyzes the reasons why noise pollution for Mexico citizen's welfare is not the main concern, why regulations have not been applied, and what are the results of absence of attention to this problem. The impact of Mexico City Airport, main roads, and construction practices are unknown and has to be studied and reflected in regulations and government policies. Recommendations of this paper include these main aspects of the problem.

2:10

4pNS3. Which are the main management issues about noise in Uruguay 2010? Alice Elizabeth González (Dpto. Ingeniería Ambiental, F. de Ingeniería UdelaR, J. Herrera y Reissig 565, CP 11.300, Montevideo Uruguay, aliceelizabethgonzalez@gmail.com)

Despite the fact that noise is not part of the management priorities of decision makers in Uruguay, conflicts about noise are being considered with increasing importance in the recent years. Mitigation measures and solutions are expensive and often difficult to implement, so the trend is to increase prevention in order to minimize correction measures. Focusing on management of some decision makers in the last years, it would be inferred that current management priorities in Uruguay are three: (1) The enactment of a National Regulation Act on noise pollution to homogenize the dissimilar criteria handled in the ordinances of the 19 departments in which Uruguay is administratively divided; (2) attention to conflicts generated by nightlife places in the neighborhoods, especially regarding the agglomeration of young people outside the clubs, that not only raise the environmental sound levels with their presence and voices, but also turn on audio equipment in their vehicles and generate other problems of connivance and security; (3) prevention of adverse occupational and environmental effects, which resulted in greater demands on noise impact studies for environmental authorizations, and systematic studies in occupational settings, including acoustic maps and periodic audiometric studies on staff exposed to high noise levels.

2:30

4pNS4. The Fortaleza noise mapping project: A tool for the definition of noise action plans for the airport, the light rail system, and the Ceara musical event. F. Aurelio Chavez Brito (SEMAM, ECPS, Fortaleza, Brazil) and J. L. Bento Coelho (TULisbon, Lisboa, Portugal)

The Fortaleza noise mapping project was set up for the spatial representation of environmental noise indicators to obtain an essential tool to analyze and define strategies for noise pollution control in Fortaleza, Brazil. This is the first large-scale noise map drawn for a large city in Brazil. Noise emissions from the most important sources contributing to the sound environment of the city, namely, road traffic, railway noise, aircraft noise, industrial noise, and noise from entertainment areas, were included. The method followed a hybrid approach, essentially calculation complemented with experimental measurements for validation and calibration. The large-scale noise assessment allowed detailed studies of the noise impact of the Fortaleza International Airport, located well within the urban city area, the impact of the passage of the underground access light rail tunnel on the local soundscape, and the impact of the Ceará musical event, which, though being part of the city cultural program, takes place in central and seaside areas close to a public hospital. These studies will be presented and discussed in the context of a geographical area where the fair climate allows long hours spent outdoors.

2:50

4pNS5. Experiences on noise in Argentina. Nilda Vechiatti and Daniel Gavinowich (Acoust. and Electroacoustic Lab., Faculty of Eng., Univ. of Buenos Aires, Paseo Colon 850, 1063 Buenos Aires, Argentina, laceac@fi.uba.ar)

It is well known that as the number of people living in an urban core increases, the consumption and production of goods and services and, proportionally, the waste generated, including noise as a common pollutant, increases. At the beginning, it might seem that noise pollution is associated with the high level of life in developed countries, but reality shows that exposure to noise in developing countries is not less, because the lack of demand in the quality of the buildings and a suitable framework of the law bring as a consequence that millions of people have no guaranteed good sound quality in their environment and suffer adverse effects on their health. Several decades before many institutions in our country conducted research works, provided services to the community, and participated in the development of standards and laws to prevent and correct noise pollution. But while the authorities of some countries have taken full awareness of the necessity to ensure the acoustic comfort of the people, in Argentina we still have much to do.

Contributed Papers

3:10

4pNS6. A comparative study of traffic noise modeling software. Robin Lin, Jeffrey Russert, Daniel Walsh, and Dominique Chéenne (Dept. of Audio Arts and Acoust., Columbia College Chicago, 33 E. Congress, Rm. 601, Chicago, IL 60605, DCheenne@Colum.edu)

CADNAA, SOUNDPLAN, and TNM are three industry-standard software programs used for road traffic noise modeling. These programs allow for the calculation and mapping of predicted sound levels over selected areas or at specific locations from a user-generated model comprised of sound sources and physical elements such as roads, buildings, etc., that define the environment. CADNAA and SOUNDPLAN also allow the user to select the standard defining the method of calculation. While many studies have been conducted to validate each of the three programs against specific test data, little research is available to compare the three together against a common set of test results. This study compares calculated sound levels in CADNAA 3.7, SOUNDPLAN 7.0, and TNM 2.5 against actual traffic noise levels acquired at nine locations across three sites, each site offering a different level of complexity in terms of modeling the environment. Models were kept highly similar in all three programs and calculation methods used both the FHWA and RLS-90 standards. Results show that no single program was consistently accurate when using either standard, and also highlights the strengths and weaknesses of the two standards.

3:25

4pNS7. Road noise mapping of Medellín´s downtown: Experiences, analysis of different calculation’s methods, and noise prediction software. Diego Mauricio Murillo Gmez (Sound Eng. Program, Univ. de San Buenaventura Medellín, Cra 56 C no 51 90, Medellin 50010, Colombia, diego.murillo@usbmed.edu.co), Carlos Alberto Echeverri Londoño (Enviromental Eng. Program, Univ. de Medellín, Carrera 87 No 30 65), and Germán Mauricio Valencia Hernandez (Environ. Eng. Program, Univ. de San Buenaventura Medellín, Cra 56 c No 51 90)

In this paper we will present the development of road noise map in the downtown of Medellin City through the implementation of predictive software. For the construction of noise mapping, the results obtained using the calculation’s French method (NMPB) and German method (RLS 90) were compared. The prediction process was carried out using two different softwares (SOUNDPLAN and CADNAA) for the purpose of analyzing the differences in the programs and the resulting levels. Finally, noise measurements were made in order to validate the prediction.

3:40

4pNS8. Road noise simulation as a management tool for redesign of public transport routes in the city of Medellin. Diego Mauricio Murillo Gmez (Sound Eng. Prog., Univ. de San Buenaventura Medellín, Cra 56 c No 51-90, Medellin 50010, Colombia, diego.murillo@usbmed.edu.co), Carlos Alberto Echeverri Londoño (Environ. Eng. Prog., Univ. de Medellín), and Germán Mauricio Valencia Hernandez (Univ. de San Buenaventura Medellín)

In this paper we present the process of road noise simulation as an analytical tool for decision-making in the conversion of radial to tranverse routes, designed for public transport in the city of Medellin. To obtain the existing noise pollution level and its subsequent comparison with the level from proposed redesign, a road map in the center of the city through the SoundPLAN software was carried out and various scenarios were predicted based on the analysis of decrease in the number of heavy vehicles in conflicting ways. Finally, air quality parameters, demand of passenger, travel time, and average speed were integrated in order to propose environmentally optimal routes and to improve mobility in the city.

3:55

4pNS9. Noise mapping: A comparison of predicted and measured levels. Steven R. Ryherd, Tina M. Ortkiese, and Kenneth A. Cunefare (Principal, Arpeggio Acoust. Consulting, LLC, Atlanta, GA 30345, srryherd@arpeggioacoustics.com)

In the United States, there are no national regulations requiring noise mapping in cities, and the noise mapping that is conducted is a consequence of local planning and policy bodies responding to or anticipating public demand. The production of oil and natural gas in urban environments has caused local governments surrounding Shreveport, LA, to utilize noise mapping and field measurements to assist in the development of a new noise ordinance. Over 1800 square miles surrounding the city was mapped considering air, rail, and road noise sources. Ten, multi-day measurement locations were strategically selected within the model area. A comparison of predicted and measured sound pressure levels is presented for consideration of future noise mapping methods and procedures.

Session 4pPA

Physical Acoustics: Thirty Years of Resonant Ultrasound II

Albert Migliori, Cochair
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Veerle M. Keppens, Cochair
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Invited Papers

1:30

4pPA1. Mode identified resonant ultrasound spectroscopy. Hirotsugu Ogi (Grad. School of Eng. Sci., Osaka Univ., Toyonaka, Osaka 560-8531, Japan, ogi@me.es.osaka-u.ac.jp)

Resonant ultrasound spectroscopy (RUS) is a powerful tool for determining all independent elastic constants (Cij) of solids. Successful inverse calculation for Cij requires mode identification for measured resonance frequencies. In this lecture, two mode-identified RUS methods developed by the author are presented. One is the electromagnetic acoustic resonance (EMAR) [Ogi *et al.*, J. Acoust. Soc. Am. **106**, 660 (1999)]. Using a solenoid coil and static magnetic field, vibrations of a specimen are measured through the electromagnetic-force mechanisms. Detectable vibrational modes are controlled by changing the magnetic-field direction. Because of the noncontacting measurement, it is applied to measurements at high temperatures easily. The second method incorporates the laser-Doppler interferometry into RUS (RUS-LDI method) [Ogi *et al.*, J. Acoust. Soc. Am. **112**, 2553 (2002)]. The specimen is set on the needle piezoelectric tripod in vacuum, and its displacement distributions at resonances are measured by laser interferometry for mode identification. It was applied to determine the piezoelectric coefficients and internal-friction tensors as well as Cij of various piezoelectric materials, and also to thin-film Cij. Because the contributions of the piezoelectric coefficients and thin-film Cij to resonance frequencies are very small, correct mode identification becomes significantly important in their determination.

2:00

4pPA2. Resonance spectroscopy: Time versus frequency and open versus closed systems. John A. Scales (Dept. of Phys., Colorado School of Mines, Golden, CO 80401, jscales@mines.edu) and Martin L. Smith (Blindgoat Geophys., Sharon, VT 05065)

The existence of resonance in any system derives from the presence of periodic wave propagation. Such periodicities can exist in closed systems [traditional resonance ultrasound spectroscopy (RUS)] or open systems (e.g., lasers), and they can be analyzed either in the time or frequency domain. From this broader perspective we can see that diverse techniques such as RUS, coda wave interferometry, diffuse wave spectroscopy, cavity ring-down spectroscopy, cavity perturbation, etc., share many common physical principles: propagating energy is trapped, at least temporarily, and allowed to repeatedly sample a medium. It is this repeated sampling that gives spectroscopy and interferometry their great sensitivity. In this paper we will show a number of examples of RUS or RUS-like techniques applied to the characterization of materials in the laboratory. Sometimes the waves will be mechanical and sometimes they will be electromagnetic, but the unity of the physical principles involved allow us to apply the lessons learned from RUS to a wide range of problems.

Contributed Papers

2:30

4pPA3. Measurement schemes for resonant ultrasound spectroscopy experiments at high temperatures. J. R. Gladden, III, G. Li, R. Adebisi, and S. Male (Dept. of Phys. and Astronomy, Univ. of Mississippi, University, MS 38677, jgladden@olemiss.edu)

The heart of any acoustic measurement is the conversion of vibrational energy into electrical energy. One of the most common and efficient methods of this energy transduction is the use of piezoelectric elements such as PZT or lithium niobate. At temperatures above about 600 °C, all traditional piezoelectric materials lose their piezoelectric properties, requiring the use of buffer rods to separate the sample in the hot zone and the transducers. Buffer rods introduce experimental difficulties such as reduced signal to noise, sample loading effects, and extraneous buffer rod resonances. We will report on experimental details of RUS measurements at temperatures up to 1000 °C including the development of a noncontact optical lever system of resonance detection. Data showing the acoustic signature of phosphorus precipitation in silicon germanium and phase transitions in Zintl phase YMS thermoelectric materials will be presented.

2:45

4pPA4. Elastic rays and coda wave interferometry. Thomas Gorin and Adrian Ortega (Dept. of Phys., Universidad de Guadalajara, C.P. 44840, Guadalajara, Mexico)

We apply the concept of rays, well known from geometric optics, to elastic waves traveling in a finite solid. Ray splitting and mode conversion between compressional *P*-waves and transversal *S*-waves are taken into account by using a random walk procedure and Monte-Carlo integration. In addition, we use the diagonal approximation from semi-classical theory to take the interference between rays traveling along different paths into account. Our approach allows us to simulate interference effects in the propagation of diffuse Coda waves in isotropic solids of different geometric shapes. We use that method to describe recent experiments by Lobkis and

Weaver [Phys. Rev. E **78**, 066212 (2008)] on the scattering fidelity in elastodynamic systems.

3:00

4pPA5. Experimental measurement of the elastic tensor of single crystal rare earth scandates utilizing resonant ultrasound spectroscopy. Eric Scott and Kenneth Pestka, II (Dept. of Phys., Rollins College, 1000 Holt Ave., Winter Park, FL 32789)

Using resonant ultrasound spectroscopy (RUS) in combination with *ab initio* calculations, the complete elastic tensor of four rare earth scandates, $SmScO_3$, $TbScO_3$, $NdScO_3$, and $EuScO_3$, was determined. These recently developed materials are useful for a variety of novel applications including strain-induced thin film growth of superlattices used in sonic lasers. In these applications, the dynamic lattice properties are of importance and are directly related to the elastic constants. The RUS method for determining the elastic constants is accomplished by iteratively comparing the experimentally determined resonant frequencies of a sample with a set of calculated frequencies. These calculated frequencies are generated from an assumed set of elastic constants. In order for the RUS method to reproduce the elastic tensor of the sample, reasonable initial elastic constants are required. For these newly developed materials, the initial elastic constants were obtained using *ab initio* calculations. Utilizing RUS, the experimentally determined elastic constants differed, on average, by less than 10% from the theoretical values.

3:15

4pPA6. Resonant ultrasound spectroscopy of sandstone. Cristian Pantea, Blake T. Sturtevant, and Dipen N. Sinha (MPA-11, Los Alamos Natl. Lab., MS D429, Los Alamos, NM 87545, pantea@lanl.gov)

Sandstones are typically found in geothermal and oil and gas reservoirs. A precise determination of the elastic moduli of sandstone is very important toward understanding the reservoir properties, especially when hydro-fracturing needs to be performed. Hydro-fracturing consists of injecting high-pressure water in the bedrock formation to create bedrock fractures that connect to existing fractures in the formation, resulting in an increased flow of water, oil, or gas. Elastic properties of various types of sandstones are investigated in this work using resonant ultrasound spectroscopy (RUS). The porosity and very low elastic moduli of sandstones introduce additional challenges in the determination of the elastic moduli when compared to materials previously investigated by this technique. Conventional RUS experiments are difficult to perform on such porous, brittle, and attenuating samples, and a re-designed experimental setup is required that can deliver more acoustical energy to the sample. A detailed description of the approach will be discussed, including transducer selection, experimental setup materials, and sample geometry. Temperature dependence of sandstone elastic moduli for temperatures typical of geothermal and oil and gas reservoirs is investigated. In addition, high pressure–high temperature determination of the elastic moduli will be described.

3:30

4pPA7. Application of resonant acoustic spectroscopy to beam shaped asphalt concrete samples. Anders Gudmarsson (Dept. of Civil and Architectural Eng., KTH Royal Inst. of Technol., Brinellvägen 34, 100 44 Stockholm, Sweden, anders.gudmarsson@byv.kth.se), Nils Ryden (Lund Univ., 221 00 Lund, Sweden), and Björn Birgisson (KTH Royal Inst. of Technol., Brinellvägen 34, 100 44 Stockholm, Sweden)

The dynamic modulus of asphalt concrete is a key parameter needed in modern pavement design and management. Traditional laboratory tests based on cyclic loading (0.1–25 Hz) at different testing temperatures are time consuming and require expensive equipment. There is therefore a need for more efficient non-destructive methods to determine the dynamic modulus of asphalt concrete. This study applies resonant acoustic spectroscopy (RAS) to beam shaped asphalt concrete samples. Multiple modes of vibration are measured at each testing temperature using a miniature accelerometer and a small steel sphere as impact source. The complex modulus from each resonant frequency is calculated using the Rayleigh–Ritz method. The heterogeneous and viscoelastic nature of asphalt concrete presents challenges to the application of conventional RAS. The number of measurable modes decreases with increasing test temperature. In an attempt to extend the usable frequency and temperature range measured, transfer functions are inverted using the finite element method along with a frequency dependent complex modulus. Initial results indicate that RAS can be an efficient method for the prediction of the high-frequency part of the asphalt concrete dynamic modulus mastercurve.

THURSDAY AFTERNOON, 18 NOVEMBER 2010 GRAND CORAL 3, 1:00 TO 5:00 P.M.

Session 4pPP

Psychological and Physiological Acoustics: Detection, Discrimination, and Spatial Hearing (Poster Session)

Daniel J. Tollin, Chair

Dept. of Physiology, Univ. of Colorado Medical School., 12800 E. 19th Ave., Aurora, CA 80045

Contributed Papers

All posters will be on display from 1:00 p.m. to 5:00 p.m. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 1:00 p.m. to 3:00 p.m. and contributors of even-numbered papers will be at their posters from 3:00 p.m. to 5:00 p.m.

1:00

4pPP1. Single- and dual-channel rate discrimination by cochlear implant listeners. Antje Ihlefeld and Robert P. Carlyon (MRC Cognition & Brain Sci. Unit, 15 Chaucer Rd., Cambridge CB2 7EF, UK, antje.ihlefeld@mrc-cbu.cam.ac.uk)

Experiment 1 measured single-electrode rate-discrimination for two different electrodes (4 and 8) as a function of modulation depth. Stimuli were sinusoidally amplitude modulated, 200-ms monopolar biphasic pulse trains with a 5000-pps carrier rate, and an envelope frequency standard of 100 Hz, presented in a 2I-2AFC paradigm. Four cochlear implant listeners (CILs), users of MedEL Pulsar processors, participated. Performance generally improved with increasing modulation depth and was better on electrode 4 than on electrode 8. Using pairs of modulation depths with similar single-electrode performances, experiment 2 examined the combination of rate information across these two electrodes, when both electrodes were stimulated simultaneously with sinusoidal amplitude modulated pulse trains whose envelopes were either in phase or out of phase across electrodes. CILs-1 and -2 performed worse in the out-of-phase condition than in the in-phase condition. CIL-3 performed better in the out-of-phase condition than in the in-phase condition. CIL-4 showed no difference. Although performance was sometimes better with dual-channel than with single-channel stimulation, results do not reveal a substantial and consistent benefit from combining information across electrodes. However, when modulation is applied to more than one electrode, results show that the phase relationship of envelopes across electrodes can affect a CIL's ability to process modulation rate.

4pPP2. A comparison of three models for the principal pitch of vibrato tones. Pablo Etchemendy, Bruno A. Mesz, and Manuel C. Eguia (Laboratorio de Acústica y Percepción Sonora, Univ. Nacional de Quilmes, R. S. Peña 352, Bernal, 1876 Buenos Aires, Argentina)

Vibrato is the more ubiquitous manifestation of a non-stationary tone that can evoke a single overall pitch, called principal pitch. Some recent results using non-symmetrical vibrato tones suggest that the perceived pitch could be governed by some stability-sensitive mechanism. A recently proposed one-parameter model based on time-frequency reassignment was able to predict previously reported results. [B. Mesz and M. C. Eguia, Ann. N.Y. Acad. Sci. **1169**, 126–130 (2009)]. The proposed model has some biological plausibility since the local computations at each auditory channel needed for pitch estimation are based on phase information, so they could be implemented in principle using the time interval between action potentials in the auditory nerve fibers. In order to test this new model against two previously formulated models of vibrato pitch [H. Gockel *et al.*, J. Acoust. Soc. Am. **109**, 701–712 (2001)] and pitch saliency [Cariani. Proc. VIII Intl. Conf. Music Perception & Cognition, Evanston, IL (2004)], a psychophysical experiment using vibratos with fast and irregular frequency modulation, which yield quite different pitch predictions in the three models, is performed.

4pPP3. Effects of conflicting auditory information on selective attention to pitch. Blas Espinoza-Varas (Commun. Sci. & Disord., OU Health Sci. Ctr., 1200 N. Stonewall Av., Oklahoma City, OK 73117)

We studied how conflicting information interferes with selective attention to task-relevant differences in pure-tone frequency. Across the observation intervals of the discrimination task, the relevant frequency differences between target tones were positive, but within an observation interval, they could appear to be small or negative relative to conflicting differences in flanker tones leading or trailing the target. Being correct required attending to the between-target and ignoring the target-flanker pitch relation (across and within observation-interval, respectively). The interference index was an elevation of conflict-laden frequency discrimination thresholds (FDTs), relative to no-conflict FDTs. When conflicting differences in frequency or level trailed the relevant differences, interference was large and persistent, increased with the target-flanker time proximity, but decreased with extensive training. Interference occurs when the target-flanker pitch relation is more prominent than the one between targets, and the physical and/or perceptual effects of relevant and conflicting differences tend to cancel one another, as with the above conflicting differences. With untrained participants, the target-flanker pitch relation is most prominent in conditions fostering both the perceptual grouping of the target and flanker and the recency and salience of the conflicting differences; conversely, by lessening such grouping and salience, prolonged training decreases or nullifies the interference.

4pPP4. The effects of ear of stimulus presentation on pitch discrimination abilities of young adult females. Ashley R. Gaal, Robert E. Moore, and Julie M. Estis (Dept. of Speech Pathol. and Audiol., Univ. of South Alabama, 307 N. Univ. Blvd. Mobile, AL 36688, jestis@usouthal.edu)

Previous research has examined various factors that influence pitch discrimination abilities in normal-hearing listeners [Deutsch (1970); (1978)], including the influence of spatial location of input stimuli. Separating the input location of interference tones from the initial (reference) and final (comparison) tones leads to better pitch discrimination accuracy (PDA) than when all tones are presented to the same ear, but the effects of the relationships between these three tone types have not been explored. This study examined the impact of ear of stimulation in 20 conditions: 5 containing no interference and 15 containing 4 interference tones. Twenty-four nonmusician females (age 19–30) with normal hearing participated. Results showed that performance was significantly better when (a) no interference was present, (b) the comparison tone was presented contralaterally to reference and interference tones, (c) the comparison tone was presented to the left ear, and (d) the reference and comparison tones were the same frequency. Overall, the comparison tone seems to play a key role in PDA. These findings should be further explored, as they could be utilized in aural training and therapy programs to strengthen pitch discrimination in musicians and clinical populations, such as individuals with prosodic language or auditory processing deficits.

4pPP5. Temporal fine structure for sequential segregation. Nicolas Grimault (UMR CNRS 5020, Université Lyon 1, 50 av. T. Garnier, 69366 Lyon, France) and Etienne Gaudrain (MRC Cognition and Brain Sci., Cambridge CB2 7EF, UK)

Pitch based sequential segregation is reputed to benefit from spectral cues and temporal envelope cues. However, some previous results with hearing impaired listeners raised the possibility that other cues could promote segregation. These other cues could be responsible for the high intersubject variability across hearing impaired listeners in streaming experiments. Temporal fine structure cues appeared then to be a valuable candidate. The current study is dedicated to assess the potential benefit from temporal fine structure cues for sequential segregation. Unresolved complex tones (A) and unresolved complex tones with the same F0 in which all components were shifted upward by the same number of Hertz (B) were presented in ABA-ABA- sequences. Using these stimuli, a rhythm discrimination task was used to estimate the amount of primitive segregation. Cochlear models will be applied first to the stimuli to establish the respective amount of spectral, temporal envelope and temporal fine structure cues. The results from the discrimination task will then be discussed at the light of the outputs of the models.

4pPP6. Localizing amplitude-modulated high-frequency tones in free field: Physical. William M. Hartmann, Brad Rakerd, and Eric J. Macaulay (Michigan State Univ., East Lansing, MI 48824, hartmann@pa.msu.edu)

Headphone experiments since the 1950s have shown that high-frequency tones can be lateralized if they are given low-frequency amplitude modulation (AM). Lateralization is mediated by the interaural time difference (ITD) of the amplitude envelope, and it resembles lateralization for a comparison sine tone having the frequency of the modulation in the AM tone. However, lateralization is distinct from localization because there are significant physical differences between the idealized AM signals in the headphone experiments and the signals arriving at a listener's ears in free-field presentation. In free field, (1) the modulation percentage becomes different for the two ears, (2) a purely amplitude-modulated source inevitably acquires some frequency modulated component, (3) signals are sometimes overmodulated in one or both ears. Also, the ITD of the AM envelope is generally less than the ITD in the comparison low-frequency sine tone because the envelope ITD depends on the group delay, not the (larger) phase delay. Nevertheless, it is known that listeners can use the envelope ITD to localize high-frequency AM tones presented in free field. [E. J. Macaulay *et al.*, J. Acoust. Soc. Am. **127**, 1440–1449 (2010).] [Work supported by the NIDCD, Grant DC 00181.]

4pPP7. A role for low-frequency sensitive neurons in the auditory brainstem in the encoding of source location distance-dependent interaural level difference cues. Daniel J. Tollin, Heath G. Jones, Jennifer L. Thornton, and Kanthaiah Koka (Dept. of Physio., Univ. of Colorado Med. Sch., RC1N, 8307, Box 6511, 12800 E 19th Ave., Aurora, CO 80045, daniel.tollin@ucdenver.edu)

The duplex theory posits that low- and high-frequency sounds are localized using two different acoustical cues, interaural time delays (ITDs) and interaural level differences (ILDs), respectively. Psychophysical data have generally supported the theory for pure tones. Anatomical and physiological studies have revealed two parallel brainstem pathways that appear to encode ITDs and ILDs separately. ITDs are extracted by medial superior olive neurons. ILDs are extracted by lateral superior olive (LSO) neurons. ILD-sensitive neurons are also found in the inferior colliculus (IC). ILDs are a complex function of both source location and frequency such that lower and higher frequencies exhibit smaller and larger ILDs, respectively. LSO and IC neurons encode ILDs for high-frequency sounds where the cues are physically available, but there are discrepancies regarding low-frequency neurons. Although acoustically, low-frequency ILDs are small, humans are sensitive to them and physiological studies have found low-frequency neurons in the LSO and IC that could encode them. Here the hypothesis that low-frequency ILDs are useful when sound source distance is varied is

explored. These data demonstrate that a population of IC neurons is sufficient to encode the range of acoustic ILDs that would be experienced as a joint function of source location and distance.

4pPP8. The role of spectral cues and minimum bandwidth in the auditory perception of distance. Ramiro Oscar Vergara Ferrari, Esteban Calcagno, and Manuel C. Eguia (Lab. de Acústica y Persepción Sonora, Univ. de Quilmes, Roque Sáenz Peña 352, Bernal, B1876BXD Buenos Aires, Argentina, ramirovergara@hotmail.com)

Little is known about the strategies that humans use to estimate the distance of a sound source. Variations in the sound pressure level and changes in the sound induced by its relationship with the acoustic environment have been proposed as important cues in the judgment of the apparent distance of a sound source. However, relatively little attention has been given to the role of frequency spectrum in the determination of auditory distance. In this work, psychophysical experiments on distance perception using auditory stimuli with different spectral characteristics are performed. The first series of experiments show that subjects perceive better the changes in the distance of the source for spectrally complex auditory stimuli compared to pure tones. In the second set of experiments, noise bands with bandwidths located on different parts of the frequency spectrum were used. From this experiment, the minimum bandwidth required for a sound signal in order to preserve information on the distance to the acoustic source is obtained as a function of frequency. These results show that the spectral characteristics of sounds can significantly influence the perception of the distance to a sound source.

4pPP9. Systematically shifting the phase of bilateral bone-conducted sound changes perceived loudness. Ross W. Deas, Robert B. Adamson, Laura L. Curran, Philip P. Garland, Manohar Bance, and Jeremy A. Brown (SENSE Lab., 1276 South Park St., Rm. 3189 Dickson Bldg., VG Site QEII Health Sci. Ctr., Halifax, NS B3H 2Y9, Canada)

Signals transmitted via bone conduction transducers travel through bone and reach both cochleae. Applying the same bone signal to different sides of the head means that signals cross the skull and potentially apply opposing forces to each other. However, if the phase of one transducer were to be shifted relative to the other, it is possible that the two signals might result in complementary forces. This study systematically manipulated the phase of one signal relative to the other, and measured the change in perceived loudness. For a wide range of frequencies (250 Hz–10 kHz), participants (n=5) were presented with pure tones that sequentially swept through the full range of phase offsets (0–2 pi) and judged whether each tone was louder or quieter than the preceding tone. Data showed a clear sinusoidal variation in perceived loudness with phase. Phase-offset that yielded maximum perceived loudness varied systematically with frequency. Audiological thresholds were obtained for optimum and pessimum phase-offsets at each frequency in a blind experimental design. Differences as large as 10 dB were found between the different conditions. We argue that this is an important finding about the nature of bone-conducted sound, with important clinical implications for patients who use bone conduction hearing devices.

4pPP10. Sensitivity to characteristics of Gaussian-shaped stimulus distributions in auditory categorization. Sarah C. Sullivan, Megan M. Kittleson, Andrew J. Lotto (Dept. of Speech, Lang. and Hearing Sci., Univ. of Arizona, 1131 E. 2nd St., Tucson, AZ 85721, scsully@email.arizona.edu), and Randy L. Diehl (Univ. of Texas, Austin, TX 78712)

The purpose of this experiment was to further investigate the role that distributional information plays in auditory categorical decision tasks. Participants were presented non-speech sounds randomly sampled from two overlapping, Gaussian-shaped distributions. The sounds consisted of 27 narrow-band noise bursts varying in center frequency from 985–1375 Hz. Each of the four conditions was composed of a distribution with a relatively large variance and one with a relatively small variance, resulting in different distributional cross-over points or ideal boundaries. Participants were presented the stimuli over headphones as part of a computer game in which they navigated through 3-D space, reacting to animated alien characters along the way. Each distributional sound category corresponded to a specific character; either a blue alien or a green alien. The participants were asked to use a keyboard to identify the alien best associated with the sound they heard. Performance was tracked by plotting identification functions and noting boundary placement for each individual block. After only six blocks of training (approximately 25 min), most subjects had established boundaries near the distributional cross-over points (i.e., optimal category boundaries). The present experiment lays the foundation for a distributional approach to auditory categorization. [Work supported by NIH.]

4pPP11. A listening test method to evaluate the dissimilarity of large sets of stimuli: Application to loudspeakers. Pierre-Yohan Michaud, Sabine Meunier, Philippe Herzog (Laboratoire de Mécanique et d'Acoustique, CNRS UPR 7051, 31 Chemin Joseph Aiguier, 13402 Marseille Cedex 20, France, michaud@lma.cnrs-mrs.fr), Mathieu Lavandier (Ecole Nationale des Travaux Publics de l'Etat, 69518 Vaux-en-Velin, France), and Gérard Drouet d'Aubigny (Laboratoire Jean Kuntzmann, 38040 Grenoble, France)

The paired comparison method is usually recommended as the standard method to collect judgments of dissimilarity between audio stimuli. However, this method is only appropriate for similarity ratings involving a small number of stimuli, because the number of pairs to be tested in the experiment rapidly increases with the number of stimuli, thereby leading to very demanding tests. A method adapted from a visual perception study will be presented, which allows one to deal with large sets of stimuli. For each trial, the task consists in selecting one stimulus among three as the most similar to a reference stimulus. Simulations of "perfect" listeners' judgments have been conducted to estimate the influence on the results of the numbers of stimuli and listeners. These simulations investigated the potential artifacts related to the proposed protocol. Listening tests have been conducted to evaluate the timbre restitution by 12 loudspeakers with the proposed method, for comparison with the results of Lavandier *et al.* (2008) involving paired comparisons. Both methods provided correlated dissimilarity data, and their MDS analysis yield to similar 2-D spaces. The proposed method allowed us to evaluate 37 loudpseakers. The MDS analysis provided a 3-D space including two dimensions common to the first panel.

4pPP12. Avoiding measurement artifacts in assessing the detection of "expected" and "unexpected" signals. Molly J. Henry, J. Devin McAuley (Dept. of Psych., Michigan State Univ., East Lansing, MI 48824), William M. Hartmann (Dept. of Phys., Michigan State Univ., East Lansing, MI 48824), and Timothy J. Pleskac (Dept. of Psych., Michigan State Univ., East Lansing, MI 48824)

Auditory perception studies using the probe-signal method [Greenberg and Larkin (1968)] have demonstrated greater detection sensitivity for tones occurring at "expected" frequencies relative to tones occurring at "unexpected" frequencies. An important, but largely overlooked, methodological issue in studies using the probe-signal and related methods is that measured sensitivity (d´) can underestimate true sensitivity for conditions with relatively few observations, implying that enhanced sensitivity observed for "expected" conditions with more observations compared to "unexpected" conditions with fewer observations can potentially be due to a measurement artifact. To address this possibility, the current research combines results of two behavioral studies with a set of simulations and finds that heightened sensitivity to an expected signal relative to an unexpected signal can be observed when there are no differences in true sensitivity across conditions. Specifically, enhanced detection sensitivity for more frequent signals attributable to a measurement artifact is more likely when true detection sensitivity is high, the number of stimulus presentations is relatively low, and a correction for response proportions of 0 and 1 is required to calculate d'. Implications are discussed for psychoacoustics research using the probe-signal method and more broadly for psychological studies examining effects of listener expectations on perception.

4pPP13. Efficient coding of attenuated correlation among complex acoustic dimensions. Christian E. Stilp, Timothy T. Rogers, and Keith R. Kluender (Dept. of Psych., Univ. of Wisconsin, 1202 W. Johnson St., Madison, WI 53706, cestilp@wisc.edu)

If sensorineural systems are efficient, redundancy should be extracted to optimize information processing. Stilp *et al.* reported efficient coding of a robust correlation ($r^2 = 0.95$) among complex acoustic attributes following passive exposure [J. Acoust. Soc. Am. **124**, 2493 (2008)] and through continuous active testing [J. Acoust. Soc. Am. **126**, 2203–2204 (2009)]. Across studies, discrimination of differences between sounds orthogonal to the correlation is initially poor relative to sounds consistent with the correlation.

Only following continued testing, listeners discovered variance inconsistent with the correlation, and discrimination of orthogonal sounds improved. Present studies examine how strong a correlation must be to create these perceptual changes. Listeners discriminated stimuli (AXB) for which the correlation between two complex independent dimensions, attack/decay (AD) and spectral shape (SS), varied. Poorer discrimination of differences that do not respect the correlation persists when the range over which AD/SS covary is truncated ($r^2 = 0.65$), and when orthogonal sounds are modestly oversampled ($r^2 = 0.89$). When correlations are decreased by extreme oversampling of orthogonal sounds ($r^2 = 0.69$) or addition of orthogonal sounds with more extreme values ($r^2 = 0.69$), effects diminish. Connectionist models employing principal components analysis consistently predict listener performance across experiments. [Work supported by NIDCD.]

4pPP14. Relationship between medial-olivocochlear reflex strength and masked thresholds. Angela C. Garinis, Lynne A. Werner (Speech and Hearing Sci., 1417 N.E. 42nd St., Seattle, WA, 98105, agarinis@u.washington.edu), and Carolina Abdala (House Ear Inst., Los Angeles, CA, 90057)

Otoacoustic emission amplitude is reduced by contralateral acoustic stimulation (CAS). This effect is produced by the medial-olivocochlear (MOC) reflex. Past studies have shown that the MOC reflex is related to listening in noise and selective attention. The present study examined the relationship between the MOC reflex and masked thresholds in 14 normally hearing adults. Detection thresholds were determined for a 1000-Hz, 300-ms tone presented simultaneously with a 300-ms masker. Three masking conditions in which energetic or informational masking predominate were investigated. DPOAEs were evoked from 500–4000 Hz with primary tones swept in frequency at 8 s/oct, using a fixed f2/f1 ratio of 1.22 at 65/55 dB sound pressure level. The MOC reflex and MOC-shift were measured at DPOAE fine structure maxima at 948 and 1091 Hz with and without CAS. An inverse fast-Fourier transform was performed to evaluate MOC effects on individual DPOAE components. Preliminary data analyzes showed MOC activity to be correlated with masked thresholds for both indices of medial efferent function. Most notably, higher masked thresholds for the random and broadband conditions were associated with a stronger MOC reflex for DPOAE maxima around 948 Hz. These results suggest that MOC activation plays a role in selective listening.

4pPP15. The use of the kurtosis metric in the evaluation of industrial noise exposures. Roger Hamernik, Wei Qiu, and Robert Davis (Auditory Res. Lab., State Univ. of New York at Plattsburgh, 101 Broad St., Plattsburgh, NY 12901)

Data from a chinchilla model of noise-induced hearing loss will demonstrate the value of using the statistical metric, kurtosis, along with an energy metric to evaluate complex industrial noise environments for hearing conservation purposes. Complex noises are non-Gaussian noises consisting of a combination of Gaussian continuous noise and noise transients, either impacts or noise bursts. Chinchillas were exposed to an interrupted and intermittent 97-dB sound pressure level broadband, non-Gaussian noise for 8 h/day for 3 weeks. The experimental noises were designed to model an industrial exposure. Groups of animals with 8/group were exposed to 1 of 12 different complex noise environments having a fixed spectrum and energy but with different temporal structures achieved by varying the transient peak, interval, and duration histograms. Preliminary results show that (1) hearing and sensory cell loss increase with increasing kurtosis despite a fixed energy; (2) at a given value of kurtosis, hearing trauma is invariant with respect to changes in the temporal structure of the noise exposure. A metric such as the kurtosis may be a necessary adjunct to spectral energy in the evaluation of industrial noise exposures for the purposes of hearing conservation. [Research supported by NIOSH Grant R01 OH 002317.]

THURSDAY AFTERNOON, 18 NOVEMBER 2010 — CORAL KINGDOM 2/3, 2:00 TO 3:30 P.M.

Session 4pSA

Structural Acoustics and Vibration, Engineering Acoustics, and Underwater Acoustics: Acoustic Metamaterials II

Michael Haberman, Cochair
Applied Research Labs., Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758

Stephen D. O'Regan, Cochair
NSWC Carderock Div., 9500 MacArthur Blvd., West Bethesda, MD 20874

Thomas R. Howarth, Cochair
NAVSEA Newport, 1176 Howell St., Newport, RI 02841

Invited Papers

2:00

4pSA1. Generalized transformation acoustics for material design. Jeffrey Cipolla, Nachiket Gokhale (Weidlinger Assoc. Inc., 375 Hudson St., New York, NY 10012), and Andrew Norris (Rutgers Univ.)

Norris ["Acoustic cloaking theory," Proc. R. Soc. London, Ser. A **464**, 2411–2434 (2008)] presented a theory of transformation acoustics that enables the realization of pentamode acoustic materials having anisotropic density and finite mass. The design of acoustic materials for 3-D objects composed of combinations of simple geometric shapes, e.g., a cylinder with spherical end-caps, is considered. Certain classes of transformations which are well-suited for the design of such shapes are described and conditions under which these transformations result in materials which minimize acoustic reflections are proposed. The concept is validated using 3-D explicit transient finite element simulations, which closely emulate the physics of wave propagation in the proposed materials.

2:30

4pSA2. Acoustic characterization of magnetorheological fluids. Frank Fratantonio, Thomas R. Howarth, Jeffrey E. Boisvert, Anthony Bruno (Naval Sea Systems Command Div. Newport, Newport, RI 02841), Clyde L. Scandrett (Naval Postgrad. School, Monterey, CA 93943), and William M. Wynn (Naval Sea Systems Command Div. Panama City, Panama City Beach, FL 32407)

Magnetorheological (MR) fluids contain magnetic particles dispersed within a host fluid. These materials are considered a type of "smart" fluid in that their viscoelastic properties can be controlled by varying the magnetic field intensity. MR fluids have found favor in high-end automobile applications such as the General Motors dynamic MR suspension system which has been in Corvettes (and other GM products) since 2005. Recently there has been interest in understanding the acoustic properties of MR fluids. In particular, the ability to control the radial and orthogonal bulk moduli suggests that MR fluids are a potential candidate for acoustic metamaterial applications. This presentation will discuss MR fluids and a method for the acoustic characterization of these fluids as functions of frequency and magnetic drive levels. [Work supported by the Office of Naval Research.]

3:00—3:30 Panel Discussion

THURSDAY AFTERNOON, 18 NOVEMBER 2010 GRAND CORAL 3, 1:00 TO 5:00 P.M.

Session 4pSC

Speech Communication: Production and Perception of Spontaneous Speech II (Poster Session)

Ann R. Bradlow, Cochair
Dept. of Linguistics, Northwestern Univ., 2016 Sheridan Rd., Evanston, IL 60208

Valerie L. Hazan, Cochair
Speech Hearing and Phonetic Sci., Univ. College London, 2 Wakefield St., London, WC1N 1PF, U.K.

Contributed Papers

All posters will be on display from 1:00 p.m. to 5:00 p.m. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 1:00 p.m. to 3:00 p.m. and contributors of even-numbered papers will be at their posters from 3:00 p.m. to 5:00 p.m.

4pSC1. Using hyperarticulation to quantify interaction between discourse functions. Valerie Freeman (Dept. of Linguist., Univ. of Washington, C-104 Padelford Hall, Box 354360, Seattle, WA 98195, valerief@uw.edu)

Social factors are known to affect speech production, but in discourse and conversation analytic branches of sociolinguistics, quantitative measures are not as common as qualitative observations. This study uses acoustic measures of hyperarticulation to quantify the effects of two interacting discourse functions: new-information signaling and stance expression. For each of five speakers in an hour-long political talk show, content analysis was performed on all phrases repeated three or more times to separate neutral from stance-expressing tokens and new from given repetitions of those tokens. Word, syllable, and vowel duration were measured from spectrograms; formant (LPC) and pitch (autocorrelation) values were measured at onset, 20%, 50%, 80%, and offset of stressed vowels. Preliminary results from repeated measures analysis of variance suggest that stance is indeed a significant predictor of hyperarticulation which interacts with newness for at least some speakers. This work shows one way that acoustic measures can quantify the relative contributions of interacting discourse variables and their effects on speech variation. The findings also have implications for future work in speech recognition: physical measures of hyperarticulation could be combined with parsing for newness to identify stance expression in natural speech or produce it in synthesized speech.

4pSC2. Use of linguistic knowledge in the recognition of reduced words: Effects of age and high-frequency hearing loss. Esther Janse (Max Planck Inst. of Psycholinguistics, P.O. Box 310, 6500 AH Nijmegen, The Netherlands, Esther.Janse@mpi.nl) and Mirjam Ernestus (Radboud Univ. Nijmegen, Nijmegen, The Netherlands)

Older adults have more language experience than young adults and may therefore rely more strongly on transitional probabilities between words for spoken word recognition. We investigated recognition of acoustically reduced words (such as "promised") that were followed by a word with which it either formed a fixed expression ("promised land") or not ("promised lamp"). Our research questions were (a) whether older adults show a greater following-word context effect on target word recognition than young adults; (b) whether low-pass filtering the stimuli would bring about a stronger reliance on following context in young adults; and (c) whether there are any interactions with the morphological complexity of the target word (e.g., "promised" versus "fat"), because complex and longer words can be more reduced. Data analysis (recognition accuracy and RTs) showed that words were better recognized when part of a fixed expression, and that older adults benefited more in RT than young adults (equal benefit in accuracy), but only for the morphologically complex target words. The young adults with simulated hearing loss, however, benefited less than the other two groups (both in accuracy and RT). These results suggest that signal degradation does not immediately lead to greater reliance on following-word context.

4pSC3. Modulation of phonetic duration by morphological and lexical predictors. Michelle Sims, Benjamin V. Tucker, and Harald Baayen (Dept. of Linguist., Univ. of Alberta, 4-32 Assiniboia Hall, Edmonton, AB T5K 1T2, Canada, mnsims@ualberta.ca)

This study investigates how the duration of the stem vowel of regular and irregular English verbs is modulated by tense (present and past), regularity, lexical frequency, gang size of the vocalic alternation, imageability ratings, and vowel quality. The vocalic durations of 48 monosyllabic irregular verbs and 171 regular verbs were extracted from the Buckeye Corpus of spontaneous speech. A linear mixed effects regression model revealed that vowels of past tense forms tend to have longer durations than vowels of present tense forms, that vowels of words that are less imageable are realized with shorter durations, and that tense vowels are longer than lax vowels. Surprisingly, higher frequency irregular past tense forms were pro-

duced with longer vowels, contradicting Aylett and Turk [(2004); (2006)] and Bell *et al.* [(2003); (2009)]. Further, vowels were shorter for irregular verbs with larger vocalic alternation gangs, contradicting the predictions of Kuperman *et al.* [(2007)] but supporting hypotheses that units with a smaller information load have shorter durations. This pattern of results is interpreted as a consequence of the pressure for regularization during the production of irregular past tense forms.

4pSC4. Variation in stop releases in American English spontaneous speech. Lisa Davidson (Dept. of Linguist., New York Univ., 10 Wash. Pl., New York, NY 10003, lisa.davidson@nyu.edu)

This study extends previous examinations of pre-consonantal stop releases in American English by investigating spontaneous speech from the National Public Radio's StoryCorps project. Previous research has shown that in read speech, English speakers do not release stops in pre-stop and pre-pausal environments between 50 and 60% of the time. In spontaneous speech from more than 10 speakers, stops before obstruents and nasals in word-internal position (e.g., doctor, act), across a word boundary (e.g., pack these), or at the end of a phrase are produced without an acoustic release 62% of the time, while 29% of stops are released and 8% are deleted altogether. The only position in which there are more released stops than unreleased ones is in pre-pausal position (59.2% vs 38.8%). Deletion only occurs for coronals /t/ and /d/, which constitute 75% of the stops in the corpus. Stops are most likely to be unreleased before other stops (86%); before fricatives they are divided between being unreleased (53.7%), released (25.4%), and deleted (20.9%). The status of stop releases in English will be discussed with respect to articulatory coordination, aerodynamics, and stylistic uses by individual speakers. [Research supported by NSF.]

4pSC5. Perceptual evidence for allophonic variation of the palatal fricative /ç/ in spontaneous Berlin German. Stefanie Jannedy, Melanie Weirich, Jana Brunner, and Micaela Mertins (Ctr. for Linguist., ZAS Berlin, Schuetzenstrasse 18, 10117 Berlin, Germany, jannedy@ling.ohio-state.edu)

Auer (2004) postulates that the multi-ethnolect *Kiezdeutsch* (Berlin, Germany) differentiates three realizations of /ɪç/: [ɪç]; [ɪʃ]; [ɪφ]. Acoustic analyzes of 1192 tokens of /ç/ from the ZAS-spontaneous speech database (collected from nine adolescent speakers of the *Kiezdeutsch* multi-ethnolect as spoken in Berlin) showed no reliable differences in *kurtosis*, *skewness*, *cog*, or *peak* between items impressionistically categorized into these three groups. Further, in the spontaneous speech of middle-aged monolingual speakers of the local Berlin dialect [ʃ] variants of /ç/ were also detected, although here this alternation is not attested. The hypotheses are (1) that Berlin-German also has [ʃ] as an allophonic variant of /ç/ and (2) that *Kiezdeutsch* has a 3-way-split of this category. To evaluate the perceptual validity of these assumptions, tests are being conducted asking native Berliners to rate the category membership of excised variant realizations of /ç/ and /ʃ/ from the *Kiezdeutsch*- and Berlin-database on a scale from 1 (/ç/) to 7 (/ʃ/), with 4 indicating no real preference for either one. First results point to a two-way perceptual split for the older Berlin speakers and a more fuzzy category boundary between [ç] and [ʃ] for the *Kiez* data, suggesting a third intermediate perceptual category. [Work supported by a grant from the German Ministry of Science and Education (BMBF).]

4pSC6. Dissecting rate of speech: The effect of phrase final lengthening on articulation rate. Tyler Kendall (Linguist. Dept., Northwestern Univ., 2016 Sheridan Rd., Evanston, IL 60208, t-kendall@northwestern.edu) and Erik R. Thomas (North Carolina State Univ., Raleigh, NC 27695)

Articulation rate in a Mexican American community in Texas is examined in order to determine which social, cognitive, and linguistic factors influence the rate of speech. This research represents one facet of a larger inquiry into variation in temporal patterns in American English. Recent work has demonstrated the strong effect of utterance length on articulation rate—the (pause-exclusive) measure of speech units, typically syllables, produced per unit of time—but utterance-internal factors have been less examined. Here, particular attention is paid to the extent to which articulation rate may be a function of phrase-final lengthening, i.e., is variation in articulation rate a spurious effect of variability in phrase-final lengthening? The analysis of finely time-aligned orthographic transcripts conducted here, coded for intonational and intermediate phrases, indicates that rates of speech for final feet are tightly correlated with rates for non-final feet. Hence variation in articulation rate is not an artifact of the final feet. Non-linguistic factors, such as speaker sex and age, are shown to differentiate articulation rates significantly, though individual variability is also substantial. Statistical models of utterance-level-based and within-utterance articulation rate measures are compared and implications for studies of speech rate are discussed.

4pSC7. Gesture reorganization in Mandarin tone 3 Sandhi. Fang-Ying Hsieh (Dept. of Linguist., Univ. of Southern California, Los Angeles, CA 90089)

This study hypothesizes that the tone 3 Sandhi rule in Mandarin Chinese results from a change in organization of tone gestures. Tone 3 in pre-pausal position has a low falling-rising pitch contour, but when it precedes another tone 3 in the same domain, it becomes a derived tone 2 (rising contour). It is hypothesized that tone 3 is composed of sequential L(ow) and H(igh) tone gestures; since tone 2 is composed of synchronous L and H [Gao (2008)], the derived tone 2 results from shifting the sequential relationship to be synchronous. This hypothesis is tested by a production experiment in which speakers repeated phrases that include a pre-pausal tone 3 followed by another tone 3 in a separate domain. Production rate was gradually increased since it has been shown that increasing speech rates could induce gestural synchrony. Results indicate that the H gesture becomes more synchronous with the L gesture as has been predicted by the hypothesis.

4pSC8. Visual influences on interactive speech alignment. James W. Dias and Lawrence D. Rosenblum (Dept. of Psych., Univ. of California, 900 Univ. Ave., Riverside, CA 92521)

Speech alignment describes the tendency to produce speech that shares characteristics with a perceived speech signal. Visual speech information has also been found to modulate speech alignment [R. M. Miller *et al.*, Attention, Perception, & Performance (in press)]. The present study evaluated whether visual speech information enhances alignment over auditory-alone information in a live context. Pairs of subjects performed an interactive search task for which they were required to utter nine key words multiple times. Half of the subjects performed the task while interacting face-to-face, while the other half performed the task separated by a cloth preventing them from seeing each other without inhibiting the heard speech. Subjects also uttered the key words before and after the interaction. Alignment was evaluated using an AXB matching task in which raters judged whether post-interaction or pre-interaction utterances of a word spoken by a subject were more similar to the post-interaction word spoken by that subject's partner. Raters also judged similarity between partner utterances from early and late in the interaction. Evidence showed that visual information did enhance interactive speech alignment. These findings add to the evidence that amodal, gestural properties of perceived speech can induce alignment.

4pSC9. Prosodic and segmental convergence in spontaneous German conversations. Natalie Lewandowski and Antje Schweitzer (Inst. for Natural Lang. Processing (IMS), Univ. of Stuttgart, 70174 Stuttgart, Germany, natalie.lewandowski@ims.uni-stuttgart.de)

Phonetic convergence is the process of adapting one's speech to that of an interlocutor and is hence seen as a dynamic process causing continuous updates in multiple speech parameters. Current methods of measuring phonetic convergence are often impressionistic or perceptual, or have only been tested in limited scenarios. The aim of this ongoing project is to investigate convergence in German in a natural open scenario and to develop computational objective methods of accounting for it. A variety of phonetic parameters with respect to their relevance for convergence will be examined. On the prosodic level, one goal is to examine phonological features such as the prosodic inventory or distribution of prosodic events as well as the detailed phonetic implementation of prosodic events. The latter can be measured by the PaIntE F0 parametrization method. Convergence effects have been shown to be very subtle; thus large amounts of data are required in order to obtain statistically valid and robust results. Developing and implementing a new automatized procedure allows us to use a large corpus of spontaneous speech without foregoing the chance of capturing fine phonetic detail.

4pSC10. Speech articulator movements recorded from facing talkers using two electromagnetic articulometer systems simultaneously. Mark Tiede (Haskins Labs., 300 George St., New Haven, CT 06511), Rikke Bundgaard-Nielsen, Christian Kroos, Guillaume Gibert, Virginie Attina, Benjawan Kasisopa (Univ. Western Sydney, Bankstown, NSW, Australia), Eric Vatikiotis-Bateson (Univ. of British Columbia, Vancouver, BC, Canada), and Catherine Best (Univ. Western Sydney, Bankstown, NSW, Australia)

Two 3-D electromagnetic articulometer systems, the Carstens AG500 and Northern Digital WAVE, have been used simultaneously without mutual interference to record the speech articulator movements of two talkers facing one another 2 m apart. A series of benchmark tests evaluating the stability of fixed distances between sensors attached to a rotating rigid body was first conducted to determine whether the two systems could operate independently, with results showing no significant effect of dual operation on either system. In the experiment proper, two native speakers of American English participated as subjects. Sensors were glued to three points on the tongue, the upper and lower incisors, lips, and left and right mastoid processes for each subject. Independent audio tracks were recorded using separate directional microphones, which were used to align the kinematic data from both subjects during post-processing. Data collected were of two types: extended spontaneous conversation and repeated incongruent word sequences (e.g., talker one produced "cop top...;" talker two "top cop..."). Both talkers show strong positive correlations between speech rate (in syllables/s) and head movement. The word sequences also show error and rate effects related to mutual entrainment. [Work supported by ARC Human Communication Science Network (RN0460284), MARCS Auditory Laboratories, NIH.]

4pSC11. Analyses of fundamental frequency in infants and preschoolers with hearing loss. Mark VanDam, Mary Pat Moeller (Ctr. for Childhood Deafness, Boys Town Natl. Res. Hosp., 555 N 30th St., Omaha, NE 68131, mark.vandam@boystown.org), and Bruce Tomblin (Dept. Comm. Sci. & Disord., Univ. Iowa, Wendell Johnson Speech & Hearing Ctr., Iowa City, IA 52242)

Technological advances in the last 15 years have resulted in earlier identification of children with mild and moderate hearing loss. Little is known about the impact of early provision of amplification on the development of prosodic speech characteristics such as fundamental frequency (*F0*). This study aims to address that gap. Children enter this study at 12–36 months of age and contribute 1 whole-day audio recording each month for one year. The wearable recorder and associated software (*LENA Foundation*) output (i) a continuous (PCM) audiofile of the whole day and (ii) a time-aligned, XML-coded file at millisecond resolution identifying periods of speech (adult female or male, other children, target child) and other acoustic events (overlapping vocals, noise, silence, etc). In this study, children's *F0* is examined directly and in response to certain talkers or in selected turn-taking relationships (e.g., child-directed speech, father-child turn-taking exchanges). This work includes a detailed methodological description of the use of the LENA system and the customization of that technology relevant to speech and language science. Data from *F0* analyzes are interpreted in descriptive and theoretical terms, complementing an ongoing, multi-center study investigating outcomes of children with mild and moderate hearing loss. [Work supported by NIH/NIDCD Grant Nos. DC009560 and DC009560-01S1.]

4pSC12. The development of temporal structure in mother-child speech. Nicholas A. Smith (Boys Town Natl. Res. Hospital, 555 North 30th St., Omaha, NE 68131, nicholas.smith@boystown.org)

The present study examined age-related changes in the timing of mothers' and children's utterances during free interaction. Using a longitudinal database of recordings of mother-child speech between 4 and 48 months of age, the onset and offset times of maternal and child utterances were coded and analyzed. The goal of the analysis was to use changes in the temporal intervals between adjacent utterances as a measure of mutual responsiveness and coordination. Although considerable individual variation was found across mother-child dyads between 4 and 18 months of age, at 24 months both mothers and children showed tighter temporal coupling, indicated by consistently shorter latencies in their responses to each other's utterances. This change in the temporal pattern likely corresponds to the emergence of turn-taking and reciprocity in dialogue. This study is part of a large project examining the effects of hearing loss on mother-child vocal interaction, and a parallel analysis of dialogue between mothers and their hearing-impaired children is currently underway. [Work supported by NIH Grant R03DC009884.]

4pSC13. Comparison of acoustic characteristics of American English tense and lax vowels in maternal speech to prelingually deaf infants, normal-hearing infants, and adults. Maria V. Kondaurova (Dept. of Otolaryngol.—Head and Neck Surgery, Indiana Univ. School of Medicine, Indianapolis, IN 46202, mkondaur@iupui.edu), Tonya R. Bergeson (Indiana Univ. School of Medicine, Indianapolis, IN 46202), and Laura Dilley (Michigan State Univ., MI 48824)

Recent studies have demonstrated that mothers exaggerate phonetic properties of infant-directed (ID) speech. However, these studies focused on a single acoustic dimension, i.e., frequency, whereas speech sounds are composed of multiple acoustic cues. Moreover, little is known how mothers adjust phonetic properties of speech to children with hearing loss. This study examined the mothers' production of frequency and duration cues to the American English tense/lax vowel contrast in speech to profoundly deaf ($N = 12$) and normal-hearing ($N = 12$) infants and to an adult experimenter. First and second formant frequencies and vowel duration of tense ([i, [u]) and lax ([ɪ], [ʊ]) vowels were measured. Results demonstrated that mothers exaggerated vowel duration in ID relative to adult-directed speech. However, only a trend suggesting an exaggeration of vowel space in ID speech was observed. These findings suggest that although both spectral and duration cues to the tense/lax distinction are modified in a systematic way in ID speech, vowel duration is exaggerated to a greater extent relative to spectral properties. Future analysis will include larger groups of hearing-impaired and normal-hearing infants to identify effects of hearing loss on the relationship between segmental and prosodic characteristics in ID speech.

4pSC14. Naturalistic social communication and speech development in monolingual and bilingual infants. Nairan Ramirez-Esparza, Adrian Garcia-Sierra, and Patricia K. Kuhl (Inst. for Learning and Brain Sci., Univ. of Washington, Box 357988, Seattle, WA 98195)

This investigation explores how everyday social communication between parents and infants relates to speech development. This goal is accomplished by using the digital recorder LENA that monolingual ($N=11$) and Spanish-English bilingual ($N=10$) 14-month-old infants wore for 4 days. Infants' sample files (i.e., 160, 30-s intervals per infant) were coded according to the social communication coding inventory, which includes categories such as "babbling" (e.g., canonical versus variegated babbling), "social interactions" (e.g., baby was with one other person or with a group of people), "how adults are talking" (e.g., "motherese" versus adult talk), and "activities" (e.g., adult is reading and/or teaching). The results showed that babbling relates to relevant social communication categories in both groups. For example, the percentage of time "motherese" is used relates positively to babbling, but adult directed speech relates negatively to babbling. Interestingly, the amount of time monolingual parents spend teaching and reading to their infants relates positively to variegated babbling, but the amount of time bilingual parents have the TV on relates negatively to variegated babbling. These findings shed some light about how natural everyday social communication influences speech development in monolingual and bilingual infants. [Work supported by an NSF grant to the UW LIFE Center.]

4pSC15. Cross-correlation based detection of self-vocalization for the purpose of minimizing the perceived effects of occlusion. Matthew Green and Thomas Burns (6600 Washington Ave. S., Eden Prairie, MN 55344, matthew.green@starkey.com)

Partial or complete occlusion of the ear canal by an assistive listening device results in an unnaturalness in the sound of a person's own voice, known as the occlusion effect. When the occlusion effect is present, if the sound presented to the eardrum is dominated by the amplified output of the assistive listening device, then the frequency response of the device could be adjusted in order to minimize the perceived effects of occlusion. Such an alteration of the frequency response must be applied during self-vocalization and at no other time. A robust and accurate method for detecting self-vocalization will be presented, consisting of a MEMS accelerometer and a cross-correlation based signature detection algorithm. The requirements for signature capture and selection will be discussed, as well as detection performance by gender. Results for detection accuracy, immunity to tasks other than self-vocalization, and binaural agreement will be presented.

THURSDAY AFTERNOON, 18 NOVEMBER 2010 CORAL ISLAND 1/2, 1:00 TO 5:30 P.M.

Session 4pUW

Underwater Acoustics and Signal Processing in Acoustics: Physics-Based Signal Characterization Classification and Processing II

Lisa M. Zurk, Cochair
Electrical and Computer Engineering Dept., Portland State Univ., 1900 S.W. Fourth Ave., Portland, OR 97207

Timothy K. Stanton, Cochair
Dept. of Applied Ocean Physics and Engineering, Woods Hole Oceanographic Inst., Woods Hole, MA 02543-1053

Contributed Papers

1:00

4pUW1. Interpretation of the compressed pulse output for broadband acoustic scattering from inhomogeneous weakly scattering objects. Wu-Jung Lee, Andone C. Lavery, and Timothy K. Stanton (Dept. of Appl. Ocean Phys. and Eng., Woods Hole Oceanograph. Inst., Woods Hole, MA 02543, wjlee@whoi.edu)

Pulse compression signal processing techniques are commonly used to identify dominant scattering features for weakly scattering marine organisms. In cases where the organism can be accurately represented as homogeneous (e.g., eupausiids), the separation of two distinct peaks in the compressed pulse output (CPO) envelope produced by the front and back water-tissue interfaces can be used to estimate the size of the organism at some angles of orientation. In order to investigate the effect of internal inhomogeneities on the scattering, a 3-D distorted-wave Born approximation-based numerical model has been used to model the CPO due to scattering from weakly scattering inhomogeneous objects. It has been found that when the organism is highly inhomogeneous (e.g., sea-water filled cavities in squid), the CPO envelope does not provide an accurate estimate of size. Characteristics of the replica signal can further complicate the issue by introducing spurious peaks with high side lobes or smearing out the CPO peaks from internal inhomogeneities when the main lobe is wide.

1:15

4pUW2. Echo statistics due to a randomly rough, randomly oriented prolate spheroid that is randomly located in a directional active sonar beam. Saurav Bhatia and Timothy K. Stanton (Dept. Appl. Ocean Phys. and Eng., MS #11, Woods Hole Oceanograph. Inst., Woods Hole, MA 02543-1053, saurav30@gmail.com)

The statistical characteristics of sonar echoes from a randomly rough, randomly oriented prolate spheroid are studied through the use of a previously published exact analytical formula by Ehrenberg, which involves a convolution between the probability density functions (PDFs) associated with the sonar beam-pattern and scatterer. This is a direct-path scattering geometry in which only the echo from the scatterer is involved (no reflections from boundaries, such as in a waveguide). The scattering by the rough prolate spheroid is modeled approximately through the use of the Rayleigh PDF for each orientation angle with the mean value at each angle determined by the high-frequency (Kirchhoff) solution to a smooth prolate spheroid. Once the orientation is randomized, the echo from the rough spheroid (without beam-pattern effects) is non-Rayleigh. It had previously been established that random location in the beam pattern causes strongly non-Rayleigh echoes for resolved scatterers whose echoes were otherwise Rayleigh distributed (such as a rough spherical scatterer) before beam-pattern effects. The presence of the randomly rough, randomly oriented prolate spheroid that is randomly located in the beam increases the degree to which the echoes are non-Rayleigh beyond non-Rayleigh effects from the beam pattern.

1:30

4pUW3. Analysis of mixed assemblages of fish using the statistics of echoes from a single beam broadband echosounder. Wu-Jung Lee and Timothy K. Stanton (Dept. of Appl. Ocean Phys. and Eng., Woods Hole Oceanograph. Inst., Woods Hole, MA 02543, wjlee@whoi.edu)

Broadband echoes (30–70 kHz) were collected from fish aggregations with a downward-looking single beam echosounder over Georges Bank near Cape Cod, MA in the fall of 2008. The echoes were collected in a short-range, direct path geometry. Echoes from sequential pings occurring within sampling windows that lie entirely in a given patch were grouped to obtain the distribution of the magnitude of matched filtered echoes. Non-Rayleigh distributed echoes were observed in patches both near the surface and bottom. Two-component mixed probability density functions (pdf's) were formed by accounting for beampattern effects and assuming various proportions of sizes of fish with correspondingly different scattering strengths and are used to fit the measured echo distribution. For each patch, parameters of the best fitting pdf's for a series of sampling windows within the given patch were used to infer the fish size composition of the patch. The best-fit parameters for the surface patches were different from those from near the bottom. Results of the acoustically inferred fish assemblages are compared with the net-tow data and are used to investigate the biological significance of the fish aggregations.

1:45

4pUW4. Detection of a target in shallow water using the feedback phenomenon: A small-scale experimental demonstration. Christian Marandet, Philippe Roux, Patrick La Rizza (LGIT, BP 53 38041, Grenoble Cedex 9, France, christian.marandet@obs.ujf-grenoble.fr), and Barbara Nicolas (GIPSA-LAB, 38402 St. Martin d'Hres Cedex, France, barbara .nicolas@gipsa-lab.inpg.fr)

We experimentally demonstrate the detection of a wavelength-size target in a shallow ultrasonic waveguide between two source-receiver transducers using acoustic feedback. The waveguide represents a 1-km-long, 50-m-deep ocean acoustic channel at the 1/1000 scale. People are familiar with the feedback phenomenon that results in the loud sound heard when a musician plays an electric instrument directly into a speaker. Feedback occurs when a source and a receiver are connected both acoustically through the propagation medium and electrically through an amplifier in such way that the re-

ceived signal is simultaneously and continuously added to the emitted signal. A resonance is then obtained when the emitter and the receiver are in phase. The resonant frequency appears to be highly sensitive to fluctuations of the propagation medium. During the crossing of the target trough the waveguide, fluctuations inferred by the target allow the detection. Experimental results are presented with target of different sizes in the presence of surface fluctuations.

2:00

4pUW5. Tomography of the surface elevation in shallow water between two source-receiver ultrasonic arrays: A small scale experiment demonstration. Christian Marandet (LGIT, BP 53, 1381 rue de la piscine, 38400 St. Martin d'Hres, France, christian.marandet@obs.ujf-grenoble.fr), Philippe Roux, and Barbara Nicolas (Gipsa-Lab, Cedex 9, 38042 Grenoble, France)

Based on single-scattering effects, the diffraction-based sensitivity kernels which make the link between the acoustic perturbations and the medium fluctuations have been extensively studied. We propose a similar study for perturbations at the air-water interface in an ultrasonic waveguide that scales down with a 1-km-long, 50-m-deep ocean acoustic channel in the kilohertz regime. Using array processing between two source-receiver arrays, the sensitivity kernel for both time and amplitude has been experimentally inferred for each acoustic echo reverberated in the waveguide to a small surface elevation. We demonstrate a good agreement between experimental and theoretical sensitivity kernels. Finally, using theoretical sensivity kernels for both amplitude and time ultrasonic fluctuations, we give a real-time estimation of the surface elevation at any point between the source and receiver arrays.

2:15

4pUW6. Enhancing resonant features in the aspect dependence of target spectra by reversible imaging filtration techniques. Timothy M. Marston, Philip L. Marston (Dept. Phys. and Astronomy, Washington State Univ., Pullman, WA 99164-2814), and Kevin L. Williams (Univ. of Washington, Seattle, WA 98105)

The aspect dependence of the spectral response of simple objects is sometimes examined in scattering research. The use of reversible imaging algorithms for the approximate de-coupling of the impulse response and associated spectral-responses of two adjacently located targets with overlapping signals was previously demonstrated [T. M. Marston, P. L. Marston, and K. L. Williams, J. Acoust. Soc. Am. **127**, 1749 (2010)]. The current research concerns how resonant features of targets can be enhanced by using related imaging and filtering algorithms. When resonant targets are imaged using synthetic aperture techniques, the resonant portion is often distinct enough from specular or edge-diffraction portions so that it is possible to design spatial filters that isolate the resonant from the non-resonant portions of the signal in the image domain. The filtered image is then returned to the original domain, and frequency response is plotted with the non-resonant portions of the data filtered out. The resulting plots show enhanced resonant features. Demonstrations using line-scan SAS and circular SAS data will be shown. [Work supported by ONR.]

2:30

4pUW7. Feature timing and identification for different solid cylinder exposures revealed using reversible image filtering. Grant C. Eastland, Timothy M. Marston, and Philip L. Marston (Phys. and Astronomy Dept., Washington State Univ., Pullman, WA 99164-2814)

Understanding scattering features of proud and partially exposed cylinders is relevant to the scattering by a variety of simple targets. In the present investigation, partial exposure was studied by lowering a solid aluminum cylinder through a flat free surface into a tank of water while monitoring the evolution of the scattering as a function of the amount of exposure. Interactions with the free surface simulate aspects of interactions with flat sediment. Unlike related prior work, however, the present investigation allows for the recording of bistatic scattering and reversible image filtering based on a form of synthetic aperture sonar (SAS). The source and receiver grazing angles are held fixed while the amount of exposure h of a right circular aluminum cylinder is varied. Short pulses are used to distinguish between different types of scattering contributions. The slope of the feature timing as a function of the exposure h, expressed by the derivative dt/dh where t is the measured time of the feature, depends on the feature type as well as the source and receiver grazing angles. Free surface interactions for features revealed by the slope dt/dh are consistent with feature identification using reversible SAS filtering. [Work supported by ONR.]

2:45

4pUW8. Measured acoustic scattering from a water-filled aluminum pipe in contact with various interfaces. Aubrey L. Espana, Kevin L. Williams, Steven G. Kargl (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Box 355640, Seattle, WA 98105-6698, aespana@apl.washington.edu), Timothy M. Marston, and Philip L. Marston (Washington State Univ., Pullman, WA 99164-2814)

Detection and classification of objects on or imbedded in the ocean sediment are difficult problems due to the complexity of the sediment. The measured scattering will include interactions of the incident sound with the sediment interface as well as the possibility for multiple interactions between the target and sediment. Recent experiments conducted in the test pond at the Naval Surface Warfare Center, Panama City Division, investigated the impact the environment has on the measured acoustic response from various targets. A particular subset of these experiments is presented, focusing on the acoustic scattering from a water-filled aluminum pipe having length-to-diameter ratio equal to 2. Initial measurements examined the scattering from the pipe in a proud configuration on the flattened sand sediment. Subsequent measurements placed the pipe in contact with interfaces with known reflection coefficients, including suspending the pipe just below the air/water interface and placing it in a proud configuration on a flat acrylic panel on the sand sediment. Acoustic templates depicting the absolute target strength as a function of azimuthal angle over the frequency range 1–30 kHz are compared for the three different configurations. [Work supported by the Office of Naval Research.]

3:00—3:15 Break

Invited Paper

3:15

4pUW9. The interplay of acoustic signals with boundary and volume scattering phenomena: An overview. Roger C. Gauss (Naval Res. Lab., Code 7144, Washington, DC 20375-5350)

Active sonar systems use acoustic sources and receivers coupled with signal processing to detect, classify, localize, and track undersea targets. The limiting influence of scattering phenomena, manifesting as reverberation and clutter, on sonar performance has long been recognized, with acoustic echoes from the waveguide boundaries and biologics capable of both masking desired signals and creating false targets/tracks. This talk provides a broad overview of reverberation and clutter phenomena in both deep and shallow water in the context of their impact on sonar performance for frequencies of 50 Hz to 5 kHz. The interplay of system (geometry, transmitter /receiver design, frequency, waveform, and signal processing) and environmental (oceanography, geology, ice, and biology) factors will be discussed, with a series of data and model examples demonstrating the importance of using high-fidelity, physics-based approaches to both characterize the acoustic field and guide the signal processing. While the environment presents many challenges to active sonar systems because of its spatiotemporal variability and strong coupling between scattering and propagation structures, it should also be viewed as both potentially manageable and presenting opportunities. If the mechanisms are understood, they can be dealt with, and even exploited. [Work supported by ONR.]

Contributed Paper

3:35

4pUW10. Tracking sperm whales with a towed acoustic vector sensor using physics-based noise analysis. Aaron M. Thode, Jeff Skinner, Pam Scott, Jeremy Roswell (Marine Physical Lab., Scripps Inst. of Oceanogr., San Diego, CA 92093-0205, athode@ucsd.edu), Janice Straley (Univ. of Alaska Southeast, Sitka, AK 99835), and Kendall Folkert (P.O. Box 6497, Sitka, AK 99835)

Passive acoustic towed linear arrays are increasingly used to detect marine mammal sounds during mobile anthropogenic activities. However, these arrays cannot resolve between signals arriving from the port or starboard without vessel course changes, and their performance is degraded by vessel self-noise and non-acoustic mechanical vibration. In principle, acoustic vector sensors can resolve these directional ambiguities, as well as flag the presence of non-acoustic contamination, provided that the vibration-sensitive sensors can be successfully integrated into compact tow modules. Here a vector sensor module attached to the end of an 800-m towed array is used to detect and localize 1813 sperm whale clicks off the coast of Sitka, AK. Three physics-based methods are used to identify frequency regimes relatively free of non-acoustic noise contamination, and then the active intensity of the signal was computed between 4 and 10 kHz along three orthogonal directions, providing unambiguous bearing estimates of two sperm whales over time. These bearing estimates are consistent with those obtained via conventional methods, but the standard deviations of the vector sensor bearing estimates are twice those of the conventionally derived bearings. The resolved ambiguities of the bearings deduced from vessel course changes match the vector sensor predictions.

Invited Paper

3:50

4pUW11. High fidelity noise and reverberation waveforms for sonar stimulators. Chris Harrison (NATO Undersea Res. Ctr., Viale San Bartolomeo 400, 19126 La Spezia, Italy)

A stimulator produces multiple hydrophone signal, noise, and reverberation waveforms sophisticated enough to produce a realistic behavior in the output displays of a sonar system run on the bench. These waveforms need to be able to deceive the sonar's given signal processing, detection and tracking algorithms, etc. This talk focuses on the difficult problem of reproducing the directionality, spatial and temporal coherence, statistics, and spectra in ambient noise and reverberation, and instilling this into the channels of a hydrophone array. Frequency domain approaches work well for noise where directionality hardly changes in time. Time domain approaches work better for reverberation where the directionality changes as the arrivals come from longer and longer ranges. Since the aim is real-time operation, the level of fidelity is an important consideration; one needs correct mechanisms for the sonar system under consideration, although they can often be crudely, rather than exactly, modeled. For instance, simulating noise for a single hydrophone system would be much simpler than for a vertical array system intended for beam-cross-correlation and passive fathometry; nevertheless, both simulation approaches are possible. Examples of noise and reverberation will be shown.

Contributed Papers

4:10

4pUW12. Time domain calculation of propagation interference fringes (striations). Chris Harrison (NATO Undersea Res. Ctr., Viale San Bartolomeo 400, 19126 La Spezia, Italy)

The waveguide invariant, which manifests itself as interference fringes or striations in a plot of frequency versus source-receiver separation, is usually thought of as a modal phenomenon. This paper shows that this phenomenon can be explained very simply through the variation of the eigenray arrival times with range; in short, the variation of the multipath impulse response. It is possible to calculate the value of beta [= (df/dr) /(f/r)] for a number of sound speed profiles analytically and to see what beta depends on, why it switches from one value to another, how it depends on source-receiver depth, and so on. The findings will be confirmed by calculating striation patterns numerically, starting from eigenray travel times in arbitrary stratified environments. Most importantly, the approach throws some light on what can be deduced from beta alone and the likelihood and utility of striations in reverberation.

4:25

4pUW13. Active invariance structure in deep water towed array data. Lisa M. Zurk (Portland State Univ., Portland, OR, zurkl@pdx.edu), Zoi-Heleni Michalopoulou (New Jersey Inst. of Tech., Univ. Heights, Newark, NJ), and Dan Rouseff (Univ. of Wash., Seattle, WA)

The concept of a waveguide invariant has been used extensively to explain and exploit the time-frequency structure present in passive lofargrams. Recently this concept has been extended to active sonar data and has been demonstrated experimentally for both target and bottom scattering returns obtained in shallow water channels. In this talk, characteristics of the active invariance structure observed with towed horizontal arrays operating in deep water channels are presented and discussed. A towed array offers an ideal array aperture to observe the active invariance structure because a single pulse received across the array can be processed to recover the invariance striations. This processing offers higher resolution than what would be obtained with multiple returns from a moving target when sampled across a single receiver. Furthermore, for larger apertures operating at higher frequencies, the ability to observe the invariance intensity structure may be affected by details of the beamforming. The framework for towed array invariance processing is presented theoretically and illustrated with data obtained from a towed array operating in a deep water environment. As expected, the values for the deep water invariant differ from shallow water in both magnitude and sign.

4:40

4pUW14. A summary of filtering approaches in ocean acoustics. Caglar Yardim (Marine Physical Lab., Univ. California, San Diego, 9500 Gilman Dr., La Jolla, CA 91915, cyardim@ucsd.edu), Zoi-Heleni Michalopoulou (New Jersey Inst. of Tech., Newark, NJ 07102), and Peter Gerstoft (Univ. California, San Diego, La Jolla, CA 91915)

Sequential filtering provides an optimal framework for estimating and updating the unknown parameters of a system as data become available. Filtering is a powerful estimation tool, employing prediction from previous estimates and updates stemming from physical and statistical models that relate acoustic measurements to the unknown parameters. The foundations of sequential Bayesian filtering with emphasis on practical issues are presented covering both Kalman and particle filter approaches. Extended, unscented, and ensemble Kalman filters are compared to particle filtering approaches such as sequential importance resampling and advanced variants. Complex problems in ocean acoustics, where the state vector order is uncertain or the most suitable physical model for the problem at hand is not known *a priori*, are also addressed. Examples of model order selection and multiple model particle filters in underwater acoustic applications are given. Ocean acoustic examples are presented in target tracking, wave estimation, geoacoustic inversion, and frequency tracking.

4:55

4pUW15. Model-based estimation and uncertainty characterization of modal arrival times and amplitudes in ocean acoustics. Zoi-Heleni Michalopoulou (Dept. of Math. Sci., New Jersey Inst. of Technol., Newark, NJ 07102, michalop@njit.edu)

A Bayesian approach is proposed for modal arrival time and amplitude estimation from short-time Fourier transforms of broadband acoustic signals in the ocean. The goal is to obtain accurate estimates of arrival times of propagating modes and corresponding amplitudes using a physical model to describe short-time Fourier transform slices and a signal processing method to estimate parameters that are integrated in the physical modeling. The method provides uncertainty information on both modal arrival time and amplitude estimates, which is desirable in ocean acoustic inversion that utilizes a combination of modal information and sound propagation models. [Work supported by ONR.]

5:10—5:30 Panel Discussion