



Transducer models in the ultrasound simulation program FIELD II and their accuracy

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Session 2aAO

Acoustical Oceanography and Underwater Acoustics: Environmental Effects on Acoustic Propagation

Roger M. Oba, Cochair

Naval Research Lab., Code 7120, Washington, DC 20375

Ying-Tsong Lin, Cochair

*Woods Hole Oceanographic Inst., 210 Bigelow Bldg., Woods Hole, MA 02543***Contributed Papers**

9:15

2aAO1. Acoustic ducting and refraction by sea bottom relief in shallow water. James F. Lynch, Alexey A. Shmelev, Ying-Tsong Lin, and Arthur E. Newhall (Dept. of Appl. Ocean Phys. and Eng., Woods Hole Oceanograph. Inst., 98 Water St., Woods Hole, MA 02543, jlynch@whoi.edu)

Observations show that shallow water bottom relief often has a band-limited directional spectrum produced by various oceanographic and geological processes. This directional bottom feature is shown to have a noticeable effect on three-dimensional low-frequency acoustic propagation. An analytical study with an idealized model of straight sea bottom ripples has shown that acoustic energy can be partially ducted between neighboring ripples, and this ducting will affect acoustic propagation in shallow water. In our work, we also study ducting and refracting due to idealized curved sea bottom ripples. Previous research has shown that non-linear internal waves can also create acoustical ducts. Comparative analysis of these two different ducts is performed using our idealized model. The combined effects of internal waves and bathymetry are studied for various relative directions of internal wave front and bottom ripples. A numerical simulation of three-dimensional sound propagation across realistic bathymetry and internal wave fluctuations is performed. In conclusion, both water column fluctuations and bathymetry variability need to be taken into account when studying three-dimensional acoustic propagation in shallow water.

9:30

2aAO2. Three-dimensional sound propagation over submarine canyons. Ying-Tsong Lin, Timothy F. Duda (Dept. Appl. Ocean Phys. & Eng., Woods Hole Oceanograph. Inst., Woods Hole, MA 02543, ytlin@whoi.edu), Jon M. Collis (Colorado School of Mines, Golden, CO 80401), James F. Lynch, and Arthur E. Newhall (Woods Hole Oceanograph. Inst., Woods Hole, MA 02543)

Submarine canyons are common features of continental shelf and slope regions, e.g., Hudson Canyon in the Mid-Atlantic Bight. In this paper, the impact of submarine canyons on low-frequency sound propagation is studied using a three dimensional (3-D) parabolic approximation numerical program, which is implemented in a Cartesian coordinate system and utilizes the split-step Fourier technique and a 3-D variant of the Thomson and Chapman wide-angle approximation. This program will be first benchmarked with a classic wedge problem, and then used to study an idealized canyon environment to understand distinct 3-D sound propagation effects. The idealized environment has a Gaussian shaped canyon incising a slope. Horizontal focusing of sound in the canyon and energy flow into the canyon from an off-axis sound source are observed. A realistic model using the Hudson Canyon bathymetry shows even more complex sound propagation situations. Propagation conditions over different seabed types are also compared, and the 3-D field sensitivity to bottom properties is investigated. [Work supported by the Office of Naval Research.]

9:45

2aAO3. Three dimensional parabolic equation modeling of an internal wave event during Shallow Water 2006. Georges A. Dossot, James H. Miller, Gopu R. Potty (Dept. of Ocean Eng., Univ. of Rhode Island, Narragansett Bay Campus, Narragansett, RI 02882), James F. Lynch, Ying-Tsong Lin (Woods Hole Oceanograph. Inst., Woods Hole, MA 02543), Mohsen Badiy (Univ. of Delaware, Newark, DE 19716), and Kevin, B. Smith (Naval Postgrad. School, Monterey, CA 93943)

During the Shallow Water 2006 (SW06) experiment, a J-15 acoustic source deployed from the Research Vessel Sharp transmitted broadband (100–500 Hz) chirp signals 15 km away from a vertical line array. The array was intentionally positioned near the shelf-break front and in an area where internal waves are known to occur. During the same time an internal wave, “Event 44,” passed through the sound field such that the internal wave front was near parallel to the acoustic transmission path. Measured data show substantial intensity fluctuations that vary over time and space due to complex multimode and multipath (both two and three dimensional) interference patterns. Of specific interest are fluctuations of measured intensity preceding the internal wave’s arrival. Additionally, depth variability of the measured acoustic intensities can be attributed to a warm water intrusion coinciding with the internal wave event. This presentation shows recent modeling results using the experimental geometry, acoustic signal parameters, and a simulated oceanographic environment based on environmental moorings and ship-born sensors. A new version of the three-dimensional Monterey–Miami parabolic equation code, which incorporates a user-defined sound speed field, is used. [Work sponsored by the Office of Naval Research.]

10:00

2aAO4. Horizontal focusing/defocusing due to shallow-water internal waves. Jing Luo, Mohsen Badiy (Univ. of Delaware, Robinson Hall 112B, 261 S. College Ave., Newark, DE 19716, luojing@udel.edu), and Ying-Tsong Lin (Woods Hole Oceanograph. Inst., Woods Hole, MA 02543)

During the New Jersey Shallow Water 2006 (SW06) experiment, an acoustic source was towed by the Research Vessel Sharp and followed the front of an internal wave packet. The source was transmitting broadband acoustic signals (50–450 Hz) in different angles with respect to the internal wave front. The receptions of transmitted signal on a vertical hydrophone line array are analyzed to study the horizontal focusing/defocusing that occurred when the internal wave front and acoustic track aligned closely. Based on ship-board radar images and temperature data collected on the environmental moorings at various locations along the acoustic track, a detailed three-dimensional (3D) environment is reconstructed for a 3-D parabolic approximation model to study the unique propagation scenario. Construction of index of normal mode refraction for these data provides a clear picture of acoustic energy focusing for this event. Data and model comparison are in good agreement. [Work supported by ONR321 OA]

10:15—10:30 Break

10:30

2aAO5. Spatial and temporal sound fluctuations in shallow water in presence of internal soliton in transition areas. Boris Katsnelson and Andrey Malikhin (Voronezh Univ., 1 Universitetskaya sq, Voronezh 394006, Russia)

Behavior of the sound field is considered in a 1–2-h time interval in presence of train of internal soliton (IS) crossing an acoustic track. During this period moving ISs pass through several stages: approaching an acoustic track, consequent covering source (receiver) only, both source and receiver, receiver (source) only, and receding from an acoustic track. Correspondingly there are different regimes of interaction of the sound field with IS: horizontal reflection, capture of signals in horizontal waveguide, adiabatic variations, and transition areas between these regimes. So, rather complex spatial and temporal structure of the sound field takes place, including frequency and modal dependence of its parameters. Theoretical analysis of variations in the sound field on the basis of techniques of vertical modes and horizontal rays (PE in horizontal plane) is carried out, and estimation of feasibility of an experimental setup is presented. [Work was supported by CRDF.]

10:45

2aAO6. Calculating the waveguide invariant for non-ideal waveguides. Kevin L. Cockrell and Henrik Schmidt (Dept. of Mech. Eng., MIT, Cambridge, MA 02139)

The waveguide invariant describes striations in a range versus frequency plot of a waveguide's Green's function. Analytic expressions for the waveguide invariant only exist for a few select waveguides, but experiments and simulations have shown that the waveguide invariant is approximately equal to unity for almost all realistic shallow-water waveguides. A quasi-analytic method will be presented for estimating the value of the waveguide invariant in waveguides with arbitrary sound speed profiles, including the effects of a bottom fluid halfspace. The method is approximate but allows for an intuitive understanding of why the value of the waveguide invariant does not strongly depend on the details of the sound speed profile.

11:00

2aAO7. Arrival structure variability of single-bounce paths for high-frequency transmissions during the experiment KAM08 (Kauai acoustics communications multidisciplinary research initiative 2008). Joseph M. Senne, Aijun Song (Univ. of Delaware, 210 Robinson Hall, Newark, DE 19711, sennejm@udel.edu), and Kevin B. Smith (Graduate School of Eng. and Appl. Sci., Monterey, CA 93943)

During the summer of 2008 an experiment was conducted that included both chirp and M-sequence transmissions at 16-kHz center frequency. Source and receiver arrays were located west of Kauai Island HI, along an isobath of about 100 m. Moored thermistor strings and a wave-rider buoy provided detailed oceanographic data, while shipboard measurements recorded wind variations. Micro-multi-paths were observed from single surface bounces, and their variability has been examined for a variety of surface wave conditions. A surface wave model has been integrated into a parabolic equation model (MMPE) to approximate variations in the micro-multi-path structure over geotime. Model results are used to examine the correlation between environmental variability and observed single-bounce signal fluctuations. Comparisons are made with a variety of surface wave conditions, including both calm and rough seas. [Work supported by ONR 3210A.]

11:15

2aAO8. Impact of surface gravity waves on high-frequency acoustic propagation in shallow water. Entin A. Karjadi, Mohsen Badiey (College of Earth, Ocean, and Environment, Univ. of Delaware, 210 Robinson Hall, Newark, DE 19716, karjadi@udel.edu), and James T. Kirby, Jr. (Univ. of Delaware, Newark, DE 19716)

Sea surface roughness is one of several factors that significantly influences high-frequency (1–50-kHz) acoustic wave propagation in shallow water. The evolving sea surface introduces several variability effects including Doppler shift. Data analyses from high-frequency acoustic experiments show high-correlation between time, angle, and intensity fluctuations of received signals and varying sea surface conditions. In order to assess detailed acoustic signal interactions with the sea surface, a realistic wave model is developed and combined with an acoustic ray-based model. Model validity is evaluated by comparing the results with data from multiple experiments. [Work supported by ONR 3210A.]

11:30

2aAO9. Acoustic observations of subsurface instability. Justin M. Eickmeier, Mohsen Badiey (College of Earth, Ocean and Environment, Univ. of Delaware, Newark, DE 19716, jeickmei@udel.edu), and Tokuo Yamamoto (Rosenstiel School of Marine and Atmospheric Sci., Univ. of Miami, Miami, FL 33149)

A high-frequency (0.6–18-kHz), shallow water acoustic experiment (HFA2000) was conducted in Delaware Bay (15-m depth) during December 2000. Reciprocal transmissions of chirp signals (0.345-s duration) were radiated between three bottom mounted source-receiver tripod stations separated by 70–353 m. Environmental data were collected at a nearby oceanographic platform; simultaneously, a shipboard ADCP and CTD were deployed. Analysis of direct path station-to-station arrival times (between December 18th 00:00 and December 19th 10:00, during which 126 acoustic transmissions consisting of 29 chirps were radiated) revealed significant deviation from arrival time patterns established during previous tidal cycles. Examination of the corresponding signal intensity reflected this deviation. Independent ADCP data displayed current profile distortion during the period along the direction of the dominant flow channel. The mean slope of a wave number vs geo-time spectrum was calculated from each geo-time's respective chirp series. Slope changes correlate to variations in the total signal intensity's constituents, $I_{\text{tot}}(t) = \langle |E(t)|^2 \rangle$, particularly the incoherent or scattered intensity, $I_{\text{incoh}}(t) = I_{\text{total}}(t) - I_{\text{coh}}(t)$. Through comparison with a Kolmogorov power spectrum and calculation of the corresponding Richardson number, a profile of environmentally induced subsurface instability has been developed.

11:45

2aAO10. The three-dimensional acoustic field of primary arrivals from a seismic airgun array. Arslan M. Tashmukhambetov, George E. Ioup, Juliette W. Ioup (Dept. of Phys., Univ. of New Orleans, New Orleans, LA 70148, atashmuk@uno.edu), Natalia A. Sidorovskaia, Anca Niculescu (Univ. of Louisiana at Lafayette, Lafayette, LA), Joal J. Newcomb (Naval Oceanograph. Office, Stennis Space Ctr., MS), James M. Stephens, Grayson H. Rayborn (Univ. of Southern Mississippi, Hattiesburg, MS), and Phil Summerfield (ExxonMobil Corp., Houston, TX)

The Source Characterization Study 2007 (SCS07) measured the three-dimensional (3-D) acoustic field of a seismic airgun array. The Littoral Acoustic Demonstration Center (LADC) performed the experiment, collecting acoustic and related data on three moored hydrophone arrays and one ship-deployed hydrophone array which together spanned the full water column. Sensitive and desensitized phones were deployed at each position to extend the dynamic range. An ultra short baseline localization system was deployed with the EARS moorings to provide array shape. With postanalysis this results in time-dependent positions for each of the acoustic sensors. Every channel is calibrated. A seismic source vessel shot a series of lines designed to give detailed angle and range information concerning the field of the primary arrival. Peak pressures, sound exposure levels, total shot energy spectra, and one-third octave band analyses give important insights into details of the acoustic field. Images of these quantities are generated to show dependence on emission and azimuthal angles and range. 3-D visualizations and two-dimensional cuts through the data are shown. [Research supported by the Joint Industry Programme through the International Association of Oil and Gas Producers.]

Session 2aBB**Biomedical Ultrasound/Bioresponse to Vibration, Animal Bioacoustics, Speech Communication, and Psychological and Physiological Acoustics: Blast-Induced Traumatic Brain Injury: Mechanisms, Assessment, Therapy, and Mitigation**

Steven G. Kargl, Cochair

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

William C. Moss, Cochair

Lawrence Livermore National Lab., 7000 East Ave., Livermore, CA 94550

Thomas J. Matula, Cochair

*Univ. of Washington, Applied Physics Lab., 1013 NE 40th St., Seattle, WA 98105-6698***Chair's Introduction—8:05*****Invited Papers*****8:10****2aBB1. Defense Advanced Research Projects Agency (DARPA) Blast Program—Preventing violent neurologic trauma.** Geoffrey Ling (Defense Advanced Res. Projects Agency, 3701 N. Fairfax Ave., Arlington, VA 22203)

DARPA is directing a comprehensive program to determine the causes of explosive neurological trauma. The current status of the program will be discussed.

8:30**2aBB2. Basics of blast physics.** David Ritzel (19 Laird Ave. North, Amherstburg, ON N9V 2T5, Canada)

Blast science concerns the processes by which the energy of an explosion source becomes propagated into its surrounding environment, then interacts, loads, and damages materials, structures, and systems. Although explosive events have been chronicled for over 2000 yrs, and good empirical insights regarding blast phenomenology were established prior to WW-II, the advent of the nuclear bomb drove the first rigorous and concerted efforts to understand the detailed physics of blast propagation. Blast science merges with classical acoustic sciences in its far-field limits including phenomena such as atmospheric focusing. However, in the interaction of blast with composite structures including the human body, a wide range of transmitted, coupled, or overdriven mechanical waves of many types can be generated. There has been a significant resurgence of R&D in protective technologies against modern blast threats such as terrorist bombings and IED attacks against our armed services. The current paper presents a review of the basic of blast physics from the near to far fields, principles of blast simulation in the laboratory, as well as recent progress in the understanding of blast induced traumatic brain injury.

8:50**2aBB3. Blast associated traumatic brain injury.** David F. Moore and Michael S. Jaffee (DVBIC-HQ, WRAMC, 6900 Georgia Ave., NW, Washington, DC 20309, david.f.moore@amedd.army.mil)

Blast tissue injury is familiar problem in military medicine for gas filled organ such as the lung and gut. Solid tissue organs such as the brain have more recently come into prominence in relation to blast through the current conflicts in Iraq and Afghanistan with blast injury regarded as the "signature injury" of these wars. Enhancements in personal protective equipment allowing survivability of abdominal and thoracic injury may have exposed a previously obscured brain vulnerability. Free field explosive detonation results in rapid conversion of chemical energy into the shock wave and pressure field, the kinetic energy associated with fragments and shrapnel, thermal energy, chemical products of detonation, and electromagnetic radiation. The variation and coupling of these physical fields result in a uniquely complex problem in understanding blast biological effects especially in such a functionally intricate organ as the brain. The clinical effects of concussion following blast exposure are still undergoing evaluation most notably in the military context. Significant effort is, however, underway to determine the relative contributions of shock wave stress, the effects of cavitation, and the effects of the electromagnetic field induced by the piezo-electric effect of the skull experiencing blast associated stress in brain injury and recovery.

9:10**2aBB4. Blast induced electromagnetic pulses in the brain from bone piezoelectricity.** Steven G. Johnson (Dept. of Mathematics, MIT, 77 Massachusetts Ave., Cambridge, MA 02139, stevenj@math.mit.edu), K. Y. Karen Lee, Michelle K. Nyein (MIT, Cambridge, MA 02139), David F. Moore (Walter Reed Army Medical Ctr., Washington, DC), John D. Joannopoulos, Simona Socrate, and Raul Radovitsky (MIT, Cambridge, MA 02139)

The mechanisms that might lead to in-brain electromagnetic pulses from an IED-scale explosive are considered, along with whether the resulting fields might have timescales and magnitudes relevant to neurological processes. In particular, due to known piezoelectric properties of bone, it is possible for a shock wave incident on the skull to directly induce large electric fields within the brain. Using

experimental data on the piezoelectric properties of bone combined with stresses from full-head-model blast simulations, the resulting in-brain electric fields shown to have timescales and magnitudes that exceed IEEE safety standards are comparable to procedures such as transcranial magnetic stimulation that are known to have neurological effects. Not only are such electromagnetic fields at least potentially relevant to the understanding of blast-induced traumatic brain injury but they may also lead to diagnostic tools in the form of blast dosimeters that measure blast-induced head stresses via the piezoelectric fields produced just outside the skull. [This work was supported by financial aid from the Joint Improvised Explosive Device Defeat Organization (JIEDDO) through the Army Research Office.]

9:30

2aBB5. Effects of low blast levels on the central nervous system. Annette Säljö, Berndt Svensson, Maria Mayorga, Hayde Bolouri, and Anders Hamberger (Dept. of Medical Chemistry and Cell Biology, Inst. of Biomedicine, Sahlgren Acad., Univ. of Gothenburg, SE 405 30 Gothenburg, Sweden, annette.saljo@gu.se)

Anaesthetized swine in crew positions were exposed to weapons in air or to explosives underwater. Blast parameters were correlated with those in the brain. The peak pressure in the brain (P_{max} brain/air) was 0.7 for a bazooka (45 kPa), 0.5 for a howitzer (10 kPa), and 0.4 for a rifle (23 kPa). The brain/water P_{max} for the detonation pulse of under water explosives was only 0.1, but 0.3–0.4 for the secondary pulses. The results indicate that low-frequency spectra penetrate easier into the brain. Histological examination revealed small hemorrhages in rear regions of the brain. In rats, we investigated the effect of shock tube blasts. After exposure to 10 or 30 kPa, cognitive performance (Morris Water Maze) decreased by 50%. The intracranial pressure (ICP) increased in a dose dependent fashion to reach peak levels 6 h after exposure to 10 kPa and 10 h after exposure to 30 or 60 kPa. An initial ICP elevation took place 30 min after exposure to 60 kPa, and 2 and 6 h after exposure to 30 and 10 kPa, respectively. A prophylaxis, consisting of a 2 week intake of hydrothermally fermented cereals, reduced significantly the blast effect both on ICP and cognitive performance. [The authors thanks Svante Hjer, Samba Sensors AB. The study was supported by the Swedish Armed Forces and FMV.]

Contributed Paper

9:50

2aBB6. Effects of a simulated blast pulse train on a simple neural model. Radia Wahab (radiaraian@yahoo.com), Yunbo Liu, Victor Krauthamer (U.S. Food and Drug Administration, Silver Spring, MD), Joseph McCabe, Chantal Moratz, Ryan Egan (Uniformed Services Univ. of the Health Sci., Bethesda, MD), Vesna Zderic (George Washington Univ., Washington, DC), and Matthew Myers (U.S. Food and Drug Administration, Silver Spring, MD)

In the treatment of traumatic brain injury (TBI) due to blast-wave exposure, an understanding of the mechanisms of injury is critical for proper intervention. One mechanism of particular importance is the interaction of blast waves with neurons. In order to investigate this interaction in isolation from other TBI mechanisms, blast waves were simulated on a local

(approximately 1-mm) scale using high-intensity focused ultrasound (HIFU). The acoustic impulse of the HIFU blast wave was selected to approximate the impulse of an actual blast. As a first step in a chain of increasingly complex neural models, the giant axon in an earthworm was used in this study. Intact earthworms were exposed to simulated blast waves from an HIFU transducer, and axonal action potentials were stimulated and recorded. The action potential amplitude decreased continuously with increasing impulse, with the signal typically vanishing at about 8000 Pa s. The conduction velocity showed more of a threshold effect, decreasing minimally from control values until an impulse of about 4000 Pa s was reached, at which point the decline was more rapid. Results indicate that exposing isolated neurons to HIFU-simulated blasts is a promising approach for studying TBI.

10:05—10:30 Break

Invited Papers

10:30

2aBB7. Toward the non-invasive determination of cerebral perfusion pressure. Pierre D. Mourad (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, pierre@apl.washington.edu)

Brains subjected to blast from an explosive or other sources of trauma often develop intracranial hemorrhage or edema or lose their ability to autoregulate the blood flowing into the brain (cerebral autoregulation). Such damage can, in turn, lead to increases in intracranial pressure (ICP) and decreases in cerebral perfusion pressure ($CPP = \text{arterial blood pressure} - ICP$) that, in turn, can lead to ischemia and/or herniation, and brain death. This is not good for the patient. To assay ICP, of intrinsic value, and CPP—two variables of critical interest to the medical staff who treat and manage patients with head injury—requires a (invasive) neurosurgical procedure. Here I describe research performed over the last several years whose target is the prediction of ICP and CPP using arterial blood pressure data as well as data derived from transcranial Doppler. In particular, I will review the process by which we collected sufficient human-derived data in order to make a decent pass at predicting ICP for patients with closed traumatic brain injury (TBI) and a great effort at predicting CPP for these and a complementary group of patients.

10:50

2aBB8. Blast-induced traumatic brain injury research at Lawrence Livermore National Laboratory. William C. Moss and Michael J. King (Lawrence Livermore Natl. Lab., 7000 East Ave., Livermore, CA 94551, wmoss@llnl.gov)

Our blast-induced TBI research has computational and experimental components. Our numerical hydro-structural simulations show that non-lethal blasts can induce sufficient flexure of the skull to generate potentially damaging loads in the brain, even if no impact occurs. The possibility that this mechanism may contribute to TBI has implications for the diagnosis of soldiers and the design of protective equipment such as helmets. Our experimental work involves designing and testing blast dosimeters, which are needed to quantify the blast environment around the soldier, independent of the mechanism(s) causing TBI. One system that uses MEMS sensors incorporated into the helmet and suspension would record peak pressure, positive-phase duration, blast direction, loads directly on the

skull, and accelerations. Another system uses less sophisticated, inexpensive, small, lightweight, disposable, unpowered sensors that act as “yes-no” gauges that indicate the blast magnitude by visual inspection of the gauge. This system is a trade-off between quantity and quality of data, which may be viable, based on current DoD needs. [This work performed under the auspices of the U.S. Department of Energy by Lawrence Livermore National Laboratory under Contract No. DE-AC52-07NA27344.]

11:10

2aBB9. Experimental and numerical evaluation of the effect of shock waves on the brain. Albert I. King (Dept. of Biomedical Eng., Wayne State Univ., 818 W. Hancock, Detroit, MI 48202, king@rrb.eng.wayne.edu)

A combined experimental and numerical study was conducted to elucidate the mechanical response of a head surrogate under air shock loading. A gel-filled egg-shaped skull/brain surrogate was exposed to blast overpressure in a shock tube environment, and static pressures within the shock tube and the surrogate were recorded throughout the event. A numerical model of the shock tube was developed using the Eulerian approach and it was validated against experimental data. An arbitrary Lagrangian–Eulerian (ALE) based fluid-structure coupling algorithm was then utilized to simulate the interaction of the shock wave with the head surrogate. A comprehensive parametric study was carried out to assess the effect of several key parameters on model response. The curvature of the surface facing the shock wave significantly affected both peak positive and negative pressures. Biological experiments exposing anesthetized rats to shock waves, using the same shock tube, produced brain injury in the form of glial cell activation which in turn can adversely affect the function of axons and neurons. This injury mechanism is not the same as that for blunt impacts to the head which causes direct diffuse axonal injury.

Contributed Papers

11:30

2aBB10. Numerical simulation of mechanisms of blast-induced traumatic brain injury. Jan Arild Teland (Norwegian Defence Res. Establishment, Postboks 25, 2027 Kjeller, Norway, jan.teland@ffi.no), Fredrik Arrhén (SP Tech. Res. Inst. of Sweden, Borås, Sweden), Anders Hamberger (Univ. of Gothenburg, Gothenburg, Sweden), Morten Huseby, Reza Rahimi (Norwegian Defence Res. Establishment, 2027 Kjeller, Norway), Annette Säljö (Univ. of Gothenburg, Gothenburg, Sweden), and Eirik Svinsås (Norwegian Defence Res. Establishment, 2027 Kjeller, Norway)

Blast-induced traumatic brain injury caused by road bombs has lately become a larger part of allied injuries. The same mechanisms may also be responsible for milder injuries of similar nature, resulting from training with large caliber weapons and explosives. In this paper, the blast effects from a weapon on the brain are investigated. Using the hydrocode AUTODYN, numerical simulations of shock wave propagation into the brain are performed. The shock wave is calculated from a complete numerical simulation of the weapon, including the burning gun powder gas inside the barrel, acceleration of the projectile, and the rapid gas flow out of the muzzle. An idealized head is placed in the simulation at the position of personnel firing the weapon. Here we focus on the qualitative mechanisms of the propagation of the shock wave through the skull and into the brain. The results are compared with experiments carried out on anesthetized animals. To simulate real training scenarios, pigs were placed in position of personnel and exposed to

impulse noise generated from weapons. Blast parameters in the air were correlated with those in the brain.

11:45

2aBB11. Influence of skull microstructure on blast-induced pressures in the brain. Joseph A. Turner and Jinjin Liu (Dept. of Eng. Mech., Univ. of Nebraska-Lincoln, Lincoln, NE 68588, jaturner@unl.edu)

The interaction of blast waves with the human head is complicated by many factors including the geometry, material nonlinearities, and skull microstructure. In particular, the flat bones of the skull are comprised of the outer and inner tables (cortical bone) and the diploë (trabecular bone) on the interior. This microstructure results in both scattering and absorption of incident pressure waves. A clear understanding of these effects is needed if pressure profiles within the brain are to be accurately modeled. The focus of this presentation is on finite element wave simulations that have been developed to account for this complex organization. The models are first used to examine the scattering attenuation and coherency loss resulting from the microstructure as a function of incidence angle and frequency for plane wave incident pressure profiles. These models are then extended to more realistic pressure profiles representative of blast waves. The statistics of the microstructure are shown to play a key role in the peak pressures observed within the skull. It is anticipated that this work will lead to a better understanding of role of skull microstructure on blast-induced traumatic brain injury. [Work supported by ARO.]

TUESDAY MORNING, 20 APRIL 2010

KOZMOS, 8:00 TO 11:55 A.M.

Session 2aMU

Musical Acoustics: The Contemporary Traditional Violin

George A. Bissinger, Chair
East Carolina Univ., Dept. of Physics, Greenville, NC 27858
Chair's Introduction—8:00

Invited Papers

8:05

2aMU1. From Catgut to the Violin Society of America: The past 10 years of violin acoustics research. Fan-Chia Tao (595 Smith St., Farmingdale, NY 11735)

In the late 1990s, most of the original founding members of the Catgut Acoustical Society (CAS) had passed away and Carleen Hutchins had departed to concentrate on her new violin octet family. Nevertheless, violin acoustics research continued around the world. In 2002, the CAS and the Violin Society of America (VSA) co-sponsored a violin acoustics workshop at Oberlin College to bring together the world's leading violin acoustics researchers and violin makers. This annual VSA-Oberlin Acoustics workshop has spawned numerous collaborative projects such as George Bissinger's Strad 3D project. In 2005, the CAS merged with the VSA to become the CAS Forum. Members of the CAS Forum have continued to promote violin acoustics by organizing workshops, research projects, and presentations at international meetings of violin makers.

8:30

2aMU2. A finite element approach towards understanding violin structural modes. Colin Gough (School of Phys. and Astronomy, Univ. of Birmingham, Birmingham B15 2TT, United Kingdom)

A COMSOL finite element package has been used to model the structural modes of vibration of the violin treated as a shell structure with orthotropic arched plates. Such computations enable the physical properties of the plates, ribs, and soundpost to be varied over many orders of magnitude. This provides major insight into the nature of the coupling of the top and back plates by the ribs and the role of the soundpost in coupling the radiating “breathing” modes of the violin to the weakly radiating anti-symmetric modes excited by the bowed string induced rocking of the bridge on the central island area. Examples will be shown of the influence of such factors as plate thickness/density, anisotropy, arching, rib strength, and position of soundpost on the frequencies and excited strengths of the strongly radiating “signature” modes below 1 kHz. In addition to determining the mode shapes and frequencies of the excited modes, their contributions to the radiated sound are also computed. Comparison will be made with experimentally observed mode shapes and frequencies.

8:55

2aMU3. The Strad 3D Project: Scientists, musicians, and violinmakers study three classic violins. Samuel Zygmuntowicz (565-A Third St., Brooklyn, NY 11215)

In 2006 the first ever three dimensional (3-D) modal laser scanning of violins was performed on three Guarneri and Stradivari violins, along with acoustic scanning and subjective evaluations. CT scanning was used to determine shape and density properties and to provide a 3-D model for future finite element analyses. These studies and images have been combined with empirical observation, photography, music recordings, and traditional documentation in an interdisciplinary survey of unprecedented scope presented in a two DVD set. From my violinmaker perspective acoustic studies of the violin have had remarkably little impact on the practice of violinmaking. Similarly, traditional empirical traditions continue to be effective in guiding violinmakers, without transferring clinically useful insights to the scientific researchers. This split is reflective of divergent vocabulary, goals, and the requirements of quantitative evidence. The Oberlin Violin Acoustics Workshops has attempted to bridge this gap by bringing together researchers, engineers, musicians, and violinmakers for interactive study, with equal weight given to these separate disciplines. The Strad3D project grows out of that ongoing collaboration and is intended to generate and collect images, documentation, and data that can be viewed and utilized from multiple perspectives and to further a mutually comprehensible dialog.

9:20

2aMU4. Global modeling of the violin radiativity profile. George Bissinger (Dept. of Phys., East Carolina Univ., Greenville, NC 27858, bissinger@ecu.edu)

The violin’s radiativity (pressure/force)profile maintains a consistent shape across quality classes, arguing for a quality-independent generalized global model. In the 196–660 Hz region the lowest cavity modes *A0* and *A1* and the two first corpus bending modes *B1* generate almost all the radiativity, with the *B1* modes treated as “pumps” for *A0* and *A1*. The *B1* modes have nodal patterns similar to the plate bending (primarily) modes 2 and 5, suggesting a link to the violin’s critical frequency. The essentially serial character of the violin’s sound chain leads naturally to a simplified expression for the averaged-over-sphere radiativity profile as a product of just two filters: (1) string-to-corpus through the tuned bridge substructure “gatekeeper”) filter and (2) the “egress” filter for the vibration-radiation transformation, the latter reliably parametrized by the radiation-total damping ratio FRAD. FRAD incorporates the violin critical frequency as well as top and back plate properties in a generalized form. The gatekeeper filter on the other hand is considerably more complex; present bridge models must be augmented by systematic empirical measurements to understand the effects of varying bridge rocking mode frequency or “wing” mass. Recent three-dimensional vibration measurements provide additional insight into bridge-corpus impedance effects.

9:45

2aMU5. The violin: Perceptual studies and acoustical correlates. Claudia Fritz (Institut Jean Le Rond d’Alembert, Université Pierre et Marie Curie, UMR CNRS 7190, 4 place Jussieu, 75005 Paris, France, claudia.fritz@upmc.fr)

This talk discusses the results of experiments in which performances were replayed on different “virtual violins” in order to explore the relationships between acoustical characteristics of violins and perceived qualities. Specifically, it explores perceptual observations reported by Dünwald [based on his measurements of over 700 instruments, *J. Catgut. Acoust. Soc.* (1991)] by modifying the amplitude of the resonance modes over five octave bands (thereby covering the violin’s entire register). When using a subset of the most distinctive verbal descriptors of violin timbre [Fritz *et al.*, Conference of Interdisciplinary Musicology (2009)] to study the relationship between human perception and these acoustical modifications, we ascertained results that partially conflict with Dünwald’s observations. In addition, the study investigated the manner by which one’s perception of the violin’s tone quality is affected by the magnitude of a player’s vibrato as well as the damping of the violin’s resonant modes. Our results do not support the conclusion that liveliness results from the combination of the use of vibrato and a “peaky” violin response. The talk concludes by discussing the limits of such psycho-physical studies, suggesting future directions for psycholinguistic-based research in this domain.

10:10—10:25 Break

10:25

2aMU6. Optimizing the taper-camber relationship in bows for string instruments. John E. Graebner (2 Woodland Rd., Short Hills, NJ 07078, jegraebner@yahoo.com) and Norman C. Pickering (East Hampton, NY 11937)

A violin bow’s taper (graduated diameter) and camber (inward or concave pre-bending of the stick) are important tools used by expert bow makers who know by intuition and experience how to match the two parameters. We have analyzed the static forces in bows and constructed a mathematical model that reveals a simple relationship between taper and camber. For any given taper, the model

predicts a preferred camber and vice versa. A machine has been constructed for accurately measuring the diameter and camber profiles, and measurements on several dozen bows of various degrees of playability provide support for the model.

10:50

2aMU7. Development of the *Titian* Stradivari finite element model. Michael A. Pyrkosz, Charles D. Van Karsen (ME-EM, Michigan Tech. Univ., 1400 Townsend Dr., Houghton, MI 49931), and George Bissinger (East Carolina Univ., Greenville, NC 27858)

As part of the continuing effort to understand the structural and acoustic behavior of one of Stradivari's masterpieces, a finite element (FE) model of the *Titian* Stradivari (1715) has been constructed from computed tomography (CT) scans. The CT data were used to extract high-fidelity geometry and density information specific to this violin. This violin is unique in that it is the only one of Stradivari's instruments that has been measured with a full three-dimensional mobility scan over the top and back plates and ribs as well as acoustic radiativity over a sphere. Hence this solid model can be updated and correlated with this comprehensive experimental data set. The current status of this solid modeling effort will be reviewed in the presentation.

11:15

2aMU8. The violin as a statistical energy analysis network. Evan B. Davis (Brugh Davis Acoustics, 8556 Burke Ave. N., Seattle, WA 98103)

This paper applies elementary statistical energy analysis (SEA) concepts to the violin. Modern makers have been experimenting with ultra-light violins trying to solve the problem increasing the overall output while maintaining the spectral balance of the classical violin. Anecdotal experimental evidence suggests the traditional bridge is not "appropriate" for the ultra-light violins. The ultra-light design problem is used as a case study for hybrid-SEA model. SEA modeling applies a high-modal density, high-frequency, approach which complements the low-frequency, low-modal density, finite element approach. The focus of the SEA modeling is the top plate or belly of the violin addressing the interaction of the cross-arching, plate thickness, wood material properties, and the dynamics of the bridge. The cross-arch of the violin is seen as a critical design feature in the violin. The current approach is more properly a "hybrid method" where the belly and box volume are represented as SEA subsystems and the bridge is represented with a two mode subsystem. The hybrid-SEA analysis demonstrates that the ultra-light's bridge mass must be in the same bridge-mass to belly-mass ratio as the traditional violin to maintain the spectral balance of the classical violin.

Contributed Paper

11:40

2aMU9. Computed cavity-air modes of Le Gruere violin and coupling to corpus modes. C. E. Gough (School of Phys. and Astronomy, Univ. of Birmingham, Birmingham B152TT, United Kingdom)

The internal cavity air modes of both a normal arched violin with f-holes and Calleen Hutchin's Le Gruyère violin with additional holes around the ribs have been computed using COMSOL MULTIDISCIPLINARY ACOUSTICS software. The calculations underpin the need to take finite size corrections into account when evaluating the acoustically important Helmholtz or A0 air resonance. The computed modal shapes and frequencies are compared with

the assumptions and predictions of the Shaw two degrees of freedom network model. This has been widely used to interpret the dependence of the A0 and A1 resonant frequencies on both cavity volume and the number of additional holes opened around the ribs. Finite size effects are shown to have a marked influence on such dependencies. They are shown to result in marked deviations from predictions based on an ideal Helmholtz resonator. The coupling of the air modes to the corpus modes that excite them is considered using a simple dynamic model. The predictions of the model are compared with those of the Shaw network model and measurements of the dependence of A0 and A1 mode frequencies on the number and placing of additional holes opened around the ribs of the Le Gruyère violin.

TUESDAY MORNING, 20 APRIL 2010

GRAND BALLROOM V, 8:00 TO 9:00 A.M.

Session 2aNCa

NOISE-CON: Plenary

Michael J. Lucas, Chair
Ingersoll-Rand, P.O. Box 1600, Davidson, NC 28036

Chair's Introduction—8:00

Invited Paper

8:05

2aNCa1. Noise and vibration phenomena in aircraft wheel and brake systems. Todd E. Rook (Aircraft Wheels and Brakes, Goodrich Corp., 101 Waco St., Troy, OH 45373, todd.rook@goodrich.com)

There is a wide variety of noise and vibration phenomena in aircraft brake systems for which must be accounted in the design process of such systems. These phenomena include such modes as whirl and squeal, the latter of which can be quite different from its counterpart in automotive systems and has likewise received much less attention in the literature than its automotive counterpart. Consequently, an overview of such phenomena with representative results from simulations and experiments will be presented to highlight the differences. Complicating matters is that brake-induced vibration often involves strong coupling with the aircraft structure, thereby necessitating a system level understanding beyond the brake itself. This aspect poses a particular problem to a brake component supplier in how to ensure favorable noise and vibration behavior for the full aircraft system, particularly early in the development cycle. To

accomplish this goal, Goodrich has developed a successful methodology combining simulation and testing to assess the noise and vibration behavior at an earlier stage. The methodology implements simulations to guide appropriate brake component tests that are more tractable for the brake supplier yet sufficiently mimic the full system environment. With the combined simulation and testing, a reduction in actual testing can be achieved while improving component noise and vibration performance.

8:45—8:50 Questions

8:50—9:00 Announcements

TUESDAY MORNING, 20 APRIL 2010

GRAND BALLROOM I/II, 9:15 A.M. TO 12:00 NOON

Session 2aNCb

NOISE-CON and Engineering Acoustics: Experimental Techniques and Instrumentation in Noise and Vibration

Jim Thompson, Chair

Bruel & Kjaer North, 6855 Commerce Rd., Canton, MI 48187

Contributed Papers

9:15

2aNCb1. Enhancing accuracy in reconstruction of vibro-acoustic responses of a complex structure using Helmholtz equation least squares based nearfield acoustical holography. Logesh Kumar Natarajan and Sean F. Wu (Dept. of Mech. Eng., Wayne State Univ., Detroit, MI 48202)

In traditional nearfield acoustical holography, the acoustic pressures in the near field are measured, and the entire acoustic field including the normal component of the surface velocity are reconstructed. This type of approach may be appropriate for reconstructing the acoustic pressure field, but not for reconstructing normal surface velocity in practice because not enough nearfield information is collected to yield satisfactory reconstruction of the normal surface velocity. In this paper, we propose to supplement the acoustic pressure measurements with a few normal surface velocity measurements, which can be used as benchmarks in optimizing the reconstruction of normal surface velocity. These data are taken as input to the Helmholtz equation least squares method based NAH to reconstruct the acoustic pressure and normal velocity on the surface of a vibrating structure. The accuracy in reconstruction is examined experimentally on a baffled rectangular plate with clamped boundary conditions subject to random excitations. To validate the results, the reconstructed field acoustic pressures are compared with those measured by microphones, and the normal surface velocities were compared with the benchmark values collected by using a scanning laser vibrometer under the same condition.

9:30

2aNCb2. Planar nearfield acoustical holography in high-speed subsonic flow. Yong-Joe Kim (Dept. of Mech. Eng., Texas A&M Univ., 315 Eng./Phys. Bldg. Office Wing, College Station, TX 77843-3123), Hyu-Sang Kwon (Korea Res. Inst. of Standards and Sci.), and Seunghwan Jung (Texas A&M Univ., College Station, TX 77843-3123)

The objective here is to develop a nearfield acoustical holography (NAH) theory that includes the effects of high-speed, subsonic flow. Recently, the speed of a transportation means such as aircraft or vehicle has significantly increased, e.g., too close to the speed of sound. As a result, the NAH data measured with a microphone array fixed on an aircraft or vehicle entail significant airflow effects such as a Doppler shift in wave number domain. Here, a convective wave equation is used to define a mapping function. The mapping function is then used to derive a NAH theory that includes the Doppler shift effect by mapping the conventional NAH procedure in the wave number domain. A numerical simulation with a monopole is performed at the high-speed airflow of Mach = 0.7. The reconstructed acoustic fields obtained by applying the proposed NAH procedure to the

simulation data match well with the exact fields. Through an experiment with two loudspeakers performed in a wind tunnel at the speed of Mach = 0.1, it is also shown that the proposed NAH procedure can be used to successfully reconstruct the sound fields radiated from the two loudspeakers.

9:45

2aNCb3. A method for measuring indoor noise levels from traffic with ulterior noise sources. Daniel Oldakowski (Polysonics Corp., 405 Belle Air Ln., Warrenton, VA 20186, danielo@polysonics.com)

This paper describes a measurement protocol used to determine indoor noise levels due to motor vehicle traffic traveling on Route 395/Route 295 flyover in southeast Washington, DC. Measurements were taken over a period of 24 h, in multiple locations of a 14-story high rise building, located in the vicinity of a baseball stadium. The challenge was to remotely measure the indoor noise due to traffic in the presence of unknown and unpredictable such as construction activity inside and outside the building during the day and nearby baseball field activity at night. Multiple sound level meters were employed, inside and outside the building. One-third octave band data, with a 1-min resolution, were taken for the duration of the measurement, as is typical for indoor noise measurements. Partial recordings of the sound field time history within the building were taken to document "loud" events. During postprocessing, loud events were manually inspected to determine if those events were due to traffic noise or due to other noises.

10:00

2aNCb4. Angle dependent effects for impulse noise reduction for hearing protectors. William J. Murphy, Amir Khan, and Edward L. Zechmann (CDC/NIOSH Hearing Loss Prevention Team, 4676 Columbia Parkway, MS C-27, Cincinnati, OH 45226)

The proposed U.S. Environmental Protection Agency regulation for labeling hearing protection devices (HPDs) includes an impulsive noise reduction rating. In 2009, the American National Standards Institute Subcommittee for noise approved a revised standard for measuring the impulsive insertion loss of HPDs, ANSI/ASA S12.42-2009. The exposure at the ear in response to a forward-propagating wave depends strongly on the orientation of the head with respect to the direction of propagation. Furthermore, the insertion loss varies with the peak sound pressure level. This paper reports the results of tests performed using an acoustic shock tube to produce peak impulses of approximately 160-dB peak sound pressure level. Two manikins were evaluated: the GRAS KEMAR manikin equipped with 1/2 and 1/4 in. microphone in a GRAS 711 IEC coupler and the Institute de Saint Louis manikin equipped with a Bruel & Kjaer IEC 711 coupler equipped with a

1/4 in. microphone. The manikin heads were rotated through ± 90 deg relative to the direction of the oncoming wavefront and impulsive peak insertion loss was measured according to S12.42-2009. [Portions of the research were supported by U.S. EPA Interagency Agreement No. 75921973-01-0.]

10:15—10:30 Break

10:30

2aNCb5. Development of NASA Glenn Research Center auditory demonstration laboratory facility and operational capabilities. Beth Cooper (NASA Glenn Res. Ctr., 21000 Brookpark Rd., MS 49-2, Cleveland, OH 44135, beth.a.cooper@nasa.gov), Jeff G. Schmitt (ViAcoust., Austin, TX), and David A. Nelson (Nelson Acoust., Elgin, TX)

The NASA Glenn Research Center Auditory Demonstration Laboratory (ADL) is a dual purpose facility, constructed in 2007 to support hearing conservation programs across the agency. Configured as a reverberant room, the ADL is an appropriate space for evaluating the performance of personal hearing protectors, using either human subjects or a test fixture. Hearing protector evaluations are conducted using NASA REATMASTER software, developed in partnership with the National Institute for Occupational Safety and Health. This software is available free on request to qualified laboratories, which are encouraged to participate in a collaborative program to fund continued software development. The ADL can also be configured as a free-field room to support the development of auditory demonstrations, widely used for a variety of training purposes within NASA and externally. The ADL provides an environment, sound system, and audio engineering tools for presenting and developing calibrated demonstrations of various acoustical and auditory phenomena that include fundamental acoustical and concepts, noise control principles, and simulations of hearing loss. Current work at the ADL will establish the capability of making three-dimensional surround sound recordings, which will expand the scope of the laboratory's educational products into additional areas of psychoacoustics such as binaural hearing and localization.

10:45

2aNCb6. Application of microphone frequency response and windscreen insertion-loss corrections on sound power determinations. Einar Ristroph, Michael Black, and John Phillips (1301 Arrow Point Dr., Cedar Park, TX 78613, einar.ristroph@ets-lindgren.com)

Engineering-grade product noise emission testing programs are qualified, and correction factors are applied to test results, based on "known" sound power levels of a reference sound source; i.e., a sound source that has been tested and qualified in accordance with ANSI S12.5/ISO 6926. The accuracy and uncertainty of ANSI S12.5/ISO 6926 test results are then transferred to the engineering-grade testing program and are thus of crucial importance. ETS-Lindgren recently commissioned the new Acoustic Research Laboratory at headquarters in Cedar Park, TX, and is developing an ANSI S12.5/ISO 6926 reference sound source testing program. This work included evaluation of factors contributing to accuracy and uncertainty. The application of microphone frequency response corrections (obtained from outside calibration laboratory) is discussed and data are presented. In addition, the use of windscreen insertion-loss corrections is discussed and data are presented.

11:00

2aNCb7. Anechoic chamber verification for calibration of reference sound sources. Kevin Herreman (Owens Corning Corp., 2790 Columbus Rd., R#16 Granville, OH 43023, kevin.herreman@owenscorning.com)

Confidence in the capability of a testing facility to perform repeatable/reproducible product testing mandates study of each test chamber utilized by the facility. The Owens Corning Acoustic Research Center (OCARC) provides product sound power level determination for many differing product types ranging from the power generation, appliance, automotive, to medical equipment industry utilizing a fully anechoic chamber with a removable horizontal reflecting plane. Recently the OARC completed a verification study identifying the performance of the chamber relative to the requirements of test standard ISO 6926: Requirements for the performance

and calibration of reference sound sources used for determination of sound power levels and ANSI S12.5: Requirements for the performance and calibration of reference sound sources. These standards provide strict guidelines for the repeatability/reproducibility of test results. This paper will review the verification testing process and present results from the study.

11:15

2aNCb8. Developing a basis for efficient railroad horn testing. John Erdreich and Joseph Keefe (Ostergaard Acoust. Assoc., 200 Executive Dr., Ste. 350, W. Orange, NJ 07052, je@acousticalconsultant.com)

In response to complaints of excess noise from residents living in the vicinity of rail grade crossings, the Federal Railroad Administration promulgated regulations mandating minimum and maximum horn sound output levels. Older fleets require testing of horns to ensure compliance with these limits. Test requirements (no reflecting surfaces within 200 ft of the horn; wind less than or equal to 12 m/ph; no precipitation) severely limit the ability of urban commuter railroads to comply with the testing, especially in northern and coastal areas. To develop an alternate test protocol, measurements of horn sound levels were carried out in a semi-anechoic chamber and outdoors to determine if a reliable transfer function could be constructed to convert the chamber measurements to the outdoor measurements. Standard deviations of the A-weighted chamber measurements were smaller than standard deviations of the outdoor measurements. For one fleet, the differences between chamber measurements and outdoor measurements resulted in a consistent difference of about 41 dB(A). Locomotive supply air pressure differences at the horn were not a significant factor in sound output.

11:30

2aNCb9. Measuring recreational firearm noise. Per Rasmussen (G.R.A.S. Sound & Vib. A/S, 33 Skovlytoften, 2840 Holte, Denmark, pr@gras.dk), Greg Flamme (Western Michigan Univ., Kalamazoo, MI), Michael Stewart (Central Michigan Univ., Mount Pleasant, MI), Deanna Meinke (Univ. of Northern Colorado, Greeley, CO), and James Lankford (Northern Illinois Univ., DeKalb, IL)

Recreational use of firearms in the United States is commonplace. There are 28×10^6 Americans who consider themselves hunters and 13×10^6 went hunting in 2000. Participation in the shooting sports, without the use of properly worn hearing protection, exposes the involved persons to high levels of impulsive noise which may cause hearing loss and/or tinnitus (ear ringing). The present study was initiated to gain a better understanding of the noise exposure created by contemporary firearms using state of the art instrumentation and to ultimately increase our knowledge and awareness of this unique noise hazard. The sound pressure signal created by recreational firearms as used in hunting or target practice is characterized by a high-frequency, short duration impulsive noise. This signal is perceived by the human ear as one single, loud impulse or "shot." However, when the firearm sound level is measured with microphones capable of sampling wide frequency ranges and combined with high-speed data acquisition computer systems, the impulses can be resolved into a number of different acoustic signals related to different source mechanisms.

11:45

2aNCb10. Exposure to recreational/occupational shooting range noise versus industrial impulsive noise. Hale Marlund (Adv. Eng. Acoust., 663 Bristol Ave., Simi Valley, CA 93065-5402, noisedoc@aol.com)

Public shooting ranges are used primarily for recreational shooting. Certain occupations require periodic certification of firearms proficiency. Single gunfire noise is loud and impulsive. However, both recreational and occupational proficiency shooters are often exposed to firearm noise from other shooters. In such cases, the shooting noise exposure can include gunfire impulses less than 1 s apart. This situation changes noise exposure from "impulsive" to "continuous" during each open-fire session. A similar situation occurs in industrial applications when the work process involves impulsive fastening activities, such as pneumatic riveting. The similarities in high-intensity noise exposure at busy shooting ranges and production line riveting include time-weighted averages, under the current OSHA regulations, that are less than 90 dB(A) but consistently exceed 140 dB(C) peak. As a result, even though firing range and workplace rules require the use of hearing pro-

tection, it is not uncommon for recreational or occupational shooters and industrial workers to be casual about the fitting and proper use of their hearing protection devices. This paper describes the similarities in daily impulsive

noise exposure at the shooting range and fastening production line. It also discusses the potential dangers in both situations with ill fitting and best practice use of hearing protectors.

TUESDAY MORNING, 20 APRIL 2010

GRAND BALLROOM III/IV, 9:15 TO 11:30 A.M.

Session 2aNcC

NOISE-CON and Noise: Automotive and Powertrain Noise and Vibration

Gordon L. Ebbitt, Cochair

Carcoustics, 29770 Hudson Dr., Novi, MI 48377

Terrence Connelly, Cochair

ESI North America, 32605 W. 12 Mile Rd., Ste. 350, Farmington Hills, MI 48334-3379

Contributed Papers

9:15

2aNcC1. Transient in cab noise investigation on a light duty diesel passenger vehicle. Dhanesh Purekar (Cummins Inc., 2727 West 450 South Walesboro, Columbus, IN 47201, dhanesh.purekar@cummins.com)

A diesel engine in cab sound quality for passenger car market is scrutinized more closely than in the mid- to heavy duty diesel truck applications. This is obviously due to the increasing expectations from the customers for gasolinelike sound quality. This paper deals with a sound quality issue recently investigated on a light duty diesel engine for a passenger van application. The objectionable noise complaint occurred during the vehicle transient operating conditions and was found to be caused by the change in the pilot quantity over a very short period of time. The root cause of the noise complaint was investigated on the noise complaint vehicle as well as simultaneously on a standalone engine in the noise test cell. Several critical combustion and performance parameters were recorded for diagnosing the issue. In addition, various standard sound quality metrics were employed to differentiate the sound quality of the objectionable noise. The issue was resolved and verified by making appropriate changes to the engine calibration without affecting key requirements such as emissions and fuel economy. Finally, the findings from the experimental tests are summarized and appropriate conclusions are drawn with respect to understanding, characterizing, and resolving this transient, combustion related impulsive powertrain interior noise issue.

9:30

2aNcC2. Two-substructure, time-domain transfer path analysis of transient dynamic response of mechanical systems with nonlinear coupling. Wenwei Jiang and Teik Lim (Univ. of Cincinnati, 688 Riddle Rd., Apt. 800L, Cincinnati, OH 45220, jiangwe@mail.uc.edu)

The traditional frequency domain transfer path analysis has become a popular method to detect and diagnose vehicle NVH problems. Unfortunately, due to its reliance on frequency transfer functions, the approach is strictly effective only for linear time-invariant system and steady-state response. This limitation prevents the method from being applied to transient and/or nonlinear behavior such as clunk, shudder, and tip-in and tip-out. In this paper, a novel time domain transfer path analysis is proposed to deal with a class of non-linear transient response problems. It combines the versatility of the transfer path analysis for tracking transmission of vibratory energy between substructures and generality of the time domain analysis for treating transient response that is also nonlinear in nature. The formulation that is derived by combining the spectral-based substructure method and a discrete, piecewise convolution theory is applied to a lumped parameter system, and the results are compared with output from a direct numerical integration of the nonlinear equations of motion. Comparison results show significant promise and appear to be usable for solving real-life vehicle problems that are highly transient with moderate level of nonlinearity present.

9:45

2aNcC3. Refinement of vehicle interior noise by reduction of driveline vibrations. Okan Tandogan, Tarkan Yapici, Mert Doganli, and Caner Sevginer (Tems R&D & Technol. Corp., Tubitak MAM Technol. Free Zone, 41470 Gebze-Kocaeli, Turkey, okan.tandogan@tems-argetek.com.tr)

Driveline vibrations can be a significant vibration source, which require emphasis in vehicle NVH development studies. In the early design phases, it is usually not easy to predict the vehicle level NVH performance induced by the torsional irregularities of the driveline. Establishing a model with inertia, mass, and stiffness data taken from engine, transmission and driveline suppliers can be possible and useful to initiate preliminary cautions. However, it is first, a challenging job for OEMs to receive all relevant data from suppliers to create a reliable model. Second, driveline vibration-driven NVH response of a vehicle is mostly figured out in real-time testing. Therefore, it is of great importance to treat driveline vibrations as a potential source of customer concern in vehicles and should be investigated throughout the testing phase. In this study, a dissatisfying noise behavior of a coach is determined with objective data and defined by means of engine operating condition, frequency, and excitation orders. As a next step, the problem is cascaded to its possible sources by examining the exciter's inputs and the transfer paths. Finally, the important parameters in the reduction in driveline torsional vibrations and, as a consequence, the refinement of vehicle interior noise are discussed.

10:00

2aNcC4. Calculation of optimal damping placement in a vehicle interior. Craig Birkett (Daimler Trucks North America, 4747 N Channel Ave., C3B-EA Portland, OR 97217, craig.birkett@daimler.com), Poh-Soong Tang, and Dieter Featherman (Altair Eng.)

A study was performed of a heavy duty truck cab to reduce interior noise with the objective of using a minimum amount of damping material. Several complementary tools were applied to the problem of determining damping material layout. Candidate locations for damping material were first identified by summing *A*-weighted velocities over the frequency range of interest. These velocities could be additionally weighted by acoustic sensitivity. Then an automatic optimization was performed using structural inputs to determine the optimal damping treatment from the candidate damping patches given weight constraints. Results were confirmed with testing on a vehicle dyno. The study was significant because it compared various practical methods of optimizing a vehicle interior.

10:15—10:30 Break

10:30

2aNCc5. Vibration transfer path analysis on fender noise control. Paul Liang (Harley-Davidson Motor Co., 11800 W. Capitol Dr., Wauwatosa, WI 53222, paul.liang@harley-davidson.com)

The fender is one of the major noise radiators in motorcycles. The source of the fender noise comes primarily from the engine vibration through the vehicle frame. A rubber isolator between the engine and frame can effectively break the vibration path, thus reducing the fender noise. A finite element method will be used to analyze the vibration transfer paths of an engine and a boundary element method will be used to predict the radiated noise from the rear fender.

10:45

2aNCc6. Prediction of muffler insertion loss and shell noise by a hybrid finite element acoustic statistical energy analysis model. Terence Connelly (32605 W, 12 Mile Rd., Ste. 350 Farmington Hills, MI 48334-3379, tco@esi-group.com), Steve Mattson, Dave Labyak, and Jeff Pruetz (Great Lakes Sound & Vib.)

A reactive automotive style muffler was used to evaluate and experimentally validate the numerical predictions of muffler acoustic performance. A CAD model of the silencer was developed and an acoustic FE mesh created. The interior of the muffler included two sections of perforated pipe, which were included in the cavity mesh. A hybrid FE-statistical energy analysis (SEA) numerical model was created from the finite element acoustic mesh and was excited by a diffuse acoustic field at the inlet and coupled via hybrid junctions to SEA semi-infinite fluids on both the inlet and outlet. From the hybrid acoustic model different FE-SEA modeling approaches were investigated to predict the shell noise. The muffler insertion loss and shell noise was measured experimentally using a broadband acoustic source piped into a hemi-anechoic chamber. A straight pipe with simple bends to recreate the path from muffler inlet to outlet was fabricated for comparison. Muffler insertion loss was calculated by means of a simple level difference between the silencer and the straight pipe. Shell vibration and radiated noise were also measured to validate the shell noise modeling. Agreement between experimental measurement and numerical prediction was found to be reasonably accurate up to 3 KHz.

11:00

2aNCc7. Innovative quieter aspirator design for in-car temperature sensor. Niranjana G. Humbad, Dan Silaghi, and Matthew Morris (Behr America, Inc., Behr America, Inc. 2700 Daley Dr., Troy, MI 48083, niranjana.humbad@us.behrgroup.com)

Many vehicles use automatic temperature sensor mounted on dash [instrumentation panel (IP)] to sense the vehicle interior temperature for controlling the climate comfort. One type of design requires airflow over the sensor to correctly measure the interior vehicle temperature. This suction airflow over sensor is produced by various means. The present study deals with a widely used aspirator design to draw/suck interior airflow from venturi effect caused by discharging HVAC module flow. Such a design has an inherent drawback of bringing the HVAC module noise and flow noise (generated by aspirator) to an opening mounted on dash (via a connecting tube from sensor to aspirator). In one application, this caused a customer noise complaint; noise levels were higher at frequencies above 2 kHz. This required a separate muffler part between aspirator and sensor to reduce this noise. A new innovative aspirator design was developed to replace the aspirator and muffler by a single part. This one piece assembly was designed to be easy for manufacturing but still keeping the same performance targets of the two part aspirator and muffler design. Test results of the new design are compared to the current design and shows meeting both the airflow and noise targets.

11:15

2aNCc8. Passenger vehicle exhaust cool down noise measurement and sound assessment. Antoni Szatkowski and Brian Butler (Chrysler LLC, NVH Development & Eng., P.O. Box 21-8004, Auburn Hills, MI 48321-8004, as32@chrysler.com)

Passenger vehicle exhaust cool down is getting more attention as NVH vehicles quality is more and more refined. A new test procedure was developed and presented in this paper. Data acquisition, processing, and sound qualifying are presented. Four different metrics were developed that quantify the sound level and sound quality. A few examples of different exhaust systems were analyzed. Presented methodology is a great tool for different exhaust system ranking, benchmarking, comparison, and target setting.

TUESDAY MORNING, 20 APRIL 2010

GRAND BALLROOM VII/VIII, 9:15 TO 10:15 A.M.

Session 2aNCd

NOISE-CON and Physical Acoustics: Flow Noise

Dean E. Capone, Chair

Pennsylvania State Univ., Applied Research Lab., P.O. Box 30, State College, PA 16801

Contributed Papers

9:15

2aNCd1. The noise from flow over rough surfaces with small and large roughness elements. Stewart Glegg (Dept of Ocean and Mech. Eng., Florida Atlantic Univ., Bld 36, 777 Glades Rd., Boca Raton, FL 33431, glegg@oe.fau.edu) and William Devenport (Virginia Tech)

This paper will review theoretical and experimental results for the sound radiation from turbulent boundary layer flows over rough surfaces. Two distinctly different regimes will be considered. The first is hydrodynamically smooth surfaces for which the surface roughness does not impact the boundary layer turbulence. These surfaces have been shown experimentally and theoretically to radiate enhanced sound levels due to the scattering of the hydrodynamic pressure fluctuations by the roughness. The second regime considers large roughness elements that directly impact the boundary layer flow. Numerical calculations show that the flow separates around the large roughness elements and theoretically this implies that the roughness radiates

sound in proportion to the unsteady force on each element. It is still an unresolved issue as to whether the unsteady loading is caused by the distortion of the turbulent flow around the element or the unsteady loading associated with the flow separation. This paper will describe the theoretical approaches which have been used to predict roughness noise levels in each of these two regimes and the transition from one regime to the other.

9:30

2aNCd2. Low wavenumber turbulent boundary layer fluctuating wall shear stress measurements from vibration data on a cylinder in pipe flow. William Bonness, Dean Capone, and Stephen Hambric (Appl. Res. Lab., The Penn State Univ., P.O. Box 30, State College, PA 16804, wkb3@only.arl.psu.edu)

Fluctuating wall shear stress under turbulent boundary layer (TBL) excitation is studied in this experimental investigation. A cylindrical shell with

a smooth internal surface is subjected to TBL excitation from water in fully developed pipe flow at 6.1 m/s. The vibration response of the cylinder is used to inversely determine low-wavenumber TBL shear stress levels. Both the cross-flow and streamwise directions are examined using directionally uncoupled low-order cylinder modes in the circumferential and axial directions. These data address a critical gap in available literature regarding experimental low-wavenumber shear stress data. The low-wavenumber shear stress levels in both the cross-flow and streamwise directions are determined to be roughly 10 dB higher than those of normal pressure. As is the case for various models of TBL pressure, these measurements suggest that a nearly constant value for normalized shear stress at low wavenumber is valid over a broad range of frequencies. A simple wavenumber white model form established for low-wavenumber TBL surface pressure is also shown to be appropriate for shear stress.

9:45

2aNCd3. Analysis of sound measurements inside a finite length ducted rotor system. Scott Morris, Jason Tomko, and David Stephens (Univ. of Notre Dame, Notre Dame, IN 46556, s.morris@nd.edu)

The sound generated by a ducted rotor system leads to a complicated acoustic field inside the duct. The source can have both broad-band and tonal sound features. The interior Green's function will add additional complexity due to the cut-on modes and finite-length (organ pipe) effects. In this study, a simple ducted rotor was considered experimentally and analytically in order to obtain observations and predictions of these features. The results will be an important component of modeling the structural vibration in systems where the rotor operates in an elastic shell.

10:00

2aNCd4. Aero-acoustic predictions of automotive dashboard HVAC (heating, ventilating, and air-conditioning ducts). Stephane Detry (VISTEON Systèmes Intérieurs Ctr., Visteon Interior System, Technique Rue Lon Duhamel, BP 87, 62440 Harnes, France, sdetry@visteon.com), Julien Manera, Yves Detandt, and Diego d'Udekem (Free Field Technologies)

The flow-induced noise generated by automotive climate control systems is today emerging as one of the main noise sources in a vehicle interior. Numerical simulation offers a good way to analyze these mechanisms and to identify the aerodynamic noise sources in an industrial context driven by permanent reduction in programs timing and development costs, implying no physical prototype of ducts before serial tooling. This paper focuses on a numerical aeroacoustic study of automotive instrument panel ducts to estimate the sound produced by the turbulent flow. The methodology is the following: the unsteady-flow field is first computed using a CFD solver—here FLUENT. Then, the acoustic finite element solver ACTRAN computes the acoustic sound sources from these time domain CFD results. The sources are finally propagated into the vehicle interior in the frequency domain. One advantage of the technique is that the CFD computations are completely separated from the acoustic computations. This allows reusing one CFD computation for many different acoustic computations. The theoretical background is presented in the first sections of this paper. Then, the accuracy of the method for real industrial cases is demonstrated by comparing the numerical results to experimental results available at VISTEON.

2a TUE. AM

TUESDAY MORNING, 20 APRIL 2010

GRAND BALLROOM VII/VIII, 10:30 TO 11:45 A.M.

Session 2aNCe

NOISE-CON, Noise, and Physical Acoustics: Sound Propagation in the Atmosphere

Kai Ming Li, Chair

Purdue Univ., School of Mechanical Engineering, 140 Martin Jischke Dr., West Lafayette, IN 47906

Contributed Papers

10:30

2aNCe1. Frequency-dependent propagation characteristics in and around forests. Michelle E. Swearingen (Construction Eng. Res. Lab., USA ERDC, P.O.B. 9005, Champaign, IL 61826, michelle.e.swearingen@usace.army.mil), Donald G. Albert (Cold Regions Res. and Eng. Lab., USA ERDC, Hanover, NH 03755-1290), Michael J. White, and Patrick Guertin (Construction Eng. Res. Lab., USA ERDC, Champaign, IL 61826)

Sound propagation in and around forests is highly influenced by the unique vegetative environment. The array of scattering objects represented can be parametrized by parameters such as tree height and diameter, spatial arrangement, and areal density. The array can be in a lattice or random configuration, depending on whether the trees were intentionally planted or naturally occurring. The trees themselves can be of uniform age and size or a mixture of sizes and types. Using data from three different study areas, this presentation will explore correlations between the physical structure of the forest and frequency-dependent attenuation of impulsive sounds.

10:45

2aNCe2. Meteorological influence on highway noise barrier performance: A case study. Paul Burgé (URS Corp., 1615 Murray Canyon Rd., Ste. 1000, San Diego, CA 92108, paul_burge@urscorp.com), Jon Sytsma, and Tom Zurburg (Michigan Dept. of Transportation)

Local meteorological conditions such as cross-winds and temperature gradients have long been recognized as factors that can negatively affect

noise barrier performance. However, this relationship is seldom demonstrated with empirical data and neither are prevailing meteorological conditions routinely taken into consideration in the design of highway noise barriers. In this case study, a post-construction acoustical investigation was undertaken for a barrier that was installed as part of a recent highway improvement project and where neighborhood residents persistently complained that the barrier was not providing expected noise reduction. Several days worth of noise level measurements and meteorological data were collected and analyzed. The results of the analysis indicated that, while the area residents were not actually impacted under applicable federal and state noise policy, the noise barrier performance was indeed being significantly influenced by documented meteorological conditions. Interaction with area residents also revealed a remarkable example of the wide variation in different individuals' subjective response to virtually the same noise environment.

11:00

2aNCe3. Quasi-wavelet cascade models for intermittent random media and application to wave scattering. D. Keith Wilson (U.S. Army Engineer Res. and Development Ctr., 72 Lyme Rd., Hanover, NH 03768, d.keith.wilson@usace.army.mil), Vladimir E. Ostashev (NOAA Earth System Res. Lab., Boulder, CO 80303), George H. Goedecke (New Mexico State Univ., Las Cruces, NM 88003), Soren Ott (Risoe Natl. Lab. for Sustainable Energy, Roskilde, Denmark), and Donald G. Albert (U.S. Army Engineer Res. and Development Ctr., 72 Lyme Rd., Hanover, NH 03768)

Terrestrial environments often possess intermittent distributions of scattering objects. Examples include atmospheric turbulence, subsurface geology, vegetation, and buildings. A quasi-wavelet (QW) cascade process model for such intermittent random media is described, and the implications for wave scattering are examined. The QW model builds the random medium from randomly positioned and oriented, wavelet-like entities, which follow prescribed distributions for number density and energy vs spatial scale. Different types of QWs, including monopole and dipole scalar fields and toroidal and poloidal vector fields, can be combined with statistically preferred orientations to create multiple field properties possessing correlated properties and anisotropy. The spatially localized nature of the QWs facilitates construction of intermittent random fields in a manner that would be extremely challenging, if not impossible, with conventional Fourier approaches. To test the QW model, we conducted a seismic propagation experiment in the vicinity of a volcanic crater in the Mojave Desert. This site was chosen for its highly inhomogeneous, intermittent distribution of basalt and sand. Propagation of impulse signals was sampled along 864 distinct paths. Statistical distributions of seismic travel times were simulated with good success using a finite-difference, time-domain method applied to a QW model for the site geology.

11:15

2aNcE4. A numerical study of impulse propagation over a hill. Santosh Parakkal, Xiao Di, and Kenneth E. Gilbert (Natl. Ctr. for Physical Acoust., Univ. of Mississippi, University, MS 38677)

In preparation for a field experiment, a numerical study of impulse propagation over a hill has been carried out. The near side of the hill corresponds to downward refraction with the associated ducted propagation. The far side of the hill corresponds to upward refraction with the associated shadow zone. Thus conditions corresponding to nighttime propagation (downward refraction) and daytime propagation (upward refraction) can be

studied in the same experiment. Of particular interest is the propagation of surface waves over the hill and into the shadow zone on the far side of the hill. For selected values of hill curvature and ground impedance, numerical predictions are presented and discussed in relation to the envisioned experiment. [Research supported by the U.S. Army TACOM-ARDEC at Picatinny Arsenal, New Jersey]

11:30

2aNcE5. Portable loudspeaker coverage capability for in-situ outdoor performance spaces. Juan Arvelo, Shawn Johnson, and Ronald Mitnick (Appl. Phys. Lab., The Johns Hopkins Univ., 11100 Johns Hopkins Rd., Laurel, MD 20723-6099, juan.arvelo@jhuapl.edu)

Observations from daily experiences reveal that sound propagation in air is influenced by the ground topography, atmospheric stratification, winds, and turbulences. A ray-based outdoor loudspeaker coverage model was developed that accounts for terrain and atmospheric conditions. Loudspeaker coverage may be adjusted to various levels of sound or speech intelligibility. Comparison against benchmark closed-form solutions and wave-based approaches demonstrate the accuracy and computational efficiency of this Gaussian-ray bundle model at aural frequencies. This model is well suited as an acoustic design aid for outdoor performance spaces. This software was packaged into an ultra-mini notebook computer with an integrated GPS antenna and a 90-m resolution worldwide terrain database to account for *in-situ* terrain effects on sound propagation, a built-in microphone for calibrated measurement of the background noise level and the loudspeaker source level, and a USB real-time logger of temperature and relative humidity. The model is also capable of importing atmospheric measurement from balloon-launched radiosondes and various atmospheric models to accurately account for atmospheric stratification in temperature, pressure, relative humidity, wind speed, and direction in forensic investigations. Atmospheric turbulence has been shown to be another important factor that must be included in the next generation portable loudspeaker coverage capability.

TUESDAY MORNING, 20 APRIL 2010

DOVER C, 8:15 A.M. TO 12:00 NOON

Session 2aNsa

Noise, Architectural Acoustics, and INCE: Ventilation, Fan, and Duct Noise Control I

Lixi Huang, Cochair

The Univ. of Hong Kong, Dept. of Mechanical Engineering, Pokfulam Rd., Hong Kong, P.R. China

Kirill V. Horoshenkov, Cochair

Univ. of Bradford, School of Engineering, Great Horton Rd., Bradford, BD7 1DP, UK

Jian Kang, Cochair

Univ. of Sheffield, School of Architecture, Western Bank, Sheffield, S10 2TN, UK

Chair's Introduction—8:15

Invited Papers

8:20

2aNsa1. Modeling the acoustic interaction between components in ventilation ductwork. Ray Kirby (Mech. Eng., School of Eng. and Design, Brunel Univ., Uxbridge, Middlesex UB8 3PH, United Kingdom)

The modeling of sound propagation in ventilation systems normally focuses on understanding the acoustic performance of individual components within the system. For example, theoretical models have long been available for ventilation silencers, elbows, branches, and bends. However, to fully understand the overall acoustic performance of a system, it is necessary also to investigate the interaction between these components. This may be achieved by using numerical methods such as the finite element method and in principle current commercial packages may now be used to examine multiple components in ventilation systems. However, commercial packages normally rely on meshing the entire ventilation system and the computational cost of such an approach quickly becomes

prohibitive at higher frequencies. An alternative, and computationally more efficient, approach is to use a modal expansion for the sound pressure field in uniform duct sections and to apply a full finite element analysis only to non-uniform elements, such as corners and/or branches. Accordingly, predictions for a ventilation system that contains multiple components are reviewed here, and the acoustic interaction between these components is examined in order to gain a better understanding of the *in situ* acoustic performance of important components such as the ventilation silencer.

8:40

2aNSa2. Simulation of acoustic noise in duct systems. M.G. Prasad and B. Rajavel (Dept. of Mech. Eng., Noise and Vib. Control Lab., Stevens Inst. of Technol., Hoboken, NJ 07030, mprasad@stevens.edu)

Acoustical noise studies of automotive muffler, HVAC, and other duct systems are important in their design and performance. A methodology is developed for simulating the acoustical noise from a duct system which can be modeled in terms of source, duct element, and termination. In this methodology first the transfer function of the preliminary design of the duct system is obtained using system geometry, source, and radiation impedances. Then the impulse response of the obtained transfer function from the preliminary design is convolved with source signal to obtain the acoustic noise output of the system in frequency domain. In order to verify this methodology, the simulated noise is compared with experimentally measured noise for two types of duct systems, namely, a straight pipe and a simple expansion chamber. The results show good agreement between the simulated and measured noise spectra. The proposed methodology is then applied to obtain a simulation of noise reduced due to automotive muffler system. The methodology presented in this work provides the capability of simulating the noise of a duct and muffler system from its design stage before it is actually built. Further, this work could also be used for sound quality studies of duct systems.

9:00

2aNSa3. Acoustic propagation in three-dimensional, rectangular ducts with flexible walls. Jane B. Lawrie (Dept. of Mathematics, Brunel Univ., Uxbridge, UB8 3PH, United Kingdom)

Analytic expressions provide valuable benchmarks against which to check sophisticated finite element codes. In this talk some such expressions are presented for two duct configurations. Consideration is given first to the propagation of sound in a three-dimensional, unlined duct formed by three rigid walls, lying along $y = 0$, $-b \leq z \leq b$ and $z = \pm b$, $0 \leq y \leq a$ of a Cartesian frame reference, and closed by a thin elastic plate lying along $y = a$, $-b \leq z \leq b$. On assuming harmonic time dependence $e^{-i\omega t}$, the fluid-coupled structural waves are expressed in the form $B_n \psi_n(y, z) e^{\pm i s_n x}$, $n=0, 1, 2, \dots$, where $\psi_n(y, z)$ are an infinite set of non-separable eigenfunctions, s_n are the admissible wavenumbers, and B_n are the wave amplitudes. An exact, closed form expression for the eigenfunctions is presented. Various properties of the eigensystem are discussed. Second, the effect of incorporating a porous lining into the above model is considered. It is shown how the analysis is extended to include this modification. Numerical results are presented for both situations.

9:20

2aNSa4. Closed form calculation of reverberant sound propagation in a thin duct with flexible walls. Michael Panza (Dept. of Mech. Eng., Gannon Univ., 109 University Square, Eire, PA, 16541, panza@gannon.edu)

An analytical method for calculating the propagation of reverberant sound between two parallel perfectly reflecting planes in a thin duct with flexible side walls is presented. The flexible walls are modeled with a complex wave number to account for dissipation within the wall material. A mathematical model based on the Euler–Maclaurin sum formula provides an approximate closed form Green's function for the acoustic space. The method gives a set of two coupled integral equations in acoustic pressure and side wall displacement consisting of convolutions with respect to the spatial dimension along the duct. Laplace transforms for the spatial dimension are applied to the integral equations to solve for the acoustic pressure in terms of the spatial Laplace transforms of the Green functions for the reverberant acoustic space and the mechanical wave in the side wall material. A closed form solution is obtained by considering the first term of a binomial series in the Laplace domain and applying a partial fraction expansion leading to the solution in the spatial dimension along the duct. Numerical simulations show the behavior of the reverberant noise reduction provided by the flexible side walls along the duct.

9:40

2aNSa5. Exploration of a hybrid noise control system in a cylindrical duct. Gee-Pinn Too (Dept. of Systems and Naval Mechatronic Eng., Natl. Cheng Kung Univ., Tainan, Taiwan 701) and Shao-Rong Chen (Natl. Kaohsiung Marine Univ., Taiwan)

The purpose of this study is to explore the effects of sound elimination in a cylindrical duct by combining a reactive muffler and active noise control (ANC) system. Besides the exploration via experiment of the combined noise control system, predictions of desired signals in ANC system are proposed for this hybrid system. These predictions are Grey prediction based on Grey theory and signal processing for path impulse response function. In the experiment, the effects of sound elimination (such as transmission loss and insertion loss) are compared between cases with ANC systems installed before the muffler and after the muffler. The results indicate several conclusions that (1) the sequence of arrangement of muffler can influence the results of active noise control, (2) the effect of noise reduction in ANC system is influenced extremely by reference signal received, (3) the hybrid system has the advantages over a traditional muffler when the muffler is not designed for the frequency of the noise, and (4) predictions of the desired signals such as Grey prediction or path impulse response function could give a better control for the hybrid system.

10:25

2aNSa6. Optimal microphone placement for active noise control in a forced-air cooling system. Raymond de Callafon and Charles Kinney (Dept. of Mech. and Aerosp. Eng., Univ. of California, San Diego, 9500 Gilman Dr., La Jolla, CA 92093-0411, callafon@ucsd.edu)

In active noise cancellation systems with relatively small acoustic coupling, feedforward compensation is an effective methodology to create a controlled emission for sound attenuation. Especially for small electronic systems where forced air-cooling is required to control the temperature of large power sensitive components in the system, an active noise cancellation (ANC) system is a viable solution to reduce acoustic emissions. In this paper we discuss the placement of the noise source microphone for feedforward based active noise control in a forced-air cooling system. Noise source microphone placement is directed by the ANC performance of an on-line output-error based affine optimization of a linearly parametrized generalized finite impulse response filter for sound compensation. For the computation of the optimal filter, generalized or orthogonal FIR models are used as they exhibit the same linear parametrization as a standard FIR filter. The procedure is demonstrated on a small portable NEC LT170 data projector. The data projector is equipped with a shielded internal directional pick-up microphone to measure the sound created by the forced-air cooling of the projector's light bulb. Non-invasive small directional speakers located at the inlet and outlet grill of the data projector are used to minimize acoustic coupling.

10:45

2aNSa7. Reflection and transmission across partial blockages in fluid-filled flexible, non-thin-walled pipes. Iain D. J. Dupere and Wenbo Duan (School of Mech., Aerosp. and Civil Eng., Univ. of Manchester, Manchester M60 1QD, United Kingdom)

A model is presented for propagation along a flexible pipe whose thickness is not small in comparison with its diameter across a partial blockage with varying sizes and material properties. Comparison is made with Flugge's well-known thin-shell theory for a propagation along a pipe where it is found that the additional computational complexity found in the current model becomes necessary when the thickness of the shell exceeds 10% of the pipe radius. Comparison is also made with experiment both for the propagation characteristics of the pipe and for the reflection from a partial blockage. Two reflection models are presented: a crude area change model with compensation for the mass of the blockage and a more accurate model using high-order modes and matching to a flexible blockage using co-location. Reasonable agreement is found for both with the more accurate model, giving better agreement but at the expense of computational efficiency. The work is useful both for blockage detection and for detecting stenosis in blood vessels.

11:05

2aNSa8. A cochlear analog bio-mimetic muffler. Sripriya Ramamoorthy (Dept. of Otolaryngol./Head and Neck Surgery, Oregon Res. Ctr., NRC 04, Oregon Health and Sci. Univ., 3181 SW Sam Jackson Park Rd., Portland, OR 97239, ramamoor@ohsu.edu)

A noise control device, the structural acoustic silencer, is designed by adopting the tailored structural acoustic filtering exhibited by the mammalian cochlea. In the cochlea, a flexible plate of gradually varying width and thickness separates the upper and lower fluid-filled ducts. Tailoring of the flexible plate properties enables frequency-specific localization of traveling waves, which are slowed down significantly at and near the site of resonance. While such slowing of the traveling wave leads to efficient energy coupling to the flexible plate thereby reducing transmitted acoustic fluctuations, the gradually varying impedance reduces reflections and allows the frequency-component to travel to its resonance site. The design of the bio-mimetic muffler employing non-biological materials is performed using three dimensional finite element analysis and validated against experimental data. The relation between coupled dispersion and transmission loss in the noise control device is explored. The coupled wave propagation in the engineered device is compared with the wave propagation in a passive cochlea.

11:25

2aNSa9. Shaped optimization of multi-chamber mufflers with open-ended perforated inlets using a simulated annealing method. Ying-Chun Chang (Dept. of Mech. Eng., Tatung Univ., Taipei, Republic of China, ycchang@ttu.edu.tw) and Min-Chie Chiu (Chungchou Inst. of Technol., Yuanlin, Changhua 51003, Taiwan, Republic of China)

Recently, research on new techniques of single-chamber mufflers equipped with a non-perforated intruding tube has been addressed; however, the research work on multi-chamber mufflers conjugated with open-ended perforated intruding inlet tubes which may dramatically increase the acoustical performance has been neglected. Therefore, the main purpose of this paper is not only to analyze the sound transmission loss (STL) of a multi-chamber open-ended perforated inlet-tube muffler but also to optimize the best design shape within a limited space. In this paper, the four-pole system matrix for evaluating the acoustic performance is derived by using a decoupled numerical method. Moreover, a simulated annealing method has been used during the optimization process. To appreciate the acoustical ability of the open-ended perforated intruding inlet-tube and chambers inside a muffler, two kinds of traditional multi-chamber mufflers hybridized with non-perforated intruding inlet tubes (one-chamber and two-chamber mufflers) have been assessed and compared. Results reveal that the maximal STL is precisely located at the desired tone. In addition, the acoustical performance of mufflers conjugated with perforated intruding inlet tubes is superior to traditional mufflers. Also, it has been shown that the acoustic performance for both pure tone and broadband noise will increase if the muffler has more chambers.

11:45

2aNSa10. Numerical and experimental investigation of sound transmission of a tee-junction in a rectangular duct at higher-order modes. Siu-Kit Lau (Architectural Eng. Program, 203C Peter Kiewit Inst., Univ. of Nebraska-Lincoln, 1110 S. 67th St., Omaha, NE 68182-0681) and Kwan-Hao Leung (The Hong Kong Polytechnic Univ., Hung Hom, Kowloon, Hong Kong)

Sound transmission and scattering properties in higher-order modes across the tee-junction of a rectangular duct used in ventilation and air-conditioning system were investigated numerically and experimentally. High-sound transmission of the fundamental mode and higher-order modes across the main duct is observed at eigen-frequencies of the main duct. The

resonance of branch modes is suppressed by the weak modal coupling of the branch-modes and the traveling wave in the main duct at or very close to the eigen-frequencies of the sidebranch, which results in high-sound transmission of the fundamental mode and higher-order modes across the main duct and excitation of the branch modes at higher frequencies. Increases in sound scattering into higher-order mode are found when the non-planar or longitudinal branch-mode excited. In the case of co-excitation of the longitudinal branch-mode and non-planar branch-modes, a broader band-stop action in sound transmission has been observed. The results of numerical simulations were verified by experiments. A formulation of a transmission matrix based on the transfer function and a two-microphone method was shown. [The work described in this paper was supported by a grant from the Hong Kong Polytechnic University (Project A/C Code: G-U362).]

TUESDAY MORNING, 20 APRIL 2010

LAUREL A/B, 8:15 A.M. TO 12:00 NOON

Session 2aNSb

Noise, Animal Bioacoustics, and INCE: Effect of Noise on Humans and Non-Human Animals I

Ann E. Bowles, Cochair

Hubbs Sea World Research Inst., 2595 Ingraham St., San Diego, CA 92109

Brigitte Schulte-Fortkamp, Cochair

Technical Univ. Berlin, Einsteinufer 25, Secr TA 7, Berlin, 10587, Germany

Chair's Introduction—8:15

Invited Papers

8:20

2aNSb1. Soundscape research in networking across countries: COST Action TD0804. Brigitte Schulte-Fortkamp (Inst. of Fluid Mech. and Eng. Acoust., Technische Universitt Berlin, 10587 Berlin, Germany) and Jian Kang (Univ. of Sheffield Western Bank, Sheffield S10 2TN, United Kingdom)

Soundscape, different from noise control engineering, is about relationships between human beings, sound environments, and society. Research in soundscape covers science, engineering, social science, humanity, and art, and it is related to many disciplines including among others acoustics, anthropology, architecture, ecology, landscape, noise control engineering, psychology, sociology, and technology. Soundscape research represents a paradigm shift in the field of environmental noise in that it combines physical, social, and psychological approaches and considers environmental sounds as a "resource" rather than a "waste." The COST Action has built a vibrant international network and delivers the foundations for soundscape indicators, an integrated database of experimental and field data, publications, and tools to support design and decision making. Therefore, the paper will discuss the underpinning science and practical guidance that supports the measurement tools and their implementation in "Soundscapes."

8:40

2aNSb2. A subject centered approach to environmental noise effects: revisiting old concepts and proposing new methods. Caroline Cance (INCAS3, Dr. Nassaulaan 9, P.O. Box 797, Assen, 9400 AT, The Netherlands, ccance@gmail.com) and Danièle Dubois (CNRS & Univ. Paris 6, Paris, 75015, France, danièle.dubois@upmc.fr)

Considering the effects of noise implies that "somebody" is affected by noise. If the noise itself is not problematic regarding its measurement (by physics), the question is less clear regarding WHO is concerned by noise? Humans, animals? Asking such a question entails some other ones: Is any living system similarly affected by noise? Generally as an organism? Or differentially as "subjects"? What scientific domains are concerned with such studies? We would address these questions from our experience and knowledge on soundscapes. Starting from psychophysics that remains within the "endemic antinomy of a world-less subject confronting a thought-less object: the antique dualism of mind and matter" [Shalins (1976)], we will consider the diversity of subjects (animal as well as human "experts" of different types, acousticians, urban planners, politicians, and inhabitants) to propose an alternative conception we name "semiophysics": it leads to reconsider the concepts of information versus meaning, as well as from a methodological point of view, the concepts of affordance versus Umwelt. Coupling field research and experimental work by accounting for meanings as relationships given to the world by the different subjects calls then for an ecological validity of laboratory investigations.

9:00

2aNSb3. Effect of combined noise sources on cognitive performance and perceived disturbance. Jin Yong Jeon and Pyoung Jik Lee (Dept. of Architectural Eng., Hanyang Univ., Seoul 133-791, Korea, jyjeon@hanyang.ac.kr)

An experiment was conducted to investigate the differences in cognitive performance and perceptions under road traffic noise and construction noise combined with road traffic noise. Under the conditions with individual noise and combined noise sources, an episodic memory task and semantic task were carried out in the laboratory. The subjects were asked to recall the presented words after the exposure to noises (episodic memory task) and to select the target word when the five words, including the target word and other words, were presented simultaneously (semantic memory task). Subjects also rated perceived annoyance and disturbance to the noise exposure during the experiment. The result showed that the percentage of correct answer significantly decreased with an increase in construction noise in an episodic memory task. In contrast, the semantic memory task was not impaired by the level of construction noise level. This indicates that only the retrieval task with a process of generating the internal cues was affected by the level of combined noise sources. And it was also found that the perceptions of combined noise sources were highly correlated with the result of the episodic memory task.

9:20

2aNSb4. Standardizing the measurement of natural and urban soundscapes. Robert Kull (Parsons, 7447 Central Business Park Dr., Ste. 100, Norfolk, VA 23513)

For the purpose of discussion, a simple model was presented to illustrate the contributions and interactions of various elements within an environment that make up soundscapes [Kull (2006)]. These elements should be considered when attempting to characterize a soundscape using acoustic measurements. For this presentation, the model will be used with the objective of creating a plan for taking sound measurements.

9:40

2aNSb5. Evaluating the prevalence of masking as a causal factor in wildlife responses to noise. Jesse Barber (Dept. of Fish, Wildlife and Conservation Biology, Colorado State Univ., 1474 Campus Delivery, Fort Collins, CO 80523, barber.jesse@gmail.com) and Kurt M. Fristrup (Natl. Park Service, Fort Collins, CO 80525)

Many protected natural areas are chronically exposed to noise. Noise exposure grows faster than the human populations whose activities generate noise. Data accumulate regarding masked hearing performance in animals, which can be coupled with models of sound propagation to predict reductions in the spatial extent of auditory awareness with elevated background sound levels. The emergence of predictive models of noise masking effects recommends a reassessment of field studies of wildlife responses to noise to identify the potential scope of this problem. A review of this literature reveals a substantial and diverse collection of scientific papers whose findings are plausibly related to masking effects and an increasing number of more decisive results from studies that were designed to control for other confounding factors.

10:00—10:10 Break

10:10

2aNSb6. Marine mammals and anthropogenic noise: Challenges from a management and regulatory prospective. Amy R. Scholik-Schlomer, Shane Guan, Jolie Harrison, and Craig Johnson (Natl. Marine Fisheries Service, Office of Protected Resources, 1315 East-West Hwy., Silver Spring, MD 20910)

Responsibilities of the National Marine Fisheries Service (NMFS) include conserving and recovering marine species protected under the U.S. Marine Mammal Protection and Endangered Species Acts. One of our primary objectives is to assess the risks anthropogenic noise in marine/coastal environments poses to animals in those environments and implement appropriate measures to reduce these risks. Many challenges to achieving these goals exist from both a scientific and a regulatory perspective. Accounting for the inherent complexity of source characteristics, noise propagation through the environment, and temporal/spatial overlap between sources and protected species, as well as understanding how noise exposure affects species are often quite difficult. Exposures typically are either high-level, short-term (e.g., seismic survey), or lower-level, long-term (e.g., construction project), with each presenting different risks. Establishing appropriate metrics for describing noise sources, assessing effects on individuals and on populations/stocks, as our statutes require, and ensuring the practicality of applying these metrics to real-world situations are essential. There are also often considerable data gaps, which require us to draw upon the knowledge gained from human and other terrestrial species. NMFS is currently re-evaluating and updating our acoustic criteria, which are used within our impact assessments, to reflect the best-available science on these issues.

10:30

2aNSb7. Predicting acoustic impact: Considering individuals versus populations. Brandon L. Southall (Southall Environ. Assoc., Inc., 911 Ctr. St., Ste. B, Santa Cruz, CA 95060), Adam S. Frankel, and William T. Ellison (Marine Acoust., Inc., Middletown, RI 02842, adam.frankel@marineacoustics.com)

Predictive models to assess the potential impacts of anthropogenic activities are increasing being used to assess environmental impacts. The most accurate models are, arguably, individual-based models that integrate species-specific biological and area-specific environmental data such as the acoustic impact model. These models simulate the movement of animals in four dimensions and typically include responses to both acoustic and/or non-acoustic environmental variables. The predicted exposures of individuals to sound sources may be assessed with a risk function; however, a recent meta-analysis of behavioral response and acoustic dose [Southall *et al.* (2007)] did not find meaningful linear relationships that were generally applicable between low to medium received sound levels and behavioral response. There appears to be significant variability in both species-specific and individual animal responses that are mediated by various contextual factors. Since regulatory requirements are often focused on population impact, the simplest approach to considering individual variability in behavioral response is that averaged over the population. Therefore, a mathematically linear dose-response function may be most applicable, at least within species for certain activity patterns, even if it is not supported for individual animals in all conditions.

10:50

2aNSb8. Harbor seals respond with aversion to 69-kHz pings: Implications for weighting procedures for marine mammal noise metrics. Ann E. Bowles, Stephanie K. Graves, Michael Shane (Hubbs-SeaWorld Res. Inst., 2595 Ingraham St., San Diego, CA 92109, abowles@hswri.org), and Samuel L. Denes (Penn State Univ., State College, PA 16802)

Author Shane tracked cultured white seabass (*Atractoscion nobilis*) instrumented with 69-kHz ultrasonic coded transmitters (UCTs) in the vicinity of harbor seals (*Phoca vitulina*), later finding the bones of fish associated with UCTs. This led the authors to suspect that seals had targeted and eaten instrumented fish. To determine whether seals could detect pings, four harbor seals and a ringed seal at SeaWorld San Diego were exposed to pings from two 69-kHz and one 83-kHz UCTs and their spontaneous responses observed. The seals were not expected to respond strongly because most of the energy in the pings was close to the upper limit of hearing, but three of the four harbor seals reacted with aversion to the 69-kHz pinger with the highest source level (147 dB *re* 1 μ Pa), swimming into a refuge pool or jumping out of the water. The received level at the closest point of approach was estimated at 20 dB above sensation level or less. The results suggest that seals may be especially responsive to high-frequency tonal signals, and that broadband weighting functions may not consistently yield efficient exposure metrics. [Funded by NOAA; in-kind support from SeaWorld San Diego.]

11:10

2aNSb9. Modeling the exposure of greater sage-grouse to noise from industrial gas drilling rigs. Stacie L. Hooper, Sean Hanser, and Gail Patricelli (Dept. Evolution and Ecology, Univ. of Calif., Davis, 1 Shields Ave., Davis, CA 95616, slhooper@ucdavis.edu)

Natural gas and methane extraction is a growing industry in Wyoming, and some greater sage-grouse leks appear to be declining in areas near industrial sites. The goal of this project is to develop a model for understanding whether industrial noise has played a significant role in these reductions in lek attendance. A software package called NMSIM, previously developed by Wylie Laboratories to measure noise exposure from aircraft for the National Park Service, is being used. NMSIM utilizes amplitude measurements, recorded a set distance from the noise source, topographic map data, and measurements of other factors affecting sound propagation such as temperature and humidity, to build a spatially explicit model simulating how noise from the industrial sites propagates over the surrounding terrain. Simulation results are then verified using a set of noise exposure measurements taken from known locations around the gas drilling rigs. In addition to explaining historic lek attendance patterns, this model will also be used to predict how noise from new industrial sites will impact nearby greater sage-grouse leks.

11:30—12:00 Panel Discussion

TUESDAY MORNING, 20 APRIL 2010

HERON, 7:55 A.M. TO 12:00 NOON

Session 2aNSc

Noise and Architectural Acoustics: Healthcare Acoustics/Noise and Occupant Perception and Performance

Erica Ryherd, Cochair

Georgia Institute of Technology, Dept. of Mechanical Engineering, 771 Ferst Dr., Atlanta, GA 30332-0405

Kenneth P. Roy, Cochair

Armstrong World Industries, 2500 Columbia Ave., Lancaster, PA 17604

Mandy Kachur, Cochair

Acoustics By Design, Inc., 303 Detroit St., Ste. 304, Ann Arbor, MI 48104

Chair's Introduction—7:55

Invited Papers

8:00

2aNSc1. Evidence-based medicine meets evidence-based design: An interdisciplinary study of acoustics and sleep disruption as a context for improving teamwork. Jo M. Solet (Div. of Sleep Medicine, Harvard Med. School, 15 Berkeley St., Cambridge, MA 02138, joanne_solet@hms.harvard.edu)

Hospital noise-induced sleep arousal probability curves were derived by exposing laboratory monitored sleeping human subjects to 14 common stimuli selected and scaled from a recorded inpatient hospital sound-scape. The research process in this recently published successful project, which informed the 2010 Guidelines for the Construction of Health Care Facilities, serves as a context for discussion of the promise and pitfalls of interdisciplinary teamwork, bringing together evidence-based medicine and evidence-based design, to solve real world problems. Revealing normally unspoken professional assumptions, work ethics, record keeping requirements, and conflict of interest concerns, this program seeks to make cross field boundary breaking collaboration easier and more successful and to improve possibilities for meaningful innovation. [Work supported by AAHF, FGI, and CHD.]

8:20

2aNSc2. Hospital soundscape modeling. Selen Okcu (College of Architecture, Georgia Inst. of Technol., 247 Fourth St., Atlanta, GA 30332, selen.okcu@gatech.edu), Erica Ryherd, and Craig Zimring (Georgia Inst. of Technol., Atlanta, GA 30332)

Effective soundscape solution considerations for hospital settings can be very complex. Some acoustic qualities of these soundscapes have been shown to have potential negative impacts on occupant outcomes. To enhance these qualities, different acoustic solutions are applied in the hospital settings. Testing the effectiveness of these implications is critical but not always practical in the real settings. In this study, we examined the soundspaces of critical care settings through acoustic models. This paper will discuss the preliminary results regarding the modeled acoustic qualities of various ICU settings such as noise levels and reverb qualities.

8:40

2aNSc3. An approach to making safe and secure indoor soundscape measurements. Kenneth Good and Kenneth Roy (Armstrong World Industries, 2500 Coulumbia Ave., Lancaster, PA 17601)

Outdoor soundscape measurements are being done for many types of noise sources and locations, and the monitor equipment must be both durable and secure from both the elements and physical disturbance. Indoor measurements require a somewhat more sophisticated setup since the measurement space is often an occupied space, and in the case of hospital corridors these will likely be very active spaces 24/7. A creative yet simple approach to setting up monitor stations in hospital hallways, nurses stations, and patient rooms was developed and is being used in a number of research programs. It is imperative that valid noise measurements be made in the designated spaces without arousing undue concern or interests on the part of the occupants (both medical professionals and patients/families), and that the equipment be unobtrusive and thus secure. Monitor system mounting technique and measurement performance will be illustrated and discussed.

9:00

2aNSc4. Description and comparison of healthcare sounds. Benjamin C. Davenny and Gladys L. Unger (Acentech Inc., 33 Moulton St., Cambridge, MA 02138, bdavenny@acentech.com)

Within the clinical literature and the popular press, the noise from particular pieces of healthcare equipment or certain clinical environments is described by giving an example that is supposed to orient the reader. But these examples are often very misleading and may be in inappropriate. Outdoor everyday sounds are compared with interior healthcare sounds and steady sounds are compared with impulsive sounds. This discussion will explore better descriptions of healthcare and everyday sounds with the goal of providing better comparisons. Illustrated by examples currently in the literature we will discuss what constitutes a useful comparison and what attributes of the sounds need to be considered. Better descriptions of healthcare noise sources will help in the control of hospital soundscapes.

9:20

2aNSc5. Impact of building design on inpatient nursing unit noise levels: A case study of evidence based design. Nathan Sevener (Acoust. By Design, Inc., 4001 Murvihill Rd., Ste. 2E, Valparaiso, IN 46383)

In spring 2006, the evidence-based design process was being applied to the design of the new inpatient pavilion at Lakeland Hospital. Because of survey feedback, the hospital was aware that noise detracted from patients satisfaction with the existing facility and therefore noise level became one of the 77 metrics used in the design of the new pavilion. To help create an improved acoustical environment, the sound levels in the existing facility were monitored, the design of the new pavilion was reviewed with respect to noise control, and follow-up measurements were made after occupation. The results of that effort are presented here.

9:40

2aNSc6. Controlling patient room reverberation with a thin acoustical wall treatment. Francis J. Babineau and Amy Sparks (SoundTech Inc., 3880 SoundTech Ct., Grand Rapids, MI 49512)

A study was conducted to explore an alternate method for reducing room reverberation time in a healthcare setting. Reverberation time (RT) is known to play an important role in overall noise levels and speech privacy, both of which are key factors in patient and staff satisfaction. However, RT is difficult to control in healthcare facilities, primarily due to durability and infection control requirements. A private patient room was treated with a thin, acoustically absorptive wall treatment. The wall treatment was not a typical acoustical finish, with absorption coefficients ranging from 0.05 to 0.6. However, several walls of the room were entirely covered with the treatment to achieve reverberation control. Reverberation times were measured before and after the installation, and results were consistent with predicted values. Despite being a small study, the results suggest that a thin acoustical wall treatment that meets durability and infection control requirements can be effective for controlling reverberation in healthcare facilities and creating a more comfortable environment for patients and staff.

10:00—10:20 Break

10:20

2aNSc7. Further studies in hospital noise control at the Johns Hopkins Hospital: Part 1. Colin Barnhill, James E. West (Dept. ECE, Johns Hopkins Univ., Baltimore, MD 21218, cb@jhu.edu), Timothy Hsu, and Erica E. Ryherd (Georgia Inst. of Technol., Atlanta, GA 30332-0445)

Hospital noise levels remain well above World Health Organization guidelines. One of the main difficulties in treating hospital noise is the inability to use conventional sound absorption techniques. In intensive care units, the hospital does not allow any materials that can harbor bacteria or produce a high level of dust. This criterion prohibits conventional drop ceilings and conventional sound absorbing panels and carpet. Based on our previous panel design [M. MacLeod *et al.*, "Weinberg 5C: A case study in hospital noise control, J. Acoust. Soc. Am. 119, 3327 (2006)] a lower cost version of our hospital noise panels was implemented by replacing the wrapping with DuPont™ Tyvek®. The lowered cost makes implementation of these new panels more feasible for hospitals. In this talk, it will be shown that this new implementation produces comparable results to the original panel design. The performance of a modified DL2 measurement will be investigated as a new way to characterize hospital noise.

10:40

2aNSc8. Further studies of hospital noise control at the Johns Hopkins Hospital: Part 2. Timothy Y. Hsu, Erica E. Ryherd (Woodruff School of Mech. Eng., Georgia Inst. of Technol., Atlanta, GA 30332-0445), James E. West, Colin L. Barnhill (Johns Hopkins Univ., Baltimore, MD 21218), Marie Swisher (Sidney Kimmel Comprehensive Care Ctr., Baltimore, MD 21231), and Natalia Levit (DuPont, Richmond, VA 23234)

It has been shown through previous research that hospital noise levels continue to rise and that there exist potential health and occupational hazards due to the soundscape. Previous studies have examined parameters such as overall average sound levels, statistical distributions of levels, and frequency content in a variety of wards. However, studying the impact of sound absorbing materials in medical facilities is still very challenging due to strict infectious control requirements. We will present a case study on installing various sound absorbing treatments in four hematological cancer wards. Four units in the Weinberg Building of the Johns Hopkins Hospital were selected due to their identical geometries, similar staff activities, and similar patient acuity levels. Researchers installed different acoustical treatments in each ward, from untreated to fully treated with prototype materials. The prototype materials tested consisted of a layer of DuPont™ Tyvek® covering a panel of fiberglass acoustic absorbing material. In each ward, extensive acoustical measurements were taken and a questionnaire survey was administered to the registered nurses on staff. Preliminary results will be presented that compare and contrast attributes such as background noise, reverberation, speech intelligibility, and subjective perception in the four different wards.

11:00

2aNSc9. Acoustic privacy and health care. Neil Moiseev (Shen Milsom & Wilke LLC, 417 Fifth Ave., New York, NY 10016)

Acoustic privacy can be differentiated into two categories: freedom from intrusive noise, such as a person snoring or wheezing in the next bed, traffic outside the windows, carts in the hallways, and footsteps on the floor above; and speech privacy—the freedom from being overheard and of overhearing others. Providing the proper acoustical environment and the protecting privacy must be a joint effort between the facility designers and hospital staff. A brief discussion of the basic requirements for speech privacy and HIPAA privacy and a quality background sound will be presented.

11:20

2aNSc10. Two healthcare buildings get the power, the noise, and the vibration from a gas turbine generator system. Chad N. Himmel (1705 W. Koenig Ln., Austin, TX 78756)

A package gas turbine generator was installed at a small, 4.3-MW cogeneration power plant of a mixed use development with office, retail, hospital, residential, and other occupancies. Shortly after commissioning, noise complaints were received from nearby medical offices about undesirable tonal noise intrusions. At another nearby hospital building, a neurosurgery suite addition would include a new intra-operative magnetic resonance imaging (iMRI) unit. Ambient vibration measurements at that site indicated prominent discrete frequencies of disturbance that could affect iMRI image quality. Investigations found loud broadband noise and dominant ground borne vibration in the vicinity of the turbine, with strong tonal peaks relating to turbine and generator rotational rates. The intrusive noise in nearby medical offices and the structure borne vibration at the hospital included matching strong tonal peaks. Various noise- and vibration-mitigating measures were proposed and some were implemented with successful results for the medical offices and new iMRI suite. Mitigating measures are discussed, along with measurement results and photographs.

11:40

2aNSc11. Strengthening the healthcare guidelines: About the new online research community. David M. Sykes (Remington Partners, 23 Buckingham St., Cambridge, MA 02138, dsykes@healthcareacoustics.org), William Cavanaugh, Gregory Tocci, and Andrew Carballeira (Cavanaugh Tocci Assoc., Sudbury, MA)

Myriad challenges and opportunities exist for researchers in the 2010 edition of the Guidelines for the Design and Construction of Healthcare Facilities. The new edition, released in January, contains the first comprehensive acoustical criteria ever included in the 60-year-old Guidelines that are now being adopted as code by most states, federal agencies, and municipalities. Enforcement—and future strengthening—of these Guidelines will require a strong, organized, well-funded research community in acoustics willing to do transdisciplinary research. Since federal agencies do not actively support research on the human impacts of noise in healthcare facilities but some private foundations do, the drafters of the Guidelines recently launched an online research community to enable research teams across the country to interact directly with interested foundations, government agencies, policy groups, and healthcare organizations. The online community has a distinguished Advisory Board of interested professionals in acoustical science, medicine, healthcare architecture, and engineering. It hosts an open database of recent research on acoustics and human health, RFPs and announcements, links to interested groups such as IC BEN, the World Health Organization and UIA-PHG, and other features such as blogs and wikis.

2a TUE. AM

Session 2aPA

Physical Acoustics and Engineering Acoustics: Ultrasonics, Nonlinear Acoustics, Acousto-Optics, and Engineering Acoustics in Honor of Mack Breazeale I

Lev A. Ostrovsky, Cochair
Zel Tech/NOAA ESRL, 325 Broadway, Boulder, CO 80305-3328

Nico F. Declercq, Cochair
Georgia Tech. Lorraine, 2 rue Marconi, Metz, 57070, France

Chair's Introduction—7:55

Invited Papers

8:00

2aPA1. Mack Breazeale and E. A. Hiedemann's group at Michigan State University. David T. Blackstock (Appl. Res. Labs., Univ. of Texas at Austin, P.O. Box 8029, Austin, TX 78713-8029 and Dept. M.E., UT Austin, Austin, TX 78712-0292)

Mack Breazeale received a Ph.D. degree in physics at Michigan State University in 1957. He was one of 19 doctoral students supervised by Egon A. Hiedemann, who headed a very active group in ultrasonics at Michigan State in the 1950s and 1960s. Mack's period there, 1954–1961, is reviewed and also the history of the Hiedemann group, which was well known for its contributions to physical acoustics. A particular interest in optical measurement of ultrasonic waves led to many theses and papers in nonlinear acoustics, at that time just beginning to enjoy a renaissance. Among those who, like Mack, went on to have very active roles in the Acoustical Society of America were Laszlo Adler, Bill Cook, Logan Hargrove, and Walter Mayer.

8:20

2aPA2. Impact of Mack Breazeale on the ultrasonic studies of interfaces. Laszlo Adler (Adler Consultants Inc./Ohio State Univ., 1560 Gulf Blvd., #1002 Clearwater, FL 33767)

Over the last 50 years, Mack Breazeale made very significant contributions to Physical Acoustics. He was a great influence to his students as well as to researchers all around the world who carried on, continued, and expanded the work which was initiated by him. His work is well known on nonlinear studies of liquids and solids, acousto-optics, ultrasonic interaction at liquid-solid interfaces at the Rayleigh angle generating leaky waves, and many others. Nonlinear acoustic techniques improved material characterization of complex materials and structures significantly. The generation and detection of leaky waves opened up several applications in nondestructive evaluation. In this presentation, several of these techniques which were recently developed will be discussed. One of this is the optical detection of leaky waves at an air-solid interface which can be used to evaluate elastic properties of solids as well as surface imperfections. A nonlinear ultrasonic method to study interfaces between solid layers will be also presented. Mack Breazeale leaves a legacy of knowledge, wisdom, and kindness and will always be remembered by us.

8:40

2aPA3. Acousto-optics as the background of a long-lasting friendship with Mack Breazeale. Oswald Leroy (Interdisciplinary Res. Ctr., Katholieke Universiteit Leuven Campus Kortrijk, Meensesteenweg 453, 8501 Kortrijk, Belgium)

Ever since I have known Mack Breazeale, our friendship has come along with discussions and bilateral suggestions for further research in acousto-optics. Among many other topics we had a common interest in the application of acousto-optics for nondestructive testing. For layered structures all information concerning the structure is contained in the phase information of the ultrasound investigating the material. This phase information is obtained using two or only one laser beam depending on the kind of required information. Using the technique of two laser beams the measured phase difference between the incident and the reflected ultrasonic beam is influenced by the geometrical shape of a plate, the thickness of a thin layer on a substrate, as well as by the presence of defects inside a layered material. In addition different methods are discussed to describe the diffraction of light by monofrequent, pulsed, and adjacent superposed ultrasound. In the optical nearfield of a diffracted laser beam, it is even possible to reconstruct all of the acoustic wave parameters. The ability of measuring the amplitude and phase of reflected and transmitted waves is a powerful tool to discover information about the surface on which scattering of sound occurs.

9:00

2aPA4. Substructural organization and acoustic harmonic generation in fatigued metals. John H. Cantrell (NASA Langley Res. Ctr., MS 231, Hampton, VA 23681, john.h.cantrell@nasa.gov)

Since the discovery of acoustic harmonic generation in solids by Breazeale and Thompson [Appl. Phys. Lett. **3**, 77 (1963)] and Gedroits and Krasilnikov [Sov. Phys. JETP **15**, 122 (1963)], nonlinear acoustics has gained popularity as a tool for nondestructive materials evaluation. A longstanding problem has been the assessment of fatigue damage. Harmonic generation measurements are

shown to provide for the first time a quantitative, unambiguous means to assess the state of fatigue in metals from the virgin state to fracture. The salient features of an analytical model are presented that account for the microelastic-plastic nonlinearities resulting from the interactions of an acoustic wave with self-organized dislocation substructures and cracks that evolve during cyclic loading. The model predicts a monotonic increase in the nonlinearity parameter of several hundred percent over the life of the material. Generally, the increase in the nonlinearity parameter during the first 80%–95% of fatigue life is dominated by the evolution of organized dislocation structures, while the last 5%–20% is dominated by crack growth. Applications of the model to aluminum alloy 2024-T4, 410Cb stainless steel, and IN100 nickel-base superalloy yield excellent agreement between theory and experiment.

9:20

2aPA5. A diffraction correction for the nonlinearity parameter measured by the harmonic generation technique. William T. Yost (NASA-Langley Res. Ctr., 3B E. Taylor St, Rm. 285, MS231, Hampton, VA 23681-2199, william.t.yost@nasa.gov)

Practical applications of harmonic generation to determine the nonlinearity parameter Beta generally tend toward the lower drive (fundamental) frequencies. As a result, diffraction effects on Beta become more significant. Derivations of a number of different diffraction correction formulas, which are applied for a piston source and a receiver located some distance away, are in the literature. In the paper by Blackburn and Breazeale [J. Acoust. Soc. Am. **76**, 1755–1760 (1984)] the correction formulas were applied to the amplitude of the fundamental frequency and the results on Beta were experimentally tested. By building on this earlier work, a diffraction correction formula that includes a factor from a numerical integration algorithm applied to the harmonically generated wave is provided. The formulation of this algorithm is discussed. Experimental results are given for measurements of Beta in AA 2024 to illustrate the agreement.

9:40—10:00 Break

10:00

2aPA6. Nonlinear acoustics: A transition from laboratory to a practical fatigue assessment tool. Jeong-Kwan Na (SID, Univ. of Dayton Res. Inst., 300 College Park, Dayton, OH 45469, jeong-kwan.na@wpafb.af.mil)

The ultrasonic second harmonic generation technique measuring the nonlinearity parameter has been developed and used in laboratories for several decades throughout the world. Many students and visiting scholars from numerous countries devoted their passions at the physical acoustics laboratories led by Dr. Mack Breazeale for 50 years. During that period of time, nonlinear elastic properties of single crystals, polycrystalline alloys, superconductors, ceramics, and composites were measured. New measurement systems and techniques were also developed to understand temperature dependent linear and nonlinear elastic behaviors of these materials over a temperature range from the liquid helium to Curie temperatures of piezoelectric ceramics. It took years of research and development efforts with a persistent funding before a practical fatigue damage measurement technique was finally developed. The transition from a rack full of equipment to a portable fatigue damage measurement system, a continuous iteration process was inevitable. The current system, specifically designed for steam turbine blades fatigue inspections, consists of a probe fixture and an industrial grade lunch-box computer with custom designed signal processing cards.

10:20

2aPA7. Nonlinear interaction of ultrasonic waves: A tribute to Mack Breazeale. Murray S. Korman (Dept. Phys., U.S. Naval Acad., Annapolis, MD 21402)

While at Brown University (~1977), this graduate student was fortunate enough to learn about Mack Breazeale's important early work in nonlinear ultrasonic waves in solids [J. Appl. Phys. **36**, 3486 (1965)] through his thesis advisor (R. T. Beyer), who had friendly and most collegial interactions with Mack, through the ASA. When I met Mack Breazeale at ASA Meetings, he would discuss how he made some of his measurements and helped me understand some complexities involving the scattering of sound by sound. While Mack Breazeale was Associate Editor of nonlinear acoustics he gave a wealth of good advice (over a period of several years) on how to improve a manuscript on crossed beam scattering in turbulence and divide the material into at least two parts—which was done. When the paper was out, one agreed with Mack that it was much better and worth the effort. Years later, doing summer research at NCPA, Mack helped improve a manuscript on nonlinear acoustic landmine detection. As a tribute to Mack Breazeale, his work on “Quantum mechanical theory on nonlinear interaction of ultrasonic waves” (with I. L. Bajak) [J. Acoust. Soc. Am. **68**, 1245 (1980)] will be discussed in some detail.

Contributed Papers

10:40

2aPA8. Peculiarities of acoustic beam reflection from a fluid-solid interface. Oleg A. Sapozhnikov (Appl. Phys. Lab., Univ. of Washington, and Dept. Phys., Moscow State Univ., Russia, olegs@apl.washington.edu), Alexander A. Karabutov, Jr., and Vladimir G. Mozhaev (Moscow State Univ., Russia)

Mack Breazeale and his colleagues published several papers on experimental study of non-specular reflection of ultrasonic beam from a fluid-solid interface [Breazeale *et al.*, J. Acoust. Soc. Am. **56**, 866 (1974); J. Appl. Phys. **48**, 530 (1977)]. To study fairly unusual details of the reflected beam structure, Schlieren visualization was employed, which clearly confirmed theoretical predictions. The present talk was motivated by this elegant ap-

proach of Breazeale. One of the goals was to improve reflected beam imaging by using pulsed Schlieren technique. The second goal was to observe a growing interface wave. It is known that the secular equation for acoustic waves at fluid-solid interfaces yields the common leaky wave and its complement. This complementary wave grows instead of decays with propagation and is time-reversed compared to the leaky wave. Ultrasonic pulses and their reflections were visualized using Schlieren imaging and stroboscopic flashing of a semiconductor laser. The source was a broadband (0.5–3 MHz) single element plane transducer. Reflectors were aluminum blocks with fine angular adjustments in an optically transparent water tank. The wave reflection and transmission were also studied numerically using finite differences. [Work was supported by RFBR, NIH, and NSBRI grants.]

10:55

2aPA9. Fatigue damage evaluation using harmonic generation in polycrystalline copper. Brian Reinhardt and Bernhard Tittmann (Dept. of Eng. Sci. and Mech., Penn State Univ., 212 Earth Eng. Sci. Bldg., University Park, PA 16802)

A fundamental goal of ultrasonic nondestructive evaluation is to characterize material defects before failure. During material fatigue, dislocations tend to nucleate and become sources of stress concentrations. Eventually, cracks start to form and lead to material failure. Recent research has indicated that nonlinear harmonic generation can be used to distinguish between materials of high- and low-dislocation densities. This research reports nonlinear harmonic generation measurements to distinguish between those areas of high- and low-dislocation densities in copper bars. The copper bars were subjected to flexural fatigue. Periodic scans were taken in order to track dislocation development during the fatigue life of the material. We show that this technique provides improved early detection for critical components of failure.

11:10

2aPA10. Simulation of nonlinear acoustic waves based on the semi-Lagrangian method. Glauber T. Silva and Andre Nachbin (Inst. of Phys., Univ. Fed. Alagoas, Maceio, Alagoas 57072-970, Brazil, glauber@pq.cnpq.br)

In this work, we introduce the semi-Lagrangian numerical method for one-dimensional nonlinear wave propagation in acoustics. The method is suitable to simulate interacting waves in fluid media. In the method, the acoustic fields are expressed in terms of Riemann invariants. These variables are computed along the characteristic curves one step back in time. The unknown characteristics are obtained by the Runge-Kutta method. The Riemann invariants are evaluated in the grid points using cubic interpolation. The method is validated comparing the numerical solution to the Earnshaw

solution for simple waves. Results show that the proposed method excels the Courant-Frederich-Lewy condition at least fourfold. The total numerical error is 12% and 3% using 16 and 32 points per wavelength, respectively. The proposed method is applied to simulate the interaction between a finite-amplitude tone with a Gaussian noise. Solutions are in qualitative agreement with earlier studies on tone and noise interaction [D. A. Webster and D. T. Blackstock, *J. Acoust. Soc. Am.* **64**, 687-693 (1978)]. [Work partially supported by CNPq.]

11:25

2aPA11. Ultrasonics in solids with frequency-dependent nonlinearity parameters. Paul A. Elmore (Naval Res. Lab., Marine Geosciences Div. Stennis Space Ctr., MS 39529, paul.elmore@nrlssc.navy.mil)

This paper reviews the current state of modeling the nonlinear generation of second-harmonic ultrasonic waves in heterogeneous solids. Experiments measure the rate in which the higher-order harmonics grow with wavenumber, k , and propagation distance, a . The lowest-order nonlinearity parameter, quantifying the growth rate of the second-harmonic amplitude, A_2 , experimentally is $\beta \propto \lim_{A_1 \rightarrow 0} (A_1/A_2 k^2 a)$, where A_1 is the fundamental harmonic amplitude. In crystals and amalgamated metals, β is constant, and the corresponding equation of motion can be written with one nonlinear term to a good approximation. In heterogeneous materials with granular contacts, however, such as lead-zirconate-titanate (PZT) and graphite-epoxy composites, β changes with wavenumber, and the corresponding equation of motion requires more nonlinear terms. An equation of motion updated from the one for crystals does provide predictions of β that appear to match the experimental results; however, it is inadequate for providing velocity predictions in graphite-epoxy composites, where there is significant dispersion, and does not explicitly account for the hysteric behavior of grain contacts. A recent model that accounts for grain contacts asserts that it can account for the frequency dependent β . This paper will review these models and compare predictions for β with PZT and composites data.

TUESDAY MORNING, 20 APRIL 2010

ESSEX A/B/C, 9:00 A.M. TO 12:00 NOON

Session 2aPP

Psychological and Physiological Acoustics: Application of Psychoacoustics to the Impaired Listener, Challenging Environments, Hearing Aids, and Cochlear Implants (Poster Session)

Sandra Gordon-Salant, Chair

Univ. of Maryland, Dept. of Hearing and Speech Science, College Park, MD 20742

Contributed Papers

All posters will be on display from 9:00 a.m. to 12:00 noon. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 9:00 a.m. to 10:30 a.m. and contributors of even-numbered papers will be at their posters from 10:30 a.m. to 12:00 noon.

2aPP1. Comparison of central masking versus monotic masking in non-simultaneous masking conditions. Mahnaz Ahmadi and Lawrence L. Feth (Dept. of Speech and Hearing Sci., The Ohio State Univ., 110 Pressey Hall, 1070 Carmack Rd., Columbus, OH 43210)

The growth of non-simultaneous masking was compared in a central masking versus a monotic masking condition for the same masker and signal parameters. Detection thresholds of normal listeners were measured utilizing a 2AFC up-down adaptive-tracking procedure. Signal-to-masker time intervals were 0, 2, 5, 10, 20, and 50-ms. Brief sinusoidal signals at 0.5, 1, 2, and 4 kHz were masked by a two-octave band of noise of 200 ms duration. The masker was centered on and off the frequency of the signal. Detection thresholds were greater for signal-to-masker intervals less than 50 ms in forward masking using both central and monotic masking conditions. Backward central masking did not result in significant threshold shifts. High-

frequency signals produced greater central masking effects; however, signals at lower frequencies were more effectively masked in monotic conditions. In contrast with monotic masking conditions, off-frequency maskers did not produce a clear central masking effect. Results of this study are consistent with neurophysiologic findings of medial olivocochlear bundle response characteristics and support the idea that central masking is mediated by efferent fibers in humans.

2aPP2. Comodulation masking release with signal-masker interactions represented in the envelope. Robert H. Pierzycki and Bernhard U. Seeber (MRC Inst. of Hearing Res., Univ. Park, Nottingham NG7 2RD, United Kingdom, rp@ihr.mrc.ac.uk)

In normal hearing, the threshold of a tone masked by a modulated narrow-band on-frequency masker (OFB) can be reduced if correlated modulation is present on spectrally distant flanking bands (FBs), an effect

known as comodulation masking release (CMR). Since electric hearing with current cochlear implants is based on envelope information, comodulation of envelopes on multiple electrodes might also be beneficial for detection in electric hearing. CMR was investigated with normal-hearing participants listening to unprocessed or vocoded stimuli. In Experiment 1, tone thresholds were determined when masked by a sinusoidally amplitude-modulated band of noise (SAM and OFB) and by zero to four FBs of noise whose envelopes were either co- or anti-modulated with the OFB envelope. In Experiment 2, envelopes of those signals were extracted in a vocoder and used to modulate noise or sinusoidal carriers, thereby replacing the original temporal fine structure (TFS). Significant CMR of 3–10 dB was found in unprocessed conditions and although reduced to 2–6 dB, CMR was still significant after vocoding. CMR did not differ significantly between the sine and the noise vocoder, suggesting that the applied SAM determined the magnitude of CMR. Since CMR withstands vocoding, comodulation is hoped to improve detection in electric hearing.

2aPP3. The speech-critical band for vowels in steady and fluctuating backgrounds. Eric W. Healy (Dept. of Speech and Hearing Sci., The Ohio State Univ., 1070 Carmack Rd., Columbus, OH 43210, healy.66@osu.edu), Kimberlee A. Crass (Univ. of South Carolina, Columbia, SC 29208), and Frederic Apoux (The Ohio State Univ., Columbus, OH 43210)

Previous work investigating the frequency resolution employed to process English vowels indicated that the speech-critical band (S-CB) for vowels was greater than the psychophysical critical band. This result suggests that the density of information in the acoustic signal is below the resolving power of the auditory system, and therefore functional resolution is limited by the signal. In the current investigation, the S-CB for vowels was measured in steady and fluctuating backgrounds. Thirty-six normal-hearing listeners heard vowels in /hVd/ context. The stimuli were restricted in overall bandwidth to 1.5 octaves centered at 1500 Hz and presented using vocoder-like processing as 1, 3, 5, 7, 9, or 11 low-noise noise carrier bands. Recognition increased and reached asymptote as the number of constituent bands increased. The carrier bandwidth at performance asymptote was again found to be larger than the psychophysical critical band. Moreover, the resolution of the acoustic signal at performance asymptote was similar across conditions in which the stimuli were presented (i) in quiet, (ii) in steady noise, or (iii) in four-talker babble at various signal-to-noise ratios. These results suggest that the current measure of speech frequency resolution is robust across a number of adverse listening conditions. [Work supported by NIDCD.]

2aPP4. Release from speech-on-speech masking under degraded signal conditions. Virginia Best (School of Med. Sci., Univ. of Sydney, Sydney, New South Wales 2006, Australia and Hearing Res. Cntr., Boston Univ., Boston, MA 02215, ginbest@physiol.usyd.edu.au), Nicole Marrone (Northwestern Univ., Evanston, IL 60208 and Boston Univ., Boston, MA 02215), Christine Mason, and Gerald Kidd, Jr. (Boston Univ., Boston, MA 02215)

Previously, Marrone *et al.* [J. Acoust. Soc. Am. (2008)] compared spatial release from masking (SRM) for a three-talker mixture of similar sentences in normal-hearing (NH) listeners and listeners with sensorineural hearing loss (HL). The HL group showed significantly less SRM. In an earlier study [Marrone *et al.*, J. Acoust. Soc. Am. (2008)] the NH group showed less SRM when the masker sentences were time-reversed while in the collocated case a large “reversed-masker release” (RMR) was also found. To investigate these findings, some listeners from the HL group were tested in the reversed-masker conditions. The difference in SRM between listener groups was much smaller for reversed speech. However, the HL group also had much less RMR. Both the overall target-to-masker ratios (TMRs) at which comparisons are made and the spectral smearing associated with sensorineural hearing loss were likely factors, perhaps affecting amounts of energetic and informational masking. We explored these factors in a new group of NH listeners in a similar task with both forward and reversed speech systematically degraded. Performance was measured at seven TMRs for unprocessed

as well as 32-, 16- or 8-channel vocoder speech. The results provide qualified support for an influence of TMR and spectral degradation on both SRM and RMR.

2aPP5. Auditory stream segregation using amplitude modulated vocoder bandpass noise. Yingjiu Nie and Peggy Nelson (Dept. of Speech-Lang.-Hearing Sci., Univ. of Minnesota, Minneapolis, MN 55455, peggynelson@umn.edu)

We investigated the contribution of amplitude modulation (AM) rate and spectral separation to stream segregation of vocoder bandpass noises. Stimulus sequences were repeated pairs of A and B bursts, where bursts were white noise or vocoder bandpass noise carrying sinusoidal AM (100% modulation depth). Bursts differed either in the center frequency of the noise, or the AM rate, or both. Eight vocoder bands were used. The lowest four bands (1-2-3-4) were combined into one bandpass noise (B bursts) and the higher three bands (3-4-5, 4-5-6, and 6-7-8) were combined to constitute the A bursts. Results show that stream segregation ability increases with greater spectral separation. Larger AM rate separations were associated with stronger segregation abilities, but not when A and B bursts were both white noise. Significant inter-subject differences were noted. Results suggest that, while both spectral and AM rate separation separations could be cues for auditory stream segregation, stream segregation based on AM rate is more successful when combined with spectral separation. Correlations between segregation ability and understanding of vocoded speech will be discussed.

2aPP6. A test battery to assess localization ability in simulated complex acoustic environments. Stefan Kerber and Bernhard U. Seeber (MRC Inst. of Hearing Res., Nottingham NG7 2RD, United Kingdom, s.kerber@ihr.mrc.ac.uk)

In complex acoustic environments, following a target sound is made difficult by the presence of noise and reverberation and especially hearing impaired people struggle understanding speech in such settings. Localizing the speaker is important in such situations to follow discussions and to gain additional benefit from visual cues. A test battery is proposed to quantify localization performance in different realistic environments and with laboratory tests when noise and reverberation are present. The aim is to relate real-life performance to that in simplified tests. In the localization tests, participants indicate the sound location with a visual pointer. Sounds are played in an anechoic chamber from loudspeakers across the frontal hemi-field. Localization performance is measured in quiet, in diffuse background noise, and with reverberation of simulated rooms. The ability to cope with a single reflection is assessed in a precedence effect paradigm where participants localize a sound and its delayed copy for various delays and levels. The test battery is completed by questionnaires (e.g., SSQ), a speech, and a cognitive test. By cross-comparing results from the tests, we attempt to predict the performance in real-world environments from outcomes of simplified tests particularly in the presence of hearing impairment.

2aPP7. On the possible influence of spectral- and temporal-envelope cues in tests of sensitivity to temporal fine structure. Christophe Micheyl (Dept. of Psych., Univ. of Minnesota, 75 East River Rd., Minneapolis, MN 55455-0344, cmicheyl@umn.edu), Huanping Dai (Univ. of Arizona, Tucson, AZ 85721), and Andrew J. Oxenham (Univ. of Minnesota, Minneapolis, MN 55455-0344)

The role of temporal fine structure (TFS) in speech and pitch perception has attracted considerable attention. Recent studies have suggested sensitivity to TFS even at high frequencies (8 kHz), well beyond the known limits of phase-locking in mammals, and a reduced ability to use TFS in hearing-impaired listeners. These conclusions were based on tasks involving the discrimination of complex tones or iterated-rippled noise, which had different power spectra but were (a) bandpass-filtered in a way that reduced tonotopic excitation-pattern (EP) cues and (b) had random component phases to eliminate temporal-envelope (TE) cues. In this study, we examine the possibility that residual EP and/or TE cues may have been available to the listeners in these experiments. Analytical and computational analyses indicated that although systematic TE differences were absent at the level of the stimuli, they were likely present at the output of the cochlea. Empirical studies tested whether the available TE and EP information might have influenced

performance. Preliminary results suggest that although listeners may not have been sensitive to the available TE cues, performance may have been influenced by EP cues in a way that could also explain the deficits shown by hearing-impaired listeners. [Work supported by NIH R01 DC05216.]

2aPP8. Discrimination of repetitive intervals by younger and older listeners. Peter Fitzgibbons (Dept. Hearing, Speech, and Lang Sci., Gallaudet Univ., 800 Florida Ave., NE, Washington, DC 20002) and Sandra Gordon-Salant (Univ. of Maryland, College Park, MD 20742)

The study measured listener sensitivity to increments in the inter-onset intervals (IOIs) separating successive 20-ms 4000-Hz tone bursts in isochronous sequences. Stimulus sequences contained from 2–6 tone bursts, separated equally by IOIs in the range of 25–100 ms across stimulus conditions. Difference limens (DLs) for increments of all tonal IOIs were measured to assess listener sensitivity to changes in sequence rate. A DL was also measured for increments of a single IOI located at a fixed position within 6-tone sequences. Listeners included younger and older normal-hearing adults and older adults with high-frequency hearing loss. The results revealed that the relative DLs for sequence rate decreased as the magnitude of the reference IOI and the number of sequence components increased. The relative DL for a single interval embedded within a six-tone sequence was smaller than corresponding DLs measured with two-tone sequences, but only for brief reference IOIs. The discrimination performance of the older listeners was poorer than that of the younger listeners, especially for two-tone sequences with the shortest reference IOIs. The findings are interpreted within the context of multiple-look mechanisms and possible age-related differences in the sensory coding of signal onsets. [Research supported by the National Institute on Aging, NIH.]

2aPP9. Age-related differences in auditory spatial attention depend on task switching complexity. Gurjit Singh (Dept. of Psych., Univ. of Toronto, 3359 Mississauga Rd. N, Mississauga, Ontario L5L 1C6, gurjit@psych.utoronto.ca), M. Kathy Pichora-Fuller, and Bruce A. Schneider (Univ. of Toronto, Mississauga, ON, L5L 1C6, Canada)

We investigated the role of simple and complex switching of auditory attention in a multi-talker, multi-spatial listening situation with target location uncertainty. In all conditions, a target sentence from an edited version of the CRM corpus was presented from one spatial location and competing sentences from two different locations, with cues specifying the target's call-sign identity and the probability of its location. Four probability specifications indicated the likelihood of the target being presented at the left, center, and right locations (0-100-0, 10-80-10, 20-60-20, and 33-33-33). In conditions requiring simple switches of attention, the task was to report key words from the target sentence. In conditions requiring complex attention switching, when target call-signs were presented from one of the unlikely locations, the listener's task was to report key words presented from the other unlikely location. A total of eight younger and eight older adults who had normal audiometric thresholds below 4 kHz participated. The key finding is that, whereas both age groups performed similarly in conditions requiring simple switches of attention, older performed worse than younger listeners in conditions requiring complex switching. Switching complexity may explain, in part, why older adults with relatively good audiograms report difficulty communicating in complex listening situations.

2aPP10. Psychophysical tuning curves and recognition of highpass and lowpass filtered speech for a person with an inverted V-shaped audiogram. Vinay Nagaraj (Dept. of Electrons and Telecommunications, Acoust. Res. Ctr., Norwegian Univ. of Sci. and Technol. (NTNU), NO-7491 Trondheim, Norway) and Brian Moore (Univ. of Cambridge, Cambridge CB2 3EB, United Kingdom)

A single subject whose audiogram resembled an inverted V shape (good hearing at 4000 Hz, poorer hearing at other frequencies) was tested. Results of the TEN(HL) test suggested that a dead region (DR) in the cochlea was present at all test frequencies from 500 to 3000 Hz, but no DR was present at 4000 Hz. Psychophysical tuning curves obtained using signal frequencies of 2000, 3000, 4000, and 6000 Hz showed upward shifted tips for the lowest two signal frequencies and a downward shifted tip for the highest frequency. The results suggested a functioning region extending from 3900 to 5100 Hz,

with DRs outside that range. The identification of nonsense syllables, amplified according to the Cambridge formula, was measured as a function of lowpass or highpass filter cutoff frequency. The results suggested that useful speech information could only be extracted from a limited frequency range around 4000 Hz.

2aPP11. The relationship between quiet threshold and the forward-masking temporal effect. Elizabeth A. Strickland (estrick@purdue.edu) and Yonit A. Shames (Dept. SLHS, Purdue Univ., 500 Oval Dr., West Lafayette, IN 47907)

Research in our laboratory has shown that the temporal effect (TE) in simultaneous masking is consistent with a decrease in gain, possibly mediated by the medial olivocochlear reflex (MOCR). The TE in simultaneous masking is a decrease in threshold signal-to-masker ratio for a signal at masker onset when a precursor is added. This work has been extended to a forward-masking paradigm. A growth of masking (GOM) function is measured with a short-duration off-frequency masker (which should not activate the MOCR) and a 4-kHz signal. Then the masker level is fixed at a point on the lower leg of the GOM function, and threshold is measured with a long-duration precursor which is intended to activate the MOCR. The estimated input-output function is compared for a precursor at and well below the signal frequency. The difference in thresholds is the TE, which is also a measure of the change in gain. In simultaneous masking, the size of the TE decreases with increasing quiet threshold. In the present study, this relationship was examined for the forward-masking TE. The TE was measured as a function of precursor level for listeners who had a range of quiet thresholds, including listeners with mild cochlear hearing impairment. [Research supported by a grant from NIH(NIDCD) R01 DC008327.]

2aPP12. Interaural time difference thresholds as a function of the duration of the beginning of a sound in hearing-impaired adults. Michael A. Akeroyd and Fiona H. Guy (MRC Inst. of Hearing Res., Glasgow Royal Infirmary, 16 Alexandra Parade, Glasgow, United Kingdom, maa@ihr.gla.ac.uk)

The precedence effect indicates that the auditory system places most weight on spatial information at the onset of a sound. In this experiment, we measured how well hearing-impaired adults can ignore what comes immediately after the onset of a sound, as any limitations in that should interfere with localization in environments with lots of reflections. The stimulus was designed to be a simplified analog of a room impulse response: it consisted of two bursts of speech-shaped noise concatenated together: the first was short and given some ITD (representing an ideal onset that marked direction correctly), but the second was far longer and was interaurally uncorrelated (representing later reflections and reverberation from all possible directions). We measured psychometric functions (over the ITD and duration of the first burst) to determine the minimum duration of the first burst needed to report its direction. Presently, 19 listeners have completed the task. No significant correlation was found between minimum first-burst duration and hearing loss (better-ear loss = 3–50 dB), but the individual variation was considerable, especially so in those listeners with larger hearing losses. The results indicate that some listeners will have difficulty in ignoring irrelevant information after a sound's onset.

2aPP13. The effects of development and hearing impairment on the ability to understand speech in temporally and spectrally modulated noise. Joseph W. Hall, III, Emily Buss, and John H. Grose (Dept. Otolaryngol., Univ. UNC Chapel Hill, CB 7070, Chapel Hill, NC 27599, jwh@med.unc.edu)

This study examines the ability of hearing-impaired children to receive speech recognition benefit from temporal modulation, spectral modulation, or combined spectral and temporal modulation of a background noise. The task involves identification of words within the context of meaningful sentences presented in a speech-shaped noise background. Control groups include normal-hearing adults and children and hearing-impaired adults. The age range of the children is approximately 5–10 years. In the procedure, the masker level is held constant and the speech level is adaptively varied to track a criterion percent correct. Preliminary data suggest that both hearing-impaired children and hearing-impaired adults show a poorer than normal

ability to benefit from temporal and spectral modulation. The hearing-impaired children require approximately 2-dB higher signal-to-noise ratio than the hearing-impaired adults for most conditions. The normal-hearing adults and children show better masked thresholds than their hearing-impaired counterparts, with normal-hearing adults generally performing 3–5 dB better than normal-hearing children across conditions.

2aPP14. Performance of phonemically targeted processing in conjunction with compression processing with spectral enhancement.

Jeffrey J. DiGiovanni (Auditory Psychophysics and Signal Processing Lab., School of Hearing, Speech and Lang. Sci., Ohio Univ., Athens, OH 45701, digiovan@ohio.edu), Janet C. Rutledge (Univ. of Maryland, Baltimore City, Baltimore, MD 21250), and Chessy S. Umble (Ohio Univ., Athens, OH 45701)

Sensorineural hearing loss is strongly linked to poorer speech intelligibility, especially in noise. The goal of the present study is to test, individually and in combination, two signal processing strategies designed to improve both consonant and vowel perception. In the first strategy, specific consonants were targeted for processing to increase amplitude and duration. For consonants with a duration increase, the adjacent vowel was decreased proportionately in order to maintain overall word and sentence duration. Second, Col-SE, an adaptive compression processing strategy incorporating spectral enhancement, was used to process stimuli. Hearing-in-noise-test sentences were presented monaurally to normal-hearing and hearing-impaired adults through an insert earphone in the presence of speech-shaped noise. Preliminary results show that normal-hearing listeners benefited from Col-SE processing in conjunction with minimal phonemically targeted speech processing more than with either processing strategy individually. Increases in consonant amplitude and duration beyond a modest amount reduce intelligibility. These data suggest that the two processing strategies are viable to improve speech intelligibility, but that there is a limit to the processing whereby benefits are no longer observed.

2aPP15. Effects of independent bilateral compressive amplification on lateralization of a single source. Ian M. Wiggins and Bernhard U. Seeber (MRC Inst. of Hearing Res., Nottingham NG7 2RD, United Kingdom, ian@ihr.mrc.ac.uk)

Use of compressive amplification in bilateral hearing aid fittings can disrupt binaural cues important to spatial hearing. The head-related transfer function introduces direction-dependent interaural time and level differences (ITDs and ILDs) which are consistent with one another. Independent bilateral compression, however, reduces ILDs such that they suggest a different, conflicting direction than ITDs. The reduction in ILDs is a dynamic effect that depends on the characteristics of the compression and the sound. Single-channel compression was applied over a wide, high-frequency band. Two conditions were run, with the high-frequency channel presented to listeners in isolation or recombined with an unprocessed low-frequency channel. Stimuli included pink noise bursts with varied onset slopes and rates of amplitude modulation and speech. The effects of compression on the perceived auditory objects were assessed using a semantic differential method, in which listeners rated various spatial attributes on scales between two bipolar adjectives, for example, image width was rated on a scale between “focused” and “diffuse.” Additionally, a lateralization task was performed. Initial results show that for sounds with slow onsets, compression tends to shift image location or causes the image to split. Interestingly, image location can also be affected for speech, particularly if high-pass filtered.

2aPP16. Horizontal localization and hearing in noise ability in adults with sensorineural hearing loss using hearing aids with binaural processing. Amy R. Mullin (Dept. of Commun. Sci. and Disord., The Univ. of Texas at Austin, 1 University Station A1100, Austin, TX 78712, amyruthie@mail.utexas.edu)

The purpose of the study was to determine whether hearing aids with binaural processing improve performance during a localization and a hearing in noise task. The study included 15 participants between the ages of 29 and 68 who had a bilateral symmetrical sensorineural hearing loss and who had no prior hearing aid experience. Participants were fitted with Oticon

Epoq XW receiver-in-the-ear hearing aids bilaterally. The participants completed a horizontal localization task and a hearing in noise task with three listening conditions: (1) without hearing aids (NO), (2) with hearing aids that were not linked (BIL), and (3) with hearing aids that were linked (BIN). For the horizontal localization task, 1.5-s pink noise bursts were used as the stimulus. Sentences from the Hearing in Noise Test were used as target stimuli for the hearing in noise task. Continuous discourse by one male and two female talkers was recorded and used as maskers. The specific aim of the localization and hearing in noise tasks was to determine which of the listening conditions resulted in the best score for each task. Data are still being collected and data analysis will follow.

2aPP17. Effect of waveform shape and polarity on loudness and on place pitch for cochlear implant users. Robert Carlyon, Olivier Macherey, and John Deeks (MRC Cognition and Brain Sci. Unit, 15 Chaucer Rd., Cambridge CB2 7EF, United Kingdom)

It has been shown that, for pseudomonophasic pulses presented in monopolar mode, less current is needed to achieve the same loudness when the short-high phase is anodic than when it is cathodic. Experiment 1 extended that finding, obtained with the Advanced Bionics (AB) device, to waveforms that can be implemented in other devices, using 99-pps, 32- μ s/phase pulse trains. For AB and MedEl devices, stimuli were triphasic pulses whose central phase had twice the amplitude of the first and third phases. For the Nucleus device, stimuli were pairs of biphasic pulses with opposite leading polarity, each with a 58- μ s interphase-gap, separated by 8 μ s—resulting in two adjacent same-polarity phases in the center of the waveform. The current level needed was always 1–2 dB lower when the central portion was anodic than when it was cathodic—including for electrode 1 of the MedEl device, which is inserted deep into the cochlea. Experiment 2 showed that, for pseudomonophasic pulses in bipolar mode, (a) pitch is lower when the “short-high” pulse is anodic relative to the more-apical than to the more-basal electrode and (b) intermediate pitches can be produced by variations in waveform shape and polarity applied to the same bipolar pair.

2aPP18. Pitch-ranking of electric and acoustic stimuli by cochlear implant users with the HiRes and Fidelity120 speech processing strategies. Benjamin A. Russell and Gail S. Donaldson (Dept. of Comm. Sci. and Disord., Univ. of South Florida, Tampa, FL 33620, barussel@mail.usf.edu)

Estimates of place-pitch sensitivity in cochlear implant (CI) users are typically obtained using electric pulse trains presented directly to the implanted electrodes. Such estimates may overestimate the place-pitch sensitivity available to these listeners through their speech processors due to spectral smearing by the analysis filters. To determine the influence of speech processing on place-pitch sensitivity, electric pitch-ranking (EPR) and acoustic pitch-ranking (APR) thresholds were compared in four users of the Advanced Bionics CI. EPR thresholds were obtained for single- and dual-electrode pulse-train stimuli presented to electrodes near the center of the array. APR thresholds were obtained for pure tones having frequencies corresponding to the tonotopic locations of the electric pulse trains, using both the HiRes and Fidelity120 speech processing strategies. Counter to expectation, APR thresholds were similar to EPR thresholds for three of four subjects, and APR thresholds were smaller than EPR thresholds for the remaining subject. APR thresholds did not differ systematically for the HiRes and Fidelity120 strategies, consistent with Nogueira *et al.* [(in press). EUR-ASIP J Adv Signal Proc]. Findings suggest that CI users can make use of across-channel cues when performing an APR task and that such cues can compensate for spectral smearing by the analysis filters.

2aPP19. Lowering mean fundamental frequency to improve speech intelligibility in noise under simulated electric-acoustic stimulation. Christopher A. Brown and Sid P. Bacon (Dept. of Speech and Hearing Sci., P.O. Box 870102, Tempe, AZ 85287-0102)

We have previously demonstrated that much or all of the benefits of electric-acoustic stimulation (EAS) can be achieved when the low-frequency speech is replaced with a tone that is modulated in both frequency and amplitude with F0 and amplitude envelope cues derived from the target speech. One advantage of this approach is that the frequency of the carrier tone can

be lowered with little decline in benefit. Lowering mean F0 has the potential to provide EAS benefit to CI users who have very limited residual hearing. One drawback to this approach is that it relies on the efficacy of the pitch extraction algorithm. This is problematic because pitch extractors have trouble in background noise, an environment in which F0 is particularly useful. Here, an alternative way of lowering mean F0 that is unaffected by the presence of noise is examined under simulated EAS conditions. Speech intelligibility was measured using an algorithm based on resampling and compared to performance with the pitch-based method we have used previously, at frequency shifts of 0, 0.5, and 1 octave. At the 0.5-octave shift, the resampling-based approach provided more benefit than the tone. However, at the 1-octave shift, resampling was less beneficial. [Work supported by NIDCD.]

2aPP20. Effect of auditory deprivation on binaural sensitivity in bilateral cochlear implant users. Ruth Y. Litovsky, Gary L. Jones, and Richard VanHoesel (Waisman Ctr., Univ. of Wisconsin, 1500 Highland Ave., Madison, WI 53705, litovsky@waisman.wisc.edu)

Increasing numbers of cochlear implant (CI) users receive implants in both ears in an effort to restore spatial hearing abilities. A known limitation of commercial CI devices is lack of synchrony to the signal fine timing cues arriving at the two ears. In addition to these hardware limitations, which may contribute in substantial ways to the gap in performance typically seen between bilateral CI users and normal-hearing people, there are considerable inter-subject differences in the type and history of hearing loss. We use a research processor which enables inter-aurally coordinated pulsatile stimulation of selected pairs of electrodes in the right and left ears. This project is concerned with the effects of the age at onset of deafness, and place of stimulation, on binaural sensitivity. Thresholds for discrimination of inter-aural time difference (ITD) and inter-aural level difference (ILD) and pointer-identification for perceived intracranial position were measured. Pre-lingually deafened subjects had no sensitivity to ITD but retained ILD sensitivity. People with childhood- or adult-onset of deafness retained sensitivity to both ITD and ILD, some within normal-hearing level of performance on some conditions. The role of auditory deprivation in the emergence and preservation of binaural sensitivity will be discussed. [Work supported by NIH-NIDCD.]

TUESDAY MORNING, 20 APRIL 2010

HARBORSIDE B, 8:20 TO 10:15 A.M.

Session 2aSAa

Structural Acoustics and Vibration, Noise, Underwater Acoustics, Animal Bioacoustics, and INCE: Noise Control of Small Marine Vehicles

Joseph M. Cuschieri, Chair
Lockheed Martin Corp., 100 East 17th St., Riviera Beach, FL 33404

Chair's Introduction—8:20

Invited Papers

8:25

2aSAa1. An overview of unmanned underwater vehicle noise in the low to mid frequency bands. Jason D. Holmes (Raytheon BBN Technologies, 10 Moulton St., Cambridge, MA 02138, jholmes@bbn.com), William M. Carey (Boston Univ., Boston, MA 02215), and James F. Lynch (Woods Hole Oceanograph. Inst., Woods Hole, MA 02543)

Unmanned (autonomous) underwater vehicles offer a unique, cost-effective platform for performing ocean acoustic measurements and surveys because multiple systems can be deployed from a single research vessel. Various data surveys can be performed including on-the-bottom geo-acoustic surveys over large areas, sub-sea-surface turbulence and microbubble structure surveys, and bi-static fish population surveys. To take advantage of the autonomous survey capabilities of underwater vehicles, sufficient signal-to-noise ratio and acoustic aperture (resolution) are required for acoustic measurements. The most commonly used vehicle sonar systems provide images utilizing high frequency hull mounted arrays and sources. In the lower-frequency band (100 Hz–10 kHz), however, vehicle noise levels and aperture remain the two most significant challenges, especially for passive systems. Previous experimental and analytical work has shown that a towed array with synthetic aperture processing can be used to obtain the necessary aperture. The major challenge of vehicle radiated noise is the focus of this paper, and both measured and archival results on vehicle noise are presented including an overview of levels, spectral character, and noise mechanisms for several vehicles. In particular, fundamental vehicle propulsion system noise is discussed along with implications on measurement performance and possible mitigation strategies.

8:45

2aSAa2. Unmanned underwater vehicle self-noise and implications for low-frequency sonar design. Brian Houston (Naval Res. Lab., Code 7130, 4555 Overlook Ave., Washington, DC 20375)

Sonars configured on small unmanned underwater vehicles tend to be higher-frequency systems that are relatively insensitive to the dominating self-noise associated with low-frequency energy sources. These include the main propulsor, control surface actuators, and navigation components. A new generation of vehicles is required having reduced low-frequency signatures in order to support the growing interest in low-frequency broadband active and passive sonars. In the work presented here, we discuss the impact of vehicle noise

on the performance of both real and synthetic receiver arrays. Self-noise mitigation methods and their impact on vehicle architecture will be discussed as well as the use of signal processing techniques employing acoustic holographic projection to reduce the impact of structural-borne noise. [Work supported by ONR.]

9:05

2aSAa3. Decreasing the radiated acoustic and vibration noise of a mid-size, prop-driven, autonomous underwater vehicle. Richard Zimmerman, Gerald D'Spain, and John Orcutt (Scripps Inst. of Oceanogr., Univ. California San Diego, 291 Rosecrans St., San Diego, CA 92106, rzimmerman@ucsd.edu)

Previously published efforts at decreasing the radiated acoustic and vibration noise of the propulsion system of an Odyssey IIB autonomous underwater vehicle (AUV) manufactured by Bluefin Robotics, Inc. resulted in noise levels recorded by an AUV-mounted hydrophone array that were at or below typical background ocean noise levels across much of the frequency band from 300 Hz to 10 kHz [IEEE J. Ocean. Eng. **30**, 179–187]. The modifications required to achieve this 20–50-dB reduction in propulsion noise levels will be reviewed in this talk. Recently, these modifications have been incorporated into the Bluefin 21 AUV at the Scripps Institution of Oceanography. In addition, the stepper motors in the linear actuators used to steer this AUV's vectored-thrust tail cone in depth and heading have been replaced. At-sea measurements show that the high-level, broadband transients that previously occurred every 2–3 s due to these actuators are no longer visible in the hydrophone data. Eliminating these sources of self-noise allow the vehicle to be used in marine biological studies without vehicle noise disturbing either the acoustic measurements themselves or the habitats under observation. [Work supported by the Office of Naval Research and British Petroleum.]

9:25

2aSAa4. Acoustic noise estimates for a quiet unmanned underwater vehicle. Carl A. Cascio (50 Myrock Ave., Waterford, CT 06385-3008)

Acoustical Technologies Inc. (ATI) has been tasked by a number of clients to predict the radiated noise of unmanned underwater vehicles (UUVs). Laboratory, dockside, and at sea measurements have been taken on 21-in.-diameter, 20-ft-long prototype vehicles. Data were obtained during operation of the propulsion motors, control surfaces, and other UUV components. Structural impact hammer testing was also used to estimate propagation path transfer functions. Acoustic noise and structural vibration sensors were recorded on the ATI multi-channel data acquisition system and analyzed to produce narrow and broad bandwidth spectral estimates over the 10-Hz–100-KHz frequency range. Radiated noise predictions from the component source level data and transfer functions were made and illustrate the importance of selecting quiet components and appropriate noise control. This paper takes a look at how quiet a 21-in.-diameter 20-ft-long UUV could be using what is currently known about typical UUV component hardware. This is not an estimate for a real UUV but a radiated noise model based on an ATI concept using quiet components, practical noise control, and generic transfer functions. Radiated noise estimates will be presented at several operating speeds.

Contributed Papers

9:45

2aSAa5. Turbulent boundary layers over hydrophone arrays. Craig N. Dolder (Dept. of Mech. Eng., The Univ. of Texas at Austin, 1 University Station C2200, Austin, TX 78712-0292, dolder@mail.utexas.edu), Michael R. Haberman, and Charles E. Tinney (The Univ. of Texas at Austin, Austin, TX 78712-0235)

The speed at which naval vessels can operate sonar receiver arrays is limited due to the noise produced by the formation of a turbulent boundary layer (TBL) over the hull of the vessel. Despite efforts in the signal processing community to reduce these signatures, flow noise levels continue to surpass the signal to noise ratio needed for effective sonar operation at high-vessel speeds. This study focuses on the hydrodynamic signatures induced by the turbulence and the means by which energy can be removed from the TBL structures to reduce pressure fluctuations on the array elements. The research presented here employs an array of hydrophones with high-spatial resolution to measure the dynamic surface pressure while simultaneously acquiring time resolved single-point velocity field measurements at various positions above the array and within the TBL using laser Doppler velocimetry. Classical statistical quantities are then computed to determine the relationship between the pressure and velocity fields. Further, an analysis of the wave number frequency makeup of the surface pressure provides insight into the convective nature of the pressure filtered flow structures. The findings provide helpful guidance in developing active control methods for reducing the TBL noise.

10:00

2aSAa6. Estimation and measurement of the acoustic signature of unmanned surface and underwater vehicles. Joseph Cuschieri (Lockheed Martin MS2, Undersea Systems, Riviera Beach, FL 33404)

As the use of unmanned surface and underwater vehicles (USVs and UUVs) increases, the acoustic signature and self-noise of these vehicles become important for certain type of applications. For low-detection probability the UUV or USV must have a low-acoustic signature. Furthermore, the self-noise of the UUV or USV may interfere with on-board sonar sensors, especially as more sophisticated sonar systems are being developed. As the concept of the USV or UUV is to design for low cost and low weight, with low availability of power, it is important to address the mitigation of the acoustic signature during the design phase through estimation techniques based on modeling and component level selection. Additionally during the test and integration phase, methods to measure the acoustic signature without resorting to expensive offshore measurements in "quiet" acoustic test ranges are required. In this presentation some of the approaches available for modeling and testing are discussed, together with the challenges faced by noise control engineers working these type problems and the type of noise sources that have to be addressed.

Session 2aSAb**Structural Acoustics and Vibration and INCE: Space Vehicle Vibroacoustics**

Dean E. Capone, Cochair

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

Stephen C. Conlon, Cochair

*Pennsylvania State Univ., Applied Research Lab., P.O. Box 30, State College, PA 16804****Invited Papers*****10:30****2aSAb1. Vibroacoustics in airplane design.** Evan B. Davis (The Boeing Co., P.O. Box 3707, MC 67-ML Seattle, WA 98124, evan.b.davis@boeing.com)

Structural and structural-acoustic (vibroacoustic) tools are used to predict the future performance of systems in order to guide design trades and system optimization decisions. The key questions to be answered are (1) what needs to be known? (2) how well does it need to be known? and (3) how reliable are the tools that will be used to make the decisions?

10:50**2aSAb2. Vibro-acoustic analysis of aerospace structures and issues with the available commercial prediction tools.** Ali R. Kolaini and Dennis L. Kern (Jet Propulsion Lab., California Inst. of Technol., 4800 Oak Grove Dr., Pasadena, CA 91109-8099)

The results of vibro-acoustic modeling using the boundary element method (BEM) that predicts the acceleration responses at critical locations and at the interfaces of selected test articles are discussed. High fidelity acoustic tests were performed in a couple of cases and the results are used to validate the BEM predictions. The accuracy of the BEM and its ability to correctly predict the acceleration responses of lightweight structures are discussed in some detail. Also a combined system level BEM, consisting of structures very responsive to acoustic pressures, and force-limited base shake random vibration analysis was performed. We will discuss how these results may be used to derive random vibration specifications for the purpose of qualifying large and lightweight structures for flight. In this paper, we also discuss the commercially available vibro-acoustic tools that are used to predict the acoustic transmission losses and vibration responses of flight structures for lift-off (assumed to be diffuse) and transonic (turbulent boundary layer) acoustic fields. The pros and cons of using the statistical energy analysis, finite element analysis, BEM, and newly developed hybrid methods within these vibro-acoustic tools are discussed in some detail.

Contributed Paper**11:10****2aSAb3. Noise control in Space Shuttle Orbiter.** Jerry R. Goodman (SF/Habitability & Environmental Factors Div., NASA-JSC, Houston, TX 77058)

Acoustic limits in habitable space enclosures are required to ensure crew safety, comfort, and habitability. Noise control is implemented to ensure compliance with the acoustic requirements. The purpose of this paper is to describe problems with establishing acoustic requirements and noise control efforts, and to present examples of noise control treatments and design

applications used in the Space Shuttle Orbiter. Included is the need to implement the design discipline of acoustics early in the design process and noise control throughout a program to ensure that limits are met. The use of dedicated personnel to provide expertise and oversight of acoustic requirements and noise control implementation has shown to be of value in the Space Shuttle Orbiter program. It is concluded that to achieve acceptable and safe noise levels in the crew habitable space, early resolution of acoustic requirements and implementation of effective noise control efforts are needed. Management support of established acoustic requirements and noise control efforts is essential.

Session 2aSC

Speech Communication: Speech and Noise

Carol Y. Espy-Wilson, Chair

Univ. of Maryland, Electrical and Computer Engineering, A. V. Williams Bldg., College Park, MD 20742

Chair's Introduction—8:00

Invited Papers

8:05

2aSC1. Noise-suppression algorithms for improved speech intelligibility by normal-hearing and cochlear implant listeners. Philipos Loizou (Dept. of Elec. Eng., Univ. of Texas-Dallas, Richardson, TX 75080, loizou@utdallas.edu)

Much research in the past few decades focused on the development of noise reduction algorithms that can suppress background noise. While these single-microphone based algorithms have been proven to improve the subjective speech quality, they have not been effective in improving speech intelligibility. This is partly due to the fact that most noise-suppression algorithms introduce speech distortion and partly because most algorithms are not optimized to operate in a particular noisy environment. Furthermore, none of the existing noise-reduction algorithms was designed to optimize a metric that correlates highly with intelligibility. This talk will present intelligibility data collected with normal-hearing and cochlear implant listeners who were presented with noisy speech processed by environment-optimized algorithms. It will also present algorithms that were designed using metrics that correlate highly with speech intelligibility. The data from these studies suggest that it is possible to develop noise reduction algorithms that improve speech intelligibility provided some constraints are imposed on the design of the suppression function and/or the intended listening environment. Research supported by NIDCD/NIH.]

8:25

2aSC2. Subjective evaluation of the speech quality from speech enhancement and segregation algorithms. Vijay Mahadevan, Srikanth Vishnubhotla, and Carol Espy-Wilson (Univ. of Maryland, 3180, A V W Bldg., College Park, MD 20770)

Automatic separation of speech from noise and segregation of overlapping co-channel speech are two of the most challenging problems in speech processing. In previous work, we have developed algorithms to both enhance noisy speech and segregate overlapping speech streams. Our single-channel speech enhancement and speech segregation algorithms have shown better performance than other reported algorithms for automatic speech recognition. Additionally, objective evaluation scores of perceptual quality have shown a significant improvement following processing by our algorithms. In this study, we focus on subjective evaluation of these algorithms for human listeners. We investigate the intelligibility and quality of speech from our algorithm on normal-hearing listeners, cochlear implant users, and hearing impaired subjects. Our preliminary results indicate a significant improvement in the perceptual quality of the speech signal after being processed by our algorithm and suggest that the proposed algorithms can be used as a pre-processing block within the signal processing in hearing aid devices.

8:45

2aSC3. Speech recognition in loud noise using an ear-insert microphone. Tarun Pruthi, Mihai Despa, and Amit Juneja (Think A Move, Ltd., 23715 Mercantile Rd. Ste. 100, Beachwood, OH 44122)

Think A Move, Ltd. has developed a patented ear-insert microphone which captures speech as acoustic vibrations inside the ear canal. These vibrations propagate to the ear canal through the flesh and bones in the human skull. A high density foam on the earpiece seals the ear canal when the earpiece is inserted. Tests show that this earpiece provides an average passive noise cancellation (PNC) of around 38 dB for noises in Aurora database. Using an in-house speech command recognizer, with a short enrollment phase, on a database of 19 speakers (11 females, 8 males), speaking a vocabulary of 56 commands, an average accuracy of 85% has been observed in 90 dBA of tank, military vehicle, and machine gun noises. To further demonstrate the noise robustness of the earpiece as compared to external microphones, pilot tests were conducted on a small set of speakers to recognize speech commands recorded simultaneously with an external microphone and our ear-insert microphone with our recognizer. Results show that while the accuracy of the recognizer drops to 27% in 90 dBA noise from 96% in quiet for external microphone, it only drops to 92% in 90 dBA of noise from 95% in quiet for internal microphone.

9:05

2aSC4. Voice conversion for enhancing various types of body-conducted speech detected with non-audible murmur microphone. Tomoki Toda (Graduate School of Information Sci., Nara Inst. of Sci. and Technol., Takayama-cho 8916-5, Ikoma-shi, Nara 630-0192 Japan, tomoki@is.naist.jp)

Our proposed statistical voice conversion approach to enhancing various types of body-conducted speech detected with Non-Audible Murmur (NAM) microphone is presented in this talk. NAM microphone, one of the body-conductive microphones [Nakajima *et al.*, IEICE Trans. Inf. and Syst., **E89-D**, 1–8 (2006)], enables us to detect various types of body-conducted speech such as extremely

soft whisper, normal speech, and so on. Moreover, it is robust against external noise due to its noise-proof structure. To make speech communication more universal by effectively using these properties of NAM microphone, body-conducted speech enhancement techniques have been developed with a state-of-the-art statistical voice conversion algorithm [Toda *et al.*, IEEE Trans. ASLP, **15**, 2222–2235 (2007)]. The proposed techniques would bring a new paradigm to human-to-human speech communication: e.g., the use of body-conducted voiced speech for noise robust speech communication, the use of body-conducted unvoiced speech for silent speech communication, and the use of body-conducted artificial speech for speaking aid [Toda *et al.*, Proc. ICASSP (2009) pp. 3601–3604]. This talk gives an overview of these promising techniques and presents their applications. [This research was supported in part by MIC SCOPE.]

9:25

2aSC5. Synthesizing speech from surface electromyography and acoustic Doppler sonar. Arthur R. Toth (Yap Inc., 2414 Shady Ave., Pittsburgh, PA 15217, atoth@cs.cmu.edu), Michael Wand (Univ. Karlsruhe, Karlsruhe, Baden-Württemberg 76131, Germany), Szu-Chen Stan Jou (ATC, ICL, Industrial Technol. Res. Inst., Chutung, Hsinchu, Taiwan 31040), Tanja Schultz (Univ. Karlsruhe, Karlsruhe, Baden-Württemberg 76131, Germany), Bhiksha Raj (Carnegie Mellon Univ., Pittsburgh, PA 15213), Kaustubh Kalgaonkar (Georgia Inst. of Technol., Atlanta, GA 30332), and Tony Ezzat (Mitsubishi Electric Res. Labs., Cambridge, MA 02139)

Numerous techniques have been devised to process speech audio in noise, but automatic speech recognition is difficult when the noise is too great. An alternative approach is to collect data that represent the speech production process but is less affected by noise in the speech audio range. Two such types of data come from surface electromyography (EMG) and acoustic Doppler sonar (ADS). EMG records muscle activation potentials. ADS records reflected ultrasound tones. Both can be used to measure facial movements related to speech, but they present their own challenges for automatic speech recognition. This work investigates the alternative approach of using these data sources for speech synthesis. The synthesis techniques explored in this work are based on Gaussian mixture model mapping techniques, which are commonly used for voice transformation. Voice transformation is traditionally concerned with changing the identity of speech audio signals, but others have demonstrated that such techniques can be used to transform different types of signals, such as non-audible murmur and electromagnetic articulography, to speech. This work demonstrates that such techniques also show promise for transforming EMG and ADS signals to speech.

9:45

2aSC6. Dealing with noise in automatic speech recognition. Douglas O’Shaughnessy (INRS-EMT, Univ. of Quebec, 800 de la Gauchetiere West, Ste. 6900, Montreal QC H5A 1K6, Canada)

While automatic speech recognition (ASR) can work very well for clean speech, recognition accuracy often degrades significantly when the speech signal is subject to corruption, as occurs in many communication channels. This paper will survey recent methods for handling various distortions in practical ASR. The problem is often presented as an issue of mismatch between the models that are created during prior training phases and unforeseen environmental acoustic conditions that occur during the normal test phase. As one can never anticipate all possible future conditions, ASR analysis must be able to adapt to a wide variety of distortions. Human listeners furnish a useful standard of comparison for ASR in that humans are much more flexible in handling unexpected acoustic distortions than current ASR is. Methods that adapt ASR features and models will be compared against ASR methods that enhance the noisy input speech. Other topics to be discussed will include estimation of noise and channel parameters, RASTA, and cepstral mean normalization. TRAP-TANDEM features Vector Taylor Series, joint speech and noise modeling, and advanced front-end feature extraction. Single-microphone versus multi-microphone approaches will also be discussed.

10:05—10:25 Break

10:25

2aSC7. Using speech models for separation in monaural and binaural contexts. Daniel P. Ellis, Ron J. Weiss, and Michael I. Mandel (Dept. of Elec. Eng., Columbia Univ., 500 W. 120th St., Rm. 1300, New York, NY 10027, dpwe@ee.columbia.edu)

When the number of sources exceeds the number of microphones, acoustic source separation is an underconstrained problem that must rely on additional constraints for solution. In a single-channel environment the expected behavior of the source—i.e., an acoustic model—is the only feasible basis for separation. We have developed an approach to monaural speech separation based on fitting parametric “eigenvoice” speaker-adapted models to both voices in a mixture. In a binaural, reverberant environment, the interaural characteristics of an acoustic source exhibit structure that can be used to separate even without prior knowledge of location or room characteristics. For this scenario, we have developed MESSL, an EM-based system for source separation and localization. MESSL’s probabilistic foundation facilitates the incorporation of more specific source models; MESSL-EV incorporates the eigenvoice speech models for improved binaural separation in reverberant environments.

10:45

2aSC8. Dual stage probabilistic voice activity detector. Ivan Tashev (Microsoft Res., One Microsoft Way, Redmond, WA 98052), Andrew Lovitt, and Alex Acero (Microsoft Res., Redmond, WA 98052)

Voice activity detectors (VADs) are critical part of every speech enhancement and speech processing system. One of the major problems in practical realizations is to achieve robust VAD in conditions of background noise. Most of the statistical model-based approaches employ the Gaussian assumption in the discrete Fourier transform domain, which deviates from the real observation. In this paper, we propose a class of VAD algorithms based on several statistical models of the probability density functions of the magnitudes. In addition, we evaluate several approaches for time smoothing the magnitude response to achieve a more robust estimate. A large data corpus of in-car noise conditions is then used to optimize the parameters of the VAD, and the results are discussed.

11:05

2aSC9. Speech enhancement beyond minimum mean squared error with perceptual noise shaping. Lae-Hoon Kim, Kyung-Tae Kim, and Mark Hasegawa-Johnson (Dept. of ECE, Univ. of Illinois at Urbana-Champaign, 405 North Mathews Ave., Urbana, IL 61820, jhasegaw@illinois.edu)

Residual error signal after speech enhancement through linear filtering can be decomposed into two disjoint portions: speech signal distortion and background noise suppression. Speech is known to follow a super-Gaussian probabilistic distribution function (PDF) such as Laplacian, while background noise follows Gaussian PDF. Minimum mean squared error estimation requires only second order statistics not only for the noise but also for the speech. Therefore higher-order dependence of observed speech on the original speech may cause leakage of speech information into the error residual. This talk will formulate an optimization problem minimizing higher-order statistics (HOS) as well as energy of the signal distortion constrained by a limit on the maximum audibility of the residual noise. Note that due to the non-stationary nature of speech, we perform the speech enhancement in short overlapping frames. Minimizing HOS of the speech distortion ensures that the speech distortion includes only noise terms, with minimum leakage from the speech signal. The constraint on the residual noise margin prevents over-suppressing, which may result in unwanted speech distortion.

11:25

2aSC10. The role of temporal modulation processing in speech/non-speech discrimination tasks. Hong You and Abeer Alwan (Dept. of Elec. Eng., UCLA, 405 Hilgard Ave., Los Angeles, CA 90095, alwan@ee.ucla.edu)

In this paper, temporal modulation characteristics of speech and noise from the point of view of speech/non-speech discrimination are analyzed. Although previous psychoacoustic studies have shown that temporal modulation components below 16 Hz are important for speech intelligibility, there is no reported analysis of modulation components from the point of view of speech/noise discrimination. Our data-driven analysis of modulation components of speech and noise reveals that speech and noise are more accurately classified by low-pass modulation frequencies than band-pass ones [H. You and A. Alwan, in Interspeech Proceedings (2009) pp. 36–39]. Effects of additive noise on the modulation characteristics of speech signals are also analyzed. Based on the analysis, a frequency adaptive modulation processing algorithm for a noise robust automatic speech recognition task is proposed. Speech recognition experiments are performed to compare the proposed algorithm with other noise robust front-ends, including RASTA and ETSI-AFE. Recognition results show that the frequency adaptive modulation processing algorithm is promising and is of low complexity. [Work supported in part by NSF.]

Contributed Paper

11:45

2aSC11. Relevant spectro-temporal modulations for robust speech and nonspeech classification. Sridhar Krishna Nemala and Mounya Elhilali (Dept. of Elec. & Comput. Eng., 3400 N. Charles St., Barton 105, Baltimore, MD 21218)

Robust speech/non-speech classification is an important step in a variety of speech processing applications. For example, in speech and speaker recognition systems designed to work in real world environments, a robust discrimination of speech from other sounds is an essential pre-processing step. Auditory-based features at multiple-scales of time and spectral resolution have been shown to be very useful for the speech/non-speech classification

task [Mesgarani *et al.*, IEEE Trans. Speech Audio Process. **10**, 504–516 (2002)]. The features used are computed using a biologically inspired auditory model that maps a given sound to a high-dimensional representation of its spectro-temporal modulations (mimicking the various stages taking place along the auditory pathway from the periphery all the way to the primary auditory cortex). In this work, we analyze the contribution of different temporal and spectral modulations for robust speech/non-speech classification. The results suggest the temporal modulations in the range 12–22 Hz, and spectral modulations in the range 1.5–4 cycles/octave are particularly useful to achieve the robustness in highly noisy and reverberant environments.

TUESDAY MORNING, 20 APRIL 2010

GALENA, 9:10 TO 11:30 A.M.

Session 2aSP

Signal Processing in Acoustics and Underwater Acoustics: Arrays in Water and Air

Charles F. Gaumont, Chair
Naval Research Lab., 4555 Overlook Ave., Code 7142, Washington, DC 20375

Chair's Introduction—9:10

Contributed Papers

9:15

2aSP1. Estimating the size and spatial distribution of bubble clouds in an underwater acoustic test tank. Fred D. Holt, IV (Appl. Res. Lab. and Graduate Program in Acoust., The Penn State Univ., State College, PA 16804), J. Daniel Park, R. Lee Culver (The Penn State Univ., State College, PA 16804), David Coles, and Timothy Leighton (Univ. of Southampton, Southampton S017 1BJ, United Kingdom)

The AB Wood Underwater Acoustic test tank at ISVR, University of Southampton, UK is outfitted with a Venturi-based apparatus for generating

bubble clouds containing a large range of bubble sizes. Larger bubbles are allowed to rise out in a settling tank before the nearly opaque bubbly water is pumped into the acoustic test tank through a discharge manifold on the bottom. A series of acoustic attenuation measurements was made at AB Wood in July 2008 in order to estimate the size and spatial distribution of bubbles in the cloud. An acoustic projector transmitted 1-ms pure tones from 25 to 100 kHz, which were received at a co-linear three-element horizontal array. The receive hydrophones were spaced 0.5 m apart, with the center hydrophone placed directly over the bubble discharge location. Measurements were taken on-axis with the projector, and at three distances off-axis,

in 12-in. increments. The attenuation measurements were used to estimate the size and spatial distribution of bubbles within the cloud. The bubble size data will be used to support a study of the effect of nearby bubbles on array performance degradation. [Work sponsored by the Office of Naval Research, Code 321.]

9:30

2aSP2. Measurement and analysis of array gain degradation due to bubble scattering. J. Daniel Park, Fred D. Holt, IV, R. Lee Culver (Appl. Res. Lab., The Penn State Univ., P.O. Box 30, State College, PA 16804), David Coles, and Timothy Leighton (Univ. of Southampton, Southampton, United Kingdom)

When an array is steered in the direction of a signal, array gain (AG) is maximum when the signal is coherent across the array, meaning that the signals add in phase for all elements and the noise or interference is incoherent across the array (i.e., it adds with random phase). For an acoustic array operating in the ocean, we would like to understand the degree to which scattering by nearby bubbles degrades AG. Bubble attenuation can also degrade array performance by attenuating the signal of interest, but that is separate from AG degradation. The degradation of AG due to scattering by nearby bubbles has been measured for different bubble densities. We have analyzed the relationship between bubble density at the array and the degradation in AG. We present probability density functions (pdfs) of signal amplitude and phase at the array elements. The amplitude pdfs can be approximated as a Rayleigh-Rice distribution, and the phase pdfs follow the von Mises distribution. With independence assumed between amplitude and phase distributions, simulated distributions generate array gain distributions closely matching the measured AG distribution. [Work sponsored by ONR Undersea Signal Processing.]

9:45

2aSP3. Array performance in a complex littoral environment. Steven L. Means (Naval Res. Lab., Code 7120, 4555 Overlook Ave. SW, Washington, DC 20375, steve.means@nrl.navy.mil), Richard M. Heitmeyer (Global Strategies Group Inc., Washington, DC 20375), and Stephen C. Wales (Naval Res. Lab., Washington, DC 20375)

In August of 2007 a long (~1-km), 500 phone linear array began collecting acoustic data in the waters (~260 m in depth) approximately 12 km off the coast of Fort Lauderdale, FL. The array consisted of four, 125-phone segments deployed closely along a line running nearly east and west. Marine-band radar data were collected concurrently so that shipping in the region of the array could be tracked. The data considered in this analysis were obtained over the month of August and contains ~19 days of measurements. Array performance is investigated by beamforming at a number of frequencies (up to ~420 Hz) and apertures then determining cumulative distribution functions as a function of bearing and noise window statistics. The results are compared for day, night, weekday, and weekend measurements during which the local shipping varies significantly. Additionally, ship tracks obtained from the acoustic array are compared against those obtained from radar. [Work supported by ONR base funding at NRL.]

10:00

2aSP4. Results of hermetic transform signal processing to enhance the resolution and array gain of underwater acoustic arrays. Harvey C. Woodsum (Bayshore Labs Div., Sonotech Corp., 10 Commerce Park North Unit 1, Bedford, NH 03110)

Results of applying the discrete hermetic transform (DHT) to the beamforming of underwater acoustic arrays are presented in terms of measured reductions in beam mainlobe width as well as the associated improvement in directivity index/array gain for practical cases of interest. Application of the DHT to array beamforming is shown to produce substantially enhanced resolution relative to the conventional beamforming diffraction limit as well as significant enhancement in array gain against ambient noise, without the use of data adaptive or nonlinear processing. As a result, enhanced sonar signal detection and/or the ability to use substantially smaller than normal arrays

can be accomplished through the judicious use of DHT based beamforming algorithms. Results are favorably compared to theoretical predictions of algorithm performance.

10:15—10:30 Break

10:30

2aSP5. A mobile acoustic multiple-input/multiple-output communication testbed. Aijun Song, Mohsen Badiy, and Arthur Trembanis (College of Earth, Ocean, and Environment, Robinson Hall-114, Newark, DE 19716)

Underwater acoustic data communication is critical for naval and scientific underwater missions. For example, high-rate telemetry between underwater autonomous underwater vehicles (AUVs) and surface platforms can facilitate adaptive sampling of the ocean. Multiple-input/multiple-output (MIMO) systems can deliver significant increased channel capacity for underwater communications. At the University of Delaware, an acoustic MIMO communication testbed on a Gavia AUV has been developed for digital communication measurements at the frequency band of 20–30 kHz. The Gavia AUV is a modular, small-size vehicle (20 cm in diameter, 77 kg in air) with a depth rating of 200 m. With advanced navigation systems (INS) and surface communication capabilities (WiFi and Iridium), it has been tested through various coastal-ocean missions. The MIMO system on board is designed for easy, low-cost deployment, and is capable of conducting acoustical sampling via a towed array with or without single-element or multi-element source transmission. In the presentation, the design concept, component detail, and engineering test results will be shown. [Work supported by ONR 3210A.]

10:45

2aSP6. Experimental validation of Helmholtz equation least squares when the Nyquist spatial sampling requirement is violated. Richard E. Dziklinski, III and Sean F. Wu (Dept. of Mech. Eng., Wayne State Univ., 5050 Anthony Wayne Dr., Detroit, MI 48202)

Previous numerical studies [Dziklinsky and Wu, J. Acoust. Soc. Am. (2009)] have shown that Helmholtz equation least squares (HELSSs) enable one to violate the spatial Nyquist sampling frequency and stand-off distance requirements inherent in Fourier acoustics. A direct benefit of this severe under sampling using HELS is a significant saving in measurement and computational effort in locating acoustic point sources in practice without loss of the required spatial resolution. The presented paper aims at validating the previous numerical studies by conducting experiments on locating two incoherent point sources separated by a small distance of 6.25 mm via HELS. The acoustic pressures are measured by a 5×5 microphone array over a 50×50 mm square plane at varying stand-off distances with fixed microphone spacing of 12.5 mm. The considered source frequencies are well above the spatial Nyquist sampling frequency. Results show that HELS is capable of locating point sources when stand-off distance is 10 mm or less and SNR is 10 dB or higher. Similar results obtained by using planar Fourier acoustics are also presented for comparison purposes.

11:00

2aSP7. Implementation of sound ball with acoustic contrast control method. Min-Ho Song (Grad. School of Culture Technol., KAIST, Sci. Town, Daejeon 305-701, Korea, godspd@kaist.ac.kr) and Yang-Hann Kim (KAIST, Sci. Town, Daejeon 305-701, Korea)

It is well known that the problem of generating sound in the region of interest, by using finite number of speakers, is mathematically ill-posed. This problem can be well-posed; in other words, the way to drive the speakers to make desired sound field in the prescribed zone can be directly determined by using energy measure, which is called acoustic brightness, contrast control method. With the method, we can maneuver sound ball or balls in space which can generate personal listening zone or virtual sound source. In this paper, a novel way that guides a way to design the speaker array in space to generate sound ball will be introduced. The signals between the zone of interest and the speakers are interpreted as vector spaces. The vector

space interpretation certainly provides insight to design the arrays for acoustic brightness or contrast control. Theoretical formulation in vector space and experimental results of generating the sound ball will be introduced.

11:15

2aSP8. Mode filters and energy conservation. Ilya A. Udovydchenkov (Dept. AOPE, MS#9, Woods Hole Oceanograph. Inst., 266 Woods Hole Rd., Woods Hole, MA 02543), Michael G. Brown (Univ. of Miami, Miami, FL 33149), and Irina I. Rypina (Woods Hole Oceanograph. Inst., Woods Hole, MA 02543)

The discrete form of the mode filtering problem is considered. The relevant equations constitute a linear inverse problem. Solutions to problems of this type, including the mode filtering problem, are subject to a well-known trade-off between resolution and precision. But, unlike the typical linear inverse problem, the correctly formulated mode filtering problem is subject to an energy conservation constraint. This work focuses on the importance of satisfying, approximately at least, the energy conservation constraint when mode filtering is performed. [Work supported by ONR.]

TUESDAY MORNING, 20 APRIL 2010

LAUREL C/D, 8:25 TO 11:30 A.M.

Session 2aUW

Underwater Acoustics, Acoustical Oceanography, and Signal Processing in Acoustics: Acoustic Particle Velocity and Vector Fields: Signal Processing and Communication Applications I

Mohsen Badiey, Cochair

Univ. of Delaware, College of Earth, Ocean, and Environment, Newark, DE 19716-3501

Ali Abdi, Cochair

New Jersey Inst. of Technology, 323 King Blvd., Newark, NJ 07102

Chair's Introduction—8:25

Invited Papers

8:30

2aUW1. Studies of acoustic particle velocity measurements in the ocean. Gerald D'Spain (Marine Physical Lab., Scripps Inst. of Oceanogr., 291 Rosecrans St., San Diego, CA 92106, gdspain@ucsd.edu)

This presentation reviews previously published results from applying various signal and array processing methods to the analysis of simultaneous measurements of acoustic particle velocity and acoustic pressure in the ocean. Particular focus is placed on interpreting the properties of the elements of the pressure/particle velocity data cross spectral matrix. These properties are used to evaluate measurements of the deep ocean infrasonic noise field during a period of rapidly changing wind speed. Numerical modeling of the evolution in directionality of the active and reactive acoustic intensity vectors during this time period demonstrates the influence of the continental slope on the flow of underwater acoustic energy. [Work supported by the Office of Naval Research.]

8:50

2aUW2. An ultra-low-frequency acoustic vector sensor. Dimitri M. Donskoy (Stevens Inst. of Technol., Hoboken, NJ 07030, ddonskoy@stevens.edu) and Benjamin A. Cray (Naval Undersea Warfare Ctr., Newport, RI 02841)

Passive underwater acoustic surveillance and oceanographic studies, for the most, have not been conducted at ultra-low frequencies (ULFs), from 0.01 Hz to a few hertz. One of the primary reasons for neglecting ULF has been the lack of sensitive, and directional, ULF acoustic sensors. Existing, and many prototyped, vector sensor designs are inertia based and thus not well suited to ULF, typically these sensors that operate at frequencies well above 100 Hz. A new type of acoustic vector sensor, capable of operating within the ULF range and possessing unsurpassed sensitivity (minimal detectable signal levels well below 1 nm/s) was built, tested, and calibrated at the Naval Undersea Warfare Center, Newport, RI. The measurements have validated the proposed detection concept, provided initial performance parameters, and have set forth the foundation for designing the next generation of fieldable prototypes.

9:10

2aUW3. Modeling the acoustic vector field using parabolic equation and normal mode models. Kevin B. Smith (Dept. of Phys., Naval Postgrad. School, Monterey, CA 93943)

Numerical algorithms for computing the acoustic vector field, i.e., particle velocity and pressure from an acoustic propagation model, are introduced. Implementation using both a parabolic equation and normal mode approach is considered. The parabolic equation model employed uses a split-step Fourier algorithm, although application of the technique is general to other parabolic equation models. Expressions for the normal mode equations are also presented for both coupled and adiabatic mode models. Results for a Pekeris waveguide are presented for a point source, prompting a brief discussion of multipath influence on the estimation of the direction of energy flow. Approximate analytic solutions are used to validate the general results of both models. Results for the range-dependent benchmark wedge are then presented and show generally good agreement between the two types of models. The results from the two-way, coupled normal mode model provide potential benchmark solutions for the wedge and a means of confirming the accuracy of other models. [Work supported by ONR 3210A.]

9:30

2aUW4. The cooperative array performance experiment: A joint China-US vector field experiment. Daniel Rouseff (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, rouseff@apl.washington.edu), Zhongkang Wang, Shihong Zhou, Hong Meng, and Lisheng Zhou (Hangzhou Appl. Acoust. Res. Inst., Hangzhou City, China)

The cooperative array performance experiment (CAPEX) was performed in Lake Washington near Seattle in September 2009. Acoustic transmissions in the 1.5–4 kHz band were recorded simultaneously on two vertical arrays: one a conventional 32-element pressure-sensor array and the other an 8-element array that measured both pressure and the three orthogonal components of acoustic particle velocity at each element. The present talk is an overview of both the data collected and the hardware used during CAPEX. Data were collected on the stationary arrays for both stationary- and towed-source scenarios in water 60 m deep. The source-receiver range varied between 10 m and 4 km. The data collected at short range demonstrate the relationship between the pressure and particle velocity fields. At more distant ranges, the particle velocity data are used to estimate the bearing to the source. Experimental results are compared to predictions generated using numerical models. [Work supported by ONR.]

9:45

2aUW5. Using the vector sensors as receivers for underwater acoustic communications. T. C. Yang and Alenka Zajić (Naval Res. Lab., Washington, DC 20375, yang@wave.nrl.navy.mil)

A vector sensor package (VSP) generally consists of three orthogonally oriented sensors and an omni directional hydrophone, all packed in a small compact housing. Because of its small size, the VSP might be more useful than a hydrophone array. In contrast to the hydrophone array, which has a non-negligible vertical aperture and is difficult to deploy on an autonomous underwater vehicle (AUV), the VSP can be easily deployed on the AUV and used for underwater acoustic communications (UAC). However, the usefulness of the VSP depends on whether it provides the same spatial diversity as a hydrophone array. Hence, to address this question, this paper investigates the spatial correlation characteristics of the VSP and the bit error rates and/or output SNRs, based on at sea data, between the VSPs and the hydrophone array. The data were collected in May of 2009 on the New Jersey shelf. A hydrophone array and a VSP were deployed close to the ocean bottom and used for UACs. The source was towed at a slow speed, transmitting signals of various modulations at distances 0.5 to a few kilometers from the receivers. Initial results will be presented. [Work supported by the Office of Naval Research.]

10:00—10:30 Break

10:30

2aUW6. Sensitivity analysis of acoustic channel characteristics to sea surface spectral uncertainty. Allan Rosenberg and Qinqing Zhang (11100 Johns Hopkins Rd., Laurel, MD 20723)

The interaction of sound with the sea surface is important for underwater acoustic communications. The high-frequency sound used, 10 kHz and above, is sensitive to surface wave frequencies well above the ~ 0.5 -Hz upper limit routinely measured by wave buoys. Accurately measuring the surface in the short gravity wave regime is difficult and even the general shape of the spectrum is uncertain. The primary measurement challenge is to disentangle the effects of the instrument, including its supporting structure, from what one is attempting to measure. There have been attempts to combine the copious data at low frequencies and the sparse data at higher frequencies to produce model spectra depending on a few parameters that describe the spectrum in the short gravity wave region and above. In this work we study the sensitivity of the acoustic channel characteristics such as the channel impulse response to our uncertain spectral knowledge. We merge low-frequency surface wave spectra measured at NDBC 44014, with various modeled higher-frequency spectra generated from measured environmental

parameters to get unified spectra. For each unified spectrum we generate surface realizations, feed them into a rough surface parabolic equation model to compute a channel impulse response, and compare the impulse responses under different surfaces.

10:45

2aUW7. Determination of the location of a sound source in three dimensional based on acoustic vector sensors on the ground. Hans Elias de Bree (Microflown Technologies B.V., P.O. Box 300, 6900 AH Zevenaar, The Netherlands, debree@microflown.com)

An acoustic vector sensor (AVS) consists of three orthogonal particle velocity sensors in combination with a sound pressure microphone. In several publications it has been proven that multiple sources can be located in three dimensions with a single AVS. In this paper it will be shown that it is possible to measure the instantaneous location (this means bearing, elevation, and range) of a single dominant sound source in three dimensional space as well as the angle dependent local ground impedance. Theory as well as results of experiments will be presented.

11:00

2aUW8. A particle velocity gradient beam forming system. Hans-Elias de Bree (Microflown Technologies B.V., P.O. Box 300, 6800AH Zevenaar, The Netherlands, debree@microflown.com)

The topic of this paper is the determination of the acoustic source distribution in the far field with a small, three dimensional (3-D) system consisting closely spaced sound pressure sensitive microphones and particle velocity sensitive Microflown. Sound pressure sensors do have a zero order directionality (that is, no directionality). Particle velocity vector sensors have a first order directionality (this is a cosine shape directionality). With two closely spaced zero order sensors, a first order system can be created. Disadvantages are the low sensitivity for low frequencies and a limited high-frequency response. With two closely spaced first order sensors, a second order system is created. The directionality is a squared cosine shape. With this higher directivity it is possible to create a very small 3-D beam forming system with a reasonable resolution which is the topic of this paper. A second order system can be made with accelerometers or pressure sensors; however, the low-frequency response is very poor. The Microflown has a very high sensitivity at low frequencies so the velocity gradient signal is good. In this paper the velocity gradient method is presented, and a 3-D velocity gradient system is demonstrated.

11:15

2aUW9. An overview of underwater acoustic communication via particle velocity channels: Channel modeling and transceiver design. Ali Abdi (Dept. Elec. & Comput. Eng., New Jersey Inst. of Technol., 323 King Blvd., Newark, NJ, 07102, ali.abdi@njit.edu), Aijun Song, and Mohsen Badiy (Univ. of Delaware, Newark, DE 19716)

Over the past few decades, the scalar component of the acoustic field, i.e., the pressure channel, has been extensively used for underwater acoustic communication. In recent years, vector components of the acoustic field, such as the three components of acoustic particle velocity, are suggested for underwater communication. Consequently, one can use vector sensors for underwater communication. The small size of vector sensor arrays is an advantage, compared to pressure sensor arrays commonly used in underwater acoustic communication. This is because velocity channels can be measured at a single point in space. So, each vector sensor serves as a multi-channel device. This is particularly useful for compact underwater platforms, such as autonomous underwater vehicles (AUVs). Funded by the National Science Foundation, our research efforts focus on the research problems in two closely related categories: channel modeling and transceiver design. Channel modeling research aims at characterization of those aspects of acoustic particle velocity channels such as delay and Doppler spread, and transmission loss, which determine the communication system performance. Transceiver design addresses optimal use of vector sensors and particle velocity for data modulation and demodulation, equalization, synchronization, and coding. [Work supported by NSF.]

Session 2pAAa

Architectural Acoustics and Speech Communication: Speech Intelligibility and Privacy

Eric L. Reuter, Chair

*Reuter Associates, LLC, P.O. Box 4623, Portsmouth, NH 03802-4623**Invited Papers*

1:00

2pAAa1. Speech privacy: The new 2010 architectural guidelines. David Sykes (Remington Partners, 23 Buckingham St., Cambridge, MA 02138, dsykes@speechprivacy.org), William Cavanaugh, and Gregory Tocci (Cavanaugh Tocci Assoc., Sudbury, MA)

In January, speech privacy performance criteria were released in the 2010 Guidelines for the Design and Construction of Healthcare Facilities, which is accepted as code by ~42 states and many federal, state, and municipal agencies. This is the first acceptance by regulatory authorities of speech privacy and will impact healthcare facilities across the United States. The criteria have also been adopted as the reference standard for two environmental quality credits in LEED HC and GGHCv2. In addition new 1.5 million dollar penalties took effect on Nov 30, 2009 for violations of the HIPAA Privacy Rule, the first serious penalties since the law was written in 1995. The new speech privacy criteria simplify enforcement by accepting as “equivalent” all four of the established measurement systems (articulation index, privacy index and sound transmission index, and speech intelligibility index), thereby leaving regulators, courts, and professionals with a choice. The developers are working on the next edition (2014) when they hope to introduce the fifth and newest measurement system, S/N(A), which provides specific metrics for both “confidential” and “secure” privacy. Acoustical professionals will need to assist healthcare organizations, regulatory authorities, judges, lawyers, and consumer groups to understand the new criteria.

1:20

2pAAa2. Speech privacy in healthcare: A facility director’s and staff. Nikki Rineer (Hope Within Community Health Ctr., 4748 East Harrisburg Pike, Elizabethtown, PA 17022)

Hope Within is a healthcare center that has gone the extra mile to have their architecture spaces evaluated for speech privacy. Once identified they have made accommodations to minimize the risk of overheard conversations and have an ongoing program in place to maintain and improve speech privacy performance. Much has been discussed regarding speech privacy in healthcare, but this paper will discuss the point of view from the director and staff.

1:40

2pAAa3. Measurement of speech privacy of closed rooms using the American Society for Testing and Materials ASTM E2638. Bradford N. Gover and John S. Bradley (Inst. for Res. in Construction, Natl. Res. Council, Ottawa, ON K1A 0R6, Canada, brad.gover@nrc-cnrc.gc.ca)

The degree of speech privacy between various locations in buildings has been of increasing concern, driven in part by legislative requirements and the desire for increased security. Until recently, ASTM E1130 “Standard Test Method for Objective Measurement of Speech Privacy in Open Plan Spaces Using Articulation Index” has been the only standardized measurement method available. This test method is, for a variety of reasons, not suitable for assessing speech privacy of closed rooms or conditions of very high-speech privacy. These issues are, however, addressed by the new ASTM E2638 “Standard Test Method for Objective Measurement of the Speech Privacy Provided by a Closed Room.” ASTM E2638 describes a test method suitable for enclosed rooms of nearly all sizes and defines a measure called speech privacy class that can be used to accurately rate speech privacy over a very broad range, from no privacy to very high secrecy. The new E2638 measurement method will be described, and measurements in several real rooms will be presented.

2:00

2pAAa4. Setting speech privacy criteria for closed rooms in terms of speech privacy class values. John S. Bradley and Bradford N. Gover (Natl. Res. Council, 1200 Montreal Rd., Ottawa, Canada)

The new ASTM E2638 standard defines speech privacy class (SPC) as the sum of the measured average noise level at the position of a potential eavesdropper outside the room and the measured level difference between a source room average and the transmitted levels at the same potential eavesdropper location. For a given situation, the likelihood of transmitted speech being audible or intelligible can be related to the probability of higher-speech levels occurring in the meeting room. Increasing speech privacy criteria can be defined in terms of increasing SPC values. For a particular meeting room speech level, there is an SPC value for which transmitted speech would be below the threshold of intelligibility or even below the threshold of audibility. One can therefore create a set of increasing SPC values corresponding to increasing speech privacy and for each SPC value, one can give the probability of transmitted speech being either audible or intelligible. This paper describes a new procedure that makes it possible to specify degrees of speech privacy for meeting rooms and offices in terms of the expected risk of a privacy lapse and varying from quite minimal to extremely high-speech privacy.

2:20

2pAAa5. Low background noise is needed for good student-to-student classroom communication. David Lubman (DL Acoust., 14301 Middletown Ln., Westminster, CA 92683, dlubman@dlacoustics.com)

ANSI standard S12.60 requires that background noise not exceed 35 dBA in unoccupied lecture classrooms. Despite the strong and broad research basis for ANSI's 35-dBA requirement, some classroom noise guidelines permit 45 dBA (notably California Collaborative for High Performance Schools and Leadership in Energy and Environmental Design). How did this come about? The 45-dBA limit was a compromise between the 50-dBA guidelines then prevailing and ANSI's 35-dBA limits. There is no audiological basis for a 45-dBA BNL limit. With 45-dBA BNLs, the speech to noise ratio (SNR) in the rear of small classrooms is well below the 15-dB minimum required for sustained learning. With 35-dBA BNLs, the SNR approaches ANSI's 15-dB minimum. It is underappreciated that BNLs of 35 dBA or less also promote better student-to-student (STS) communication. For every 10 dB of background noise reduction, the physical range of STS communication increases by a factor of about 3.16, and the floor area encompassed increases by a factor of 10. As BNL is reduced, eventually all classroom occupants are within communication range.

2:40

2pAAa6. A primer on classroom acoustics. Gary W. Siebein (School of Architecture, Univ. of Florida, P.O. Box 115702, Gainesville, FL 32611), Reece Skelton, Keely M. Siebein, and Chris P. Jones (Siebein Assoc., Inc., Gainesville, FL 32607)

Research involving quantitative and qualitative evaluation of classroom activities has shown that speech perception and intelligibility in elementary, middle, and high schools can be measured, modeled, and simulated based on an impulse response-based theory. The theory combines the vocal production of the speaker, the hearing abilities of the listeners, the strength of the direct sound and early sound reflections from room surfaces, duration of longer term reverberation, and the level, type, information content, and frequency content of intruding sounds. Regression models relating measured intelligibility in classrooms to measured, modeled, and simulated sounds show model $r^2 > 0.90$. A model relating speech perception score to the level, type, information content, and frequency content of typical classroom noise reaches $r^2 > 0.90$ as well. The impulse response based measures are reduced to a series of architectural systems that can be implemented using alternate construction systems to meet budget requirements.

3:00—3:15 Break

3:15

2pAAa7. Assessing the potential intelligibility of assistive listening systems. Peter Mapp (Peter Mapp Assoc., Colchester, United Kingdom, peter@petermapp.com)

Approximately 14% of the general population suffer from a noticeable degree of hearing loss and would benefit from some form of hearing assistance or deaf aid. Recent DDA legislation and requirements mean that many more assistive listening systems are being installed, yet there is evidence to suggest that many of these systems fail to perform adequately and provide the benefit expected. There has also been a proliferation of classroom and lecture room "soundfield" systems, with much conflicting evidence as to their apparent effectiveness. This paper reports on the results of some trial acoustic performance testing of such systems. In particular, the effects of system microphone type, distance, and location are shown to have a significant effect on the resultant performance. The potential of using the sound transmission index and, in particular, STIPa for carrying out installation surveys has been investigated, and a number of practical problems are highlighted. The requirements for a suitable acoustic test source to mimic a human talker are discussed. The findings discussed in the paper are also relevant to the installation and testing of "soundfield systems."

3:35

2pAAa8. Effects of noise type and signal-to-noise ratio on speech intelligibility and the speech transmission index. Stephen D. Secules (Arup Acoust., 13 Fitzroy St., London W1T 4BQ, United Kingdom)

The STI metric is the most reliable measure we have for assessing the intelligibility of speech in public address/voice alarm or speech reinforcement systems. A good design for speech is based on an understanding of the room acoustics and electroacoustics of the system, with a consideration for the expected signal to noise ratios in the room. In modeling, most noise signals are assumed to be broadband, whereas in reality the noise signals can be significantly variable in the frequency and time domain, which may dramatically increase or reduce the intelligibility. This study compares the effects of type of noise on intelligibility as measured through both word scores and STI. It explores masking from both theoretical and real world signals. The determined relationships will help inform the design process by clarifying the effects of non-broadband and variable noise signals on intelligibility.

Contributed Papers

3:55

2pAAa9. Comparing head rotation angle, visual localization, and recall-proficiency of school-aged children 8–12 while listening to a story by multiple discreet talkers in a virtual classroom. Daniel L. Valente, Dawna E. Lewis, John M. Franco, and Elizabeth C. Heinrichs (Dept. of Res., Boys Town Natl. Res. Hospital, 555 N. 30th St., Omaha, NE 68131)

In general, children perform more poorly in speech intelligibility tasks than adults while in a noisy environment such as a classroom. This is especially true in discussion situations where the active talkers often are located to the side or behind the student. The purpose of this study was to obtain baseline data for normal hearing (NH) students who experienced multiple-talker stimuli in a controlled virtual classroom environment. Data were gath-

ered using a gyroscopic head-tracker and a post-story comprehension task. Twenty elementary-aged students (ages 8–12, four students per year) were positioned in the center of the classroom and presented a story read by five talkers positioned around the student (reproduced by means of loudspeakers and LCD monitors). Students used different strategies in terms of head rotation over time while performing the experiment and individual differences were seen both across subjects within an age group and between age groups. The post-test comprehension task was used as a metric of performance in the environment. Results from the NH students are compared to a control group of adult NH listeners using head rotation angle over time, comprehension scores, and localization accuracy as salient metrics.

4:10

2pAAa10. Verifying speech intelligibility in classrooms. Roman Wowk (Papadimos Group, 818 Fifth St., Ste. 207, San Rafael, CA 94901)

Classroom speech intelligibility has undoubtedly become a hot topic and is capturing attention on all fronts. While there has been widespread adoption of many new guidelines and standards, even the most precise initiatives will not reach their full potential if speech intelligibility is not directly verified in classrooms after they are built. Performance verification ultimately encourages a better final product and provides crucial measured data for validating design methods. To ensure measurable performance, criteria would have to be established at the outset of the design using a metric directly related to speech intelligibility such as speech transmission index. Current guidelines generally focus on reverberation time (RT) and background noise for improving speech communication in classrooms. While low background noise is a necessary component for speech intelligibility, there is not always a direct correlation between RT and speech intelligibility. For instance, improper placement of absorptive treatments for reverberation control could eliminate early supporting reflections which are beneficial for speech intelligibility. With computer modeling becoming widespread and relatively inexpensive, designing directly for speech intelligibility is becoming more practical. This paper explores case studies and examines the overall process for optimizing speech communication in classrooms with an emphasis on performance verification.

4:25

2pAAa11. Ambient noise levels and reverberation times in Mississippi school rooms. Edward L. Goshorn, Megan N. Lucus, and Brett E. Kemker (Dept. of Speech and Hearing Sci., Psychoacoustics Lab., Univ. of Southern Mississippi, 118, College Dr. #5092, Hattiesburg, MS 39406)

Nine elementary school classrooms at three Mississippi public schools were selected at random for noise and reverberation time measures to monitor voluntary compliance with ANSI standard S12.60-2002 (Acoustical Per-

formance Criteria, Design Requirements, and Guidelines for Schools). 1-h equivalent dBA noise levels and reverberation times (T60) were measured with SPECTRAPLUS Version 5.0 software. Measures were taken on separate days at each school in unoccupied classrooms with the HVAC system operating. For ambient noise measures, an Audix TR40 omnidirectional microphone was placed in a 12 in. stand on a student's desk near the middle of the room and oriented toward the teacher's desk at a 45 deg angle. For the reverberation time measure a 400 W loudspeaker (Yamaha BR15M) was placed on the teacher's desk about 3 m from the microphone. Results showed that noise levels varied from 28 to 51 dBA with broad spectra containing peaks at 120 Hz and harmonics. Five of nine classrooms had noise levels that exceeded the ANSI recommended level of 35 dBA. Reverberation times varied from 0.21 to 0.62 s; only one classroom exceeded the recommended minimal reverberation time of 0.6 s. Acoustical modifications to reduce noise levels and reverberation times will be addressed.

4:40

2pAAa12. Voice privacy associated with raised floor law office: A case study. Christopher J. Pollock and Geoffrey Sparks (Shen Milsom & Wilke, LLC, 3300 N Fairfax Dr., Ste. 302, Arlington VA 22201, cpollock@smwllc.com)

Presentation and discussion of the acoustical transmission and voice privacy issues associated with raised floor office fit out construction. Many new LEED office fit out projects are using raised floor constructions for future flexibility and use of underfloor plenum for HVAC systems and/or technology cable plant. Issues of user expectations of privacy versus capabilities of the system are becoming more common. Case study office fit out was completed with raised floor and the extent of voice transmission due to various construction conditions limited voice privacy and created noise disruption from offices, workrooms, and conference rooms. Case study involves a law office where privacy concerns are heightened due to sensitive discussions. Measurements of noise isolation and privacy were made both before and after modifications, and results will be presented and discussed and conclusions for practitioners will be discussed.

2p TUE. PM

TUESDAY AFTERNOON, 20 APRIL 2010

1:00 TO 3:00 P.M.

GRAND BALLROOM FOYER WEST/NORTH

Session 2pAAb

Architectural Acoustics and National Council of Acoustical Consultants: Student Design Competition

Andrew Miller, Cochair

BAi, LLC, 4006 Speedway, Austin, TX 78751

Byron Harrison, Cochair

The Talaske Group Inc., 1033 South Blvd., Ste. 101, Oak Park, IL 60302

Robert C. Coffeen, Cochair

University of Kansas, School of Architecture and Urban Design, Marvin Hall, Lawrence, KS 66045

The Technical Committee on Architectural Acoustics of the Acoustical Society of America, the Robert Bradford Newman Student Award Fund, and the National Council of Acoustical Consultants are sponsoring the 2010 Student Design Competition that will be professionally judged at this meeting. The purpose of this design competition is to encourage students enrolled in architecture, engineering, physics, and other university curriculums that involve building design and/or acoustics to express their knowledge of architectural acoustics in the design of performance hall primarily for opera on a college campus of moderate size.

This competition is open to undergraduate and graduate students from all nations. Submissions will be poster presentations that demonstrate room acoustics, noise control, and acoustic isolation techniques in building planning and room design. The submitted designs will be displayed in this session and they will be judged by a panel of professional architects and acoustical consultants. An award of \$1250.00 US will be made to the entry judged "First Honors." Four awards of \$700.00 US will be made to each of four entries judged "Commendation."

Session 2pAB

Animal Bioacoustics: Estimating Spatial Density of Animal Populations with Passive Acoustics

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

Chair's Introduction—1:00*Invited Papers*

1:05

2pAB1. Minke whale boing vocalization density estimation at the Pacific Missile Range Facility, Hawaii. Stephen W. Martin (Space and Naval Warfare Systems Ctr. Pacific, 53366 Front St., San Diego, CA 92152), Len Thomas, Tiago A. Marques (Univ. of St. Andrews, St. Andrews KY16 9LZ, Scotland), Ronald P. Morrissey, Susan Jarvis, Nancy DiMarzio (Naval Undersea Warfare Ctr., Newport, RI 02841), and David K. Mellinger (Oregon St. Univ. and NOAA Pacific Marine Environ. Lab., Newport, OR 97365)

Minke whales (*Balaenoptera acutorostrata*) seasonally visit the Hawaiian Islands with an apparent peak in activity in the February/March timeframe. Their presence is readily detected acoustically from their boing vocalizations, which are detectable on the Pacific Missile Range Facilities (PMRF) deep-water (3500 m to over 4500-m depth) bottom mounted hydrophones northwest of Kauai, HI. Automated techniques were developed to detect, classify, and associate the boings across 16 separate hydrophones. Over 13 h of data for each of 12 separate days from the 2006 whale season (February/March/April) were processed for detection and classification (D/C) of boing vocalizations. A sample set of the data (10 min from each day) was analyzed to characterize the detection function (probability of detection as a function of horizontal range) and the false positive rate of the boing D/C. Boings were associated across hydrophones, defining a capture history for each boing vocalization. Spatially explicit capture recapture methods were applied to the sample data associations to derive the detection function. Applying the parameters estimated on the sample data set toward the full data set allows an estimation of the minke whale boing vocalization density in this area over these time periods in 2006.

1:25

2pAB2. Estimating density from single hydrophones by means of propagation modeling. Elizabeth T. Küsel, David K. Mellinger (Cooperative Inst. for Marine Resources Studies, Oregon State Univ., 2030 S. Marine Sci. Dr., Newport, OR 97365, elizabeth.kusel@oregonstate.edu), Len Thomas, Tiago A. Marques (Univ. of St. Andrews, Fife KY16 9AJ, Scotland), David J. Moretti, and Jessica Ward (Naval Undersea Warfare Ctr., Newport, RI 02841)

A density estimation method developed under the project Density Estimation for Cetaceans from passive Acoustic Fixed sensors is presented. It uses sound propagation modeling to estimate the probability of detecting an animal as a function of its distance from the receiving sensor. Two case studies involving Blainville's beaked whales (*Mesoplodon densirostris*) and sperm whales (*Physeter macrocephalus*) are analyzed. The study area is the Atlantic Undersea Test and Evaluation Center in the Tongue of the Ocean, Bahamas. A ray-tracing acoustic propagation model is used to estimate the environmental transmission loss as a function of depth and range in several directions away from a single hydrophone. The computed transmission loss is compared to ambient noise levels, source level, and beam pattern distributions available in the literature to estimate detection probability as a function of range. Detection threshold is characterized from the signal-to-noise ratio of detected clicks. Information on click production rate is also taken into account by the density estimation model. Beaked whale detection probability function provides a relevant comparison to both the detection function and the spatial density of whales derived empirically from the DTag data by Marques *et al.* [(2009)].

1:45

2pAB3. Population density of sperm whales in the Bahamas estimated using propagation modeling. David K. Mellinger, Elizabeth T. Küsel (Coop. Inst. for Marine Resources Studies, Oregon State Univ., 2030 SE Marine Sci. Dr., Newport, OR 97365, david.mellinger@oregonstate.edu), Len Thomas, Tiago Marques (Univ. of St Andrews, Scotland), David Moretti, Paul Baggenstoss, Jessica Ward, Nancy DiMarzio, and Ron Morrissey (Naval Undersea Warfare Ctr., Newport, RI)

The population density of sperm whales is estimated using an acoustic model to calculate the probability of receiving their clicks. The model uses estimates of (1) the source level of clicks, (2) the beampattern of the whales' emitted clicks, (3) the distribution of whale orientations, (4) the loss between source and receiver (derived from acoustic propagation modeling), (5) noise levels at the receiver, (6) the detector's rates of missed calls and false detections, and (7) sperm whales' average click rate. These data are combined in a model that propagates simulated clicks from whales at various simulated positions to the receiving hydrophone to estimate the detection function—the probability of receiving a click as a function of distance. This function is then combined with information on whale click rates (performed at a chosen time of day when sperm whales appear to click at a measurably predictable rate) to estimate the population density corresponding to the number of received clicks in a given period of time. Data from the U.S. Navy's Atlantic Undersea Test and Evaluation Center (AUTECE) in the Bahamas are used to estimate sperm whale population density there. [Thanks to ONR, NOAA, and the Navy's Environmental Readiness Division for funding.]

2pAB4. Density estimation of leopard seals using a single stationary passive acoustic sensor. Holger Klinck (Hatfield Marine Sci. Ctr., Oregon State Univ., 2030 SE Marine Sci. Dr., Newport, OR 97365, holger.klinck@oregonstate.edu), Nadine Constantinou (Univ. of New South Wales, Sydney, New South Wales 2052, Australia), David K. Mellinger (Oregon State Univ., Newport, OR 97365), and Tracey Rogers (Univ. of New South Wales, Sydney, New South Wales 2052, Australia)

The objective of this study is to estimate the spatial density of leopard seals using data recorded with a single stationary passive acoustic recording system in the Bransfield Strait, Antarctica, between 2005 and 2007. The most prominent vocalization of the leopard seal—the low double trill (LDT)—is used as a proxy for the presence of the species in the vicinity of the recording system. Because of the stereotypic nature and high frequency of occurrence of the LDT, a long-term spectrogram approach can be applied to the data sets to reliably detect the presence of the target species. Energy levels in the target frequency band (200–400 Hz) as derived by the long-term spectrogram analysis are related to number of manually counted calls extracted for selected periods. A linear regression analysis showed that energy levels are highly correlated with the number of manually counted calls. The number of recorded calls per unit time is converted into number of vocalizing animals per unit time by applying published calling rates for this species. In a last step of the analysis, the detection area is defined and leopard seal densities estimated. Opportunities and challenges of the method will be discussed.

2pAB5. Determining the spatial distribution of an Antarctic top predator using passive acoustics. Nadine E. Constantinou, Tracey L. Rogers (Evolution and Ecology Res. Ctr. (EERC), School of Biological, Earth and Environ. Sci. (BEES), Univ. of New South Wales (UNSW), Sydney, New South Wales 2052, Australia, n.constantinou@student.unsw.edu.au), Shawn W. Laffan, and David I. Warton (UNSW, New South Wales 2052, Australia)

The leopard seal (*Hydrurga leptonyx*) is one of four species of ice seals in Antarctica with each species occupying a distinct position in the Antarctic sea-ice ecosystem. Ice seals offer a potential source of information about ecosystem interactions and environmental variability integrated over a variety of spatial and temporal scales. During the austral spring and summer, leopard seals move within the pack ice to breed. Acoustic surveying is necessary to assess their distributions as male leopard seals vocalize underwater as part of their breeding display. During the 1999/2000 austral summer, the relative abundance of adult male leopard seals was determined using underwater passive acoustic point-transect surveys. The abundance data were combined with environmental data in a geographical information system, and a model was developed to determine what factors of the environment are correlated with their abundance and distribution. The model with the best predictive power showed a trend of increased abundance toward the pack ice edge. These regions are associated with areas of increased foraging potential suggesting that the distribution of leopard seals off Eastern Antarctica is influenced, in part, by increased availability of prey.

Contributed Papers

2pAB6. Combining visual and fixed passive acoustic methods to measure annual variability of cetacean occurrence at the NE-coast of Iceland. Edda E. Magnusdottir, Marianne H. Rasmussen (Husavik Res. Ctr., Univ. of Iceland, Hafnarstett 3, 640 Husavik, eem@hi.is), and Marc Lamers (Hawaii Inst. of Marine Biology, Kailua, HI 96834)

In order to measure cetacean occurrence at the NE-coast of Iceland, two seabed-mounted Ecological Acoustic Recorders (EARs) were deployed in Skjálfandi Bay in September 2008, deployments will continue throughout September 2010. To obtain species confirmation, visual observations were conducted from a lighthouse in October 2008 to September 2009 with total observation effort of 202 h. With visual observations, the aim was to acquire background information on the cetacean sighted in the area to improve the estimates of occurrence in continued long-term passive acoustic recordings in Icelandic and adjacent waters. Movements and locations of the animals were recorded using a theodolite and a real-time mapping program (CYCLOPS TRACKER). Visual observations included sightings of blue whales (*Balaenoptera musculus*), Humpback whales (*Megaptera novaeangliae*), Minke whales (*Balaenoptera acutorostrata*), white-beaked dolphins (*Lagomorhynchus albirostris*) and killer whales (*Orcinus orca*). A custom MATLAB program is used to automatically detect and analyze target biological sounds produced by cetaceans or by pelagic fish recorded on the EARs. The visual data are compared to the occurrence of identified sounds in order to better understand the vocal activity and average silence periods of the cetaceans. Also, detected vocalizations are compared to the number, location, and behavior of simultaneously sighted animals.

2pAB7. Seasonal and diurnal presence of finless porpoises at the corridor to the ocean from a semi-closed bay habitat. Tomonari Akamatsu (NRIFE, FRA, Hasaki, Kamisu, Ibaraki 314-0408, Japan), Kiyomi Nakamura (Kujyu Kushima Aquarium, Sasebo, Nagasaki 858-0922, Japan), Ryo Kawabe, Seishiro Furukawa, Hiromi Murata (Nagasaki Univ., Nagasaki 851-2213, Japan), Akihiro Kawakubo, and Masayuki Komaba (Kujyu Kushima Aquarium, Sasebo, Nagasaki 858-0922, Japan)

Finless porpoises in Omura Bay is the smallest among five populations in Japanese waters. This is a newly established population after the global warming of 9000 years ago, consisting of approximately 300 individuals and believed to be confined in Omura Bay. At the major corridor to the ocean from this bay, presence of finless porpoises was monitored acoustically from November 2007 to May 2009. A stereo acoustic event recorder (A-tag) was deployed at this strait. A-tag stored the intensity and the sound source direction of biosonar signals that provides independent traces of sound source corresponding to each animal. During 1.5 year observation, 226 individuals of porpoises were detected. Of them, 76% were presented at night and 73% observed during March to April. In the same season, anchovy landings in the Omura bay was recorded. 70% percent of the porpoises were observed during the tidal current that went out of Omura Bay that suggests the porpoises swam back and forth around the monitoring station. Although finless porpoises have been known to be confined in a local area, occasional migration of this species induced by prey availability might extend their habitat at different waters in Asian coasts.

2pAB8. Are passive acoustics an appropriate method to accurately represent a soniferous Atlantic croaker (*Micropogonias undulatus*) population? Cecilia S. Krahforst, Joseph J. Luczkovich, and Mark W. Sprague (Dept. of Biology, East Carolina Univ., Greenville, NC 27858, csk1021@ecu.edu)

It was observed that the fundamental frequency of Atlantic croaker sounds is inversely correlated with the length of the fish. We used fundamental frequency as a method to estimate the average length of the croaker population at any given time. Croaker were collected using an otter trawl and simultaneously counted and measured acoustically with a 200-kHz split-beam echosounder at two sites within the Pamlico Sound estuary, NC from

June to November 2008. Passive acoustic recorders [long-term acoustic recording system (LARS) recordings 10-s wave files < 10 kHz at 15-min intervals] were deployed near each trawling site to obtain *in-situ* recordings of croaker over the same period. Based on captive fish recordings, a linear regression analysis related total length to fundamental frequency, where $TL = 305.323 \text{ mm} - (0.270 \text{ Hz}^{-1})f_0$ ($r^2 = 0.84$). This equation was then used to estimate croaker lengths from LARS recordings. Lengths of fish collected in the trawls, compared with the estimate from the passive LARS recordings and the echosounder surveys, were significantly smaller ($P > 0.0001$). This suggests that size-selectivity of the trawl underestimates the average size of fish and that acoustic methods (active and passive) provide a more accurate size estimate for croaker populations.

TUESDAY AFTERNOON, 20 APRIL 2010

KENT A/B, 1:00 TO 5:00 P.M.

Session 2pBB

Biomedical Ultrasound/Bioresponse to Vibration and Physical Acoustics: Numerical Modeling for Medical Ultrasound I

Robert J. McGough, Cochair

Michigan State Univ., Dept. of Electrical and Computer Engineering, 2120 Engineering Bldg., East Lansing, MI 48824

Vera A. Khokhlova, Cochair

Moscow State Univ., Acoustics Dept., Leninskie Gory, Moscow, 119992, Russia

Chair's Introduction—1:00

Invited Papers

1:05

2pBB1. A parallel algorithm for high-intensity focused ultrasound simulation. Joshua E. Soneson (Div. of Solid and Fluid Mech., U.S. Food and Drug Administration, 10903 New Hampshire Ave., Silver Spring, MD 20993, joshua.soneson@fda.hhs.gov)

The Khokhlov–Zabolotskaya–Kuznetsov (KZK) equation serves as today's workhorse for the modeling and simulation of high-intensity focused ultrasound. This approximation of the full nonlinear wave equation allows serial algorithms to compute the pressure distribution produced by circular transducers usually in several hours on a modern workstation. However, more complicated transducer geometries preclude axisymmetric modeling, and extending into the third spatial dimension requires significantly more computational power, making serial algorithms intractable. This presentation details a parallel implementation of a combination time and frequency domain algorithm for rapid solution of the KZK equation. A split step method is used in which the linear terms in KZK are solved in the frequency domain and the nonlinear term is solved in the time domain. In the frequency domain, each harmonic may be computed independently, so this task is distributed over the available processors. In the time domain, the transverse spatial domain may be partitioned and distributed for independent solution of the nonlinear term. Near-linear scaling (acceleration vs number of processors) using this algorithm is demonstrated.

1:25

2pBB2. Analytic and numerical modeling of ultrasonic B-scan and echo decorrelation imaging. T. Douglas Mast and Swetha Subramanian (Dept. of Biomedical Eng., Univ. of Cincinnati, 231 Albert Sabin Way, Cincinnati, OH 45267-0586, doug.mast@uc.edu)

An numerical model is presented for *B*-scan images of weakly scattering, lossy media, based on ultrasound array beam patterns calculated analytically under the Fresnel approximation. Given these beam patterns and a three-dimensional analytic or numerical tissue model, this method yields beamformed *A*-line signals from which *B*-scan images are constructed. This approach is further employed to model echo decorrelation imaging, a method for quantitatively mapping transient heat-induced changes in pulse-echo ultrasound images. In echo decorrelation imaging, a normalized decorrelation parameter is computed between *A*-line signals separated by milliseconds. Maps of this parameter comprise echo decorrelation images, which are potentially useful for monitoring of local tissue coagulation during thermal ablation treatments for cancer therapy. Following previous studies in which scattering cross section has been related to spatial-frequency spectra of tissue sound speed, density, and impedance variations, echo decorrelation is related quantitatively to the local decoherence of these spatial-frequency spectra. Decoherence estimates are validated by simulations employing analytic array beam patterns and random-media models for ablated tissue and are further applied to quantify tissue structure changes caused by thermal coagulation during radiofrequency ablation of *ex-vivo* bovine liver tissue.

1:45

2pBB3. A modeling tool for therapeutic and imaging ultrasound applications. Francesco Curra (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105)

The fields of therapeutic and imaging ultrasound have broad medical applications. However, the inherent complexity of biological media and the nonlinear nature of ultrasound propagation at therapeutic regimes make optimization and control of the therapy still a challenging task. An accurate modeling tool for solving multi-dimensional ultrasound-based problems in complex geometries that can greatly assist in the optimization and control of the treatment is presented. The model consists of 2-D, 2.5-D (cylindrical symmetry), and 3-D coupled solutions for acoustic and elastic wave propagation in heterogeneous, lossy media complemented by the bioheat transfer equation for temperature estimation. It includes linear and nonlinear wave propagations, arbitrary frequency power law for attenuation, and can account for multiple reflections and backscattered fields. Work is underway for the inclusion of cavitation effects and the extension of the model to adaptive grid refinement and unstructured, deformable grids. Sample results for therapeutic and imaging applications will be presented. [Work supported by US Army MPMC and NIH NIDDK.]

2:05

2pBB4. Modeling of high-frequency ultrasound contrast agents. John Allen, Fanny Cugnet (Dept. of Mech. Eng., Univ. of Hawaii-Manoa, 2540 Dole St., Honolulu, HI 96822, alleniii@hawaii.edu), Jonathan Mamou, Paul Lee, Parag Chitnis, and Jeff Ketterling (Riverside Res. Inst., New York, NY 10038)

High-frequency applications of ultrasound contrast agents continue to develop with their increasing use for imaging intravascular locations as well as dermal and ocular tissue. However, conventional ultrasound contrast agents were originally designed for traditionally lower diagnostic frequencies. The optimal contrast agent design and related acoustic forcing methods for high-frequency applications are important topics of on-going research. In this respect, ultrasound contrast agent models are reviewed and, in particular, the role of the shell is highlighted for novel high-frequency applications. Shell material and related stability issues are discussed in terms of the constitutive equation assumptions. The concept of auto-resonance, which may be achieved by a slowly varying forcing frequency, is introduced as a method for enhancing the subharmonic response of free or bound agents. The phase locking characteristics of auto-resonance are analyzed using an empirical mode decomposition of the scattered signal. This provides the intrinsic mode functions from which the associated instantaneous frequencies and phases can be determined. The Shannon entropy and the variance of the phase are used to quantify the behavior. Comparisons of the numerical results are made with current experimental data. [Support from NIH-EB006372 and NIH-2G12-RR003016121.]

2:25

2pBB5. Modeling of nonlinear shock wave propagation and thermal effects in high-intensity focused ultrasound fields. Vera A. Khokhlova (Ctr. for Industrial and Medical Ultrasound, APL, Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105 and Faculty of Phys., Moscow State Univ., Moscow 119991, Russia, vera@acs366.phys.msu.ru), Olga V. Bessonova, Mikhail V. Averianov (Moscow State Univ., Moscow 119991, Russia), Joshua E. Sonesson (U.S. Food and Drug Administration, Silver Spring, MD 20993), and Robin O. Cleveland (Boston Univ., Boston, MA 02215)

Numerical simulations based on the Khokhlov-Zabolotskaya-type equation are currently used to characterize therapeutic high-intensity focused ultrasound fields in water and to predict bioeffects in tissue. Here results from three different algorithms that differ in calculating the nonlinear term in the equation are presented. Shock capturing schemes of Godunov type, exact implicit solution with further extrapolation of the waveform over a uniform temporal grid, and direct modeling in the frequency domain are tested. In the case of weak nonlinearity, all schemes give essentially the same solution. However, at high peak pressures around 50 MPa and strong shocks developed in the focal region, the predictions of acoustic variables and heat deposition become sensitive to the algorithm employed. The parameters of the schemes, such as number of harmonics or temporal samples and the inclusion of artificial absorption that provides consistent results, are discussed. It is shown that the spectral and Godunov-type approaches require about 6 points and implicit time domain approach needs more than 50 points in the shock to be accurate. In all schemes artificial absorption should be employed to obtain acceptable accuracy with fewer points per cycle. [Work supported by the NIH EB007643, ISTC 3691, RFBR 09-02-01530, and NSF 0835795 grants.]

2:45

2pBB6. Generalization of the iterative nonlinear contrast source method to realistic, nonlinear biomedical tissue. Martin D. Verweij (Lab. of Electromagnetic Res., Fac. of Elec. Eng., Mathematics and Comput. Sci., Delft Univ. of Technol., Mekelweg 4, 2628 CD Delft, The Netherlands)

Originally, the iterative nonlinear contrast source (INCS) method has been developed to compute the nonlinear, wide-angle, pulsed ultrasound field in a homogeneous and lossless medium. The method considers the nonlinear term of the lossless Westervelt equation as a distributed contrast source in a linear background medium. The full nonlinear wave field follows from the Neumann iterative solution of the resulting integral equation. Each iteration step involves the spatiotemporal convolution of the background Green's function with an estimate of the contrast source. Appropriate filtering provides accurate field predictions for a discretization approaching two points per smallest wavelength or period. The current paper first discusses the theoretical generalization of the original contrast source approach, e.g., to include tissue attenuation or inhomogeneity. Next, computational results are presented that show the directional independence (even for the nonlinear distortion) and the wide-angle performance of the original INCS method. Finally, results are presented for extensions that can deal with power law losses or inhomogeneity in the wave speed. The wide-angle capability and the versatility of the contrast source approach demonstrate that the INCS method is well-suited for modeling the various aspects of realistic biomedical tissue behavior. [Work supported by STW and NCF.]

3:35

2pBB7. Fractional wave equation for compressional and shear waves. S. Holm (Dept. Informatics, Univ. of Oslo, P.O. Box 1080, NO-0316 Oslo, Norway) and R. Sinkus (Inst. Langevin, ESPCI ParisTech, F-75231 Paris Cedex 05, France)

This study has been motivated by the observed difference in the range of the power law attenuation exponent for compressional and shear waves. Usually compressional attenuation increases with frequency to a power between 1 and 2, while shear wave attenuation often is described with powers less than 1. Another motivation is the apparent lack of partial differential equations with desirable properties such as causality that describe such wave propagation. Starting with a constitutive equation which is generalized Hooke's law with a loss term containing a fractional derivative, one can derive a causal fractional wave equation previously given by Caputo (1967) and Wismer (2006). In the low $\omega\tau$ (low-frequency) case, this equation has an attenuation with a power law in the range from 1 to 2. This is consistent with, e.g., attenuation in tissue. In the often neglected high $\omega\tau$ (high-frequency) case, it describes attenuation with a power law between 0 and 1, consistent with what is observed in, e.g., dynamic elastography. Thus a unifying wave equation derived properly from constitutive equations can describe both cases. Recent results on fractional random media will also be discussed as they give similar characteristics from a microperspective.

3:55

2pBB8. Transducer models in the ultrasound simulation program FIELD II and their accuracy. J. A. Jensen and D. Bk (Dept. of Biomedical Eng., Ctr. for Fast Ultrasound Imaging, Tech. Univ. of Denmark, Build. 349, DK-2800 Lyngby, Denmark, jaj@elektro.dtu.dk)

The FIELD II simulation program can be used for simulating any kind of linear ultrasound fields. The program is capable of describing multi-element transducers used with any kind of excitation, apodization, and focusing. The program has been widely used in both academia and by commercial ultrasound companies for investigation novel transducer geometries and advanced linear imaging schemes. The program models transducer geometries using a division of the transducer elements into either rectangles, triangles, or bounding lines. The precision of the simulation and the simulation time is intimately linked through the choice of the fundamental elements. The rectangular elements use a far-field approximation, whereas the two other methods use the full analytic solution, leading to a higher precision at the price of a slower simulation time. The talk will describe the different compromises and solutions to obtain a fast simulation and still attain a high precision including a newly developed semi-analytic solution for a convex surface elements.

Contributed Papers

4:15

2pBB9. Beamforming for vibro-acoustography using a 1.75D array transducer. Matthew W. Urban, Mostafa Fatemi, and Azra Alizad (Dept. of Physio. and Biomedical Eng., Mayo Clinic College of Medicine, 200 First St. SW, Rochester, MN 55905, urban.matthew@mayo.edu)

Vibro-acoustography is an ultrasound-based imaging modality that maps the acoustic response of an object induced by acoustic radiation force. The method employs two ultrasound beams with frequencies separated by a small difference, typically in the kilohertz range. Using a 1.75D array transducer, which has multiple rows of elements, offers an advantage over linear array transducers because it allows for focusing in the elevation direction. We have performed simulations of different arrangements for the apertures assigned to the two ultrasound frequencies and evaluated resolution and sidelobe levels for different arrangements. However, the large size of the elements produces grating lobes in the transmitted fields. We will describe metrics for evaluating grating lobe levels compared to the main lobe of the transmitted field as a quantitative approach to optimize aperture design. We present experimental measurements of the transmitted field and compare with simulation results. Finally, we present imaging results from phantom experiments. This beamforming study is directed toward improving vibro-acoustography image formation using a General Electric Vivid 7 scanner for breast and thyroid imaging. [Mayo Clinic and two of the authors have potential financial interest related to devices or technology referenced above. Work supported by NIH Grant No. R21CA121579 and AIUM Award No. AIUM-EER#1.]

4:30

2pBB10. Calculations of intensities for radiation force modeling with the commercial software package FOCUS. Robert J. McGough, Donald J. Vanderlaan, Alexander Dutch (Dept. of Elec. and Comput. Eng., Michigan State Univ., East Lansing, MI), and Matthew W. Urban (Mayo Clinic College of Medicine, Rochester, MN)

Rapid calculations of the intensity are required for models of the radiation force generated by linear ultrasound phased arrays. To determine the computation times and numerical errors produced by two medical ultra-

sound simulation packages, namely, FIELD II and FOCUS, comparisons of the calculated pressures and intensities are performed using a simulated Vermon L5 phased array probe with 128 active elements. In these simulations, the array generates a fixed focus 25 mm from the array for a 5-MHz excitation, and the pressure field is computed on a $133 \times 133 \times 261$ rectilinear grid that is centered at the focus. Half wavelength sampling is employed in all three directions. In FIELD II, this calculation takes approximately 4 h on a 64-bit laptop running Linux, and in FOCUS, the same calculation takes less than 4 min on the same computer. The numerical errors are also much smaller with FOCUS. Details of the simulation approaches employed for this transducer model will be presented, including selection of FIELD II and FOCUS simulation parameters and calculations of the reference pressure field for error evaluations. The impact of numerical errors on shear wave calculations will also be discussed.

4:45

2pBB11. Strategies to improve conformal tumor heating with waveform diversity beamforming. Robert J. McGough and Matthew R. Jennings (Dept. of Elec. and Comput. Eng., Michigan State Univ., East Lansing, MI)

Waveform diversity is a phased array beamforming approach that, when applied to tumor heating with ultrasound, simultaneously maximizes the power at control points in the tumor while minimizing the power deposited in specified sensitive tissues. When waveform diversity calculations are combined with a rank deficient beamforming method, the optimal number of focal patterns is determined, and the optimal array excitations are also obtained. Previous results have shown that waveform diversity combined with mode scanning reduces problems with intervening tissue heating that otherwise occur when standard spot scanning approaches are applied. In an effort to extend these results and achieve a more conformal tumor heat deposition, variable weights will be applied to each of the focal points as a function of distance from the center of the spherical tumor volume. Modified focal point arrangements that decrease the focal point density along the array normal to further reduce normal tissue heating will also be evaluated. Simulations demonstrating the results of these and other modified beamforming strategies will be demonstrated for a large spherical section array using the FOCUS software package.

Session 2pNCa

NOISE-CON and Noise: Construction Noise

Erich Thalheimer, Chair

Parsons Brinckerhoff, 3340 Peachtree Rd., Ste. 2400, Tower Place 100, Atlanta, GA 30326

Contributed Papers

1:00

2pNCa1. Remote construction noise monitoring. Areg Gharabegian (Parsons, 100 W. Walnut St., Pasadena, CA 91124, areg.gharabegian@parsons.com)

Los Angeles was constructing an underground 12 ft in diameter sewage tunnel stretching 25 km. The project includes nine construction access shafts where tunnel boring machines, muck trains, and materials were lowered into the ground. The excavated tunnel material was transported from the tunnel head to the access shafts by muck train, removed by a crane, and carried away via heavy haul trucks. A state of art remote construction noise system was developed and installed near each of the shaft sites, which was collecting noise, audio, and video data 24 h a day. Noise levels were recorded whenever a noise limit threshold is exceeded. Data were then transmitted via modem to a central location where noise specialists review and analyze the data. A portion of these noise limit exceedances were correlated to the audio and video files to determine if the exceedances were due to the construction activities or other non-construction related events. Construction noise at the noise sensitive sites adjacent to each shaft were minimized by constructing sound walls as high as 10 m.

1:15

2pNCa2. Rising to the challenges of noise monitoring around construction sites. Douglas Manvell and Martin Alexander (Briel & Kjaer, Skodsborgevej 307, 2850 Nrum, Denmark, dmanvell@bksv.com)

The construction of infrastructure and new buildings causes changes to the environment of the neighborhood both during and possibly after construction, particularly for major projects. These changes are often significant and result in concern among local population and government and often result in restrictions. To be effective and thus meet the demands on these restrictions, adequate enforcement is needed. Enforcement of noise limit compliance around construction sites needs to efficiently determine noise limit compliance, avoiding false positive exceedances, in order to provide a correct and true picture of the impact of the construction work. This paper describes some of the major challenges faced when designing, deploying, and operating a system to monitor compliance of noise regulations and limits around a construction site. It describes and compares alternative approaches to facing these challenges.

1:30

2pNCa3. Remote construction vibration monitoring at sensitive facilities. Marc Newmark, Jeffrey Zapfe, and Eric Wood (Acentech, Inc., 33 Moulton St., Cambridge, MA 02138, mnewmark@acentech.com)

Construction-related vibrations, while temporary, still have the potential to disrupt vibration- or noise-sensitive operations in adjacent facilities (such as hospitals, laboratories, and performing arts venues). Traditional construction vibration monitors are configured to measure levels corresponding to building damage thresholds. Unfortunately, vibration-sensitive equipment can be adversely affected at levels that are orders of magnitude lower than this. Furthermore, while most construction vibration monitors provide level information at a single dominant frequency, most criteria for sensitive equipment are specified over a range of frequencies. Generally, occupants of buildings near construction sites prefer to continue operations during the construction process. To do this, however, they need to know how the construction-related vibrations compare to their instrument criteria. To help

provide this information, we have developed a remote monitoring system that provides real-time vibration and/or noise spectra in the sensitive facility. These spectra can be directly compared to instrument criteria with appropriate alarm notifications as needed. In this paper, we will present three case studies that show the evolution of this remote monitoring capability and discuss how the information has been used by the institutions where it has been installed.

1:45

2pNCa4. An approach to noise impact assessment for long-term construction. Frank Babic (AECOM, 105 Commerce Valey Rd., West, Markham, ON L3T 7W3 Canada, frank.babic@aecom.com)

Noise generated by construction is typically characterized as a temporary noise source. Due to its temporary nature, construction noise can generally be tolerated by the community at large more readily than long-term noise sources (for example, from industry). For some projects, construction may extend beyond a temporary period to become a more long-term noise problem—especially when the construction activity can last months or even years. When an extended construction period occurs, the community may be less tolerant of the noise, and the assessment of its impact on the community is more aptly characterized as a long-term noise source, similar to that of industry. This paper proposes an approach to assessing long-term construction projects, where a “worst-case” scenario can be modeled with utility factors, duty cycles, and barrier noise controls. From this modeling, a zone of influence can be determined which identifies the extent of the noise impact (i.e., potential complaints generated by nuisance from construction noise). Receptors within this zone of influence (including residences, hospitals, churches, etc.) can then be identified as part of the environmental noise impact assessment for long-term construction activity.

2:00

2pNCa5. Estimating construction noise using the Environmental Science Research Institutes geographic information systems program. Scott Noel (Parsons Brinckerhoff, Inc., 400 SW 6th Ave., Ste. 802, Portland, OR 97219, noels@pbworld.com)

ArcGIS is a series of Geographic Information Systems (GIS) programs and tools developed by the Environmental Science Research Institute (ESRI). ArcGIS has many tools built in, however, no tools to estimate construction noise levels for either point sources or for a project construction footprint. PYTHON is a programming language that is becoming the preferred programming language for ESRI programs. Parsons Brinckerhoff, Inc. has modified the buffer tool in ArcGIS to estimate noise levels from a variety of construction equipment. The tool allows the user to choose what typical construction equipment will be used on their project. The tool then creates a noise contour showing what noise levels could be expected at certain distances from the project area or point source. The tool is useful for estimating construction noise at various distances to sensitive receptors. The tool also helps in graphics creation to aid in summarizing noise impacts. Parsons Brinckerhoff, Inc. has used this tool on transportation projects but it is applicable to other projects as well.

2:15

2pNCa6. Construction noise impact prediction using non-virtual simulation. Rob Greene (Parsons Brinckerhoff, 505 S. Main St., Ste. 900 Orange, CA 92868, greener@pbworld.com)

This paper presents a rarely used approach to predicting potential noise impacts from major infrastructure construction. The method includes the actual propagation of spectrally shaped pink noise into a community from locations within the proposed project's construction site, coupled with contemporaneous noise measurements at multiple in-community locations. Pre-test source calibration measurements were performed and ambient noise levels were measured in the surrounding community during pre- and post-test periods. A Public Information program was part of the environmental assessment process. Yes, this novel/brave/are you crazy? approach was extensively covered in the local press. The community reaction to the methodology, the noise control mitigation that resulted from the test, the success of the construction, and lessons learned will be presented.

2:30

2pNCa7. Using the Federal Highway Administration Roadway Construction Noise Model for the Jerome Park reservoir construction project. Gabriella Yanez-Urbe and Erich Thalheimer (Parsons Brinckerhoff, 3340 Peachtree Rd, Ste. 2400, Tower Pl. 100, Atlanta, GA 30326, yanezuribe@pbworld.com)

The new Jerome Park Reservoir in the Bronx, NY, is a critical water supply source for the residents and businesses of New York city. This paper will describe the noise modeling work done in support of the project as well as an odd bug found in the FHWA's RCNM model. Noise associated with the various construction phases and equipment for the project were analyzed to evaluate the potential loudest hour during each month of work for the project's 50-month duration. Noise analyses were performed for expected weekday work conditions and Saturday work conditions at several receptor locations surrounding the reservoir using the FHWA's Roadway Construction Noise Model (RCNM), Version 1.0. During the analysis, an error was found with the output function of RCNM when exporting results for multiple receptors into CSV and TXT output files. The total L_{eq} and L_{max} levels were being reported as zeros even though the correct total noise levels were displayed on screen. This error was submitted to FHWA for review, and as a result, RCNM Version 1.01 was soon released addressing the problem. More importantly, the project was performed to the client's satisfaction and has allowed the necessary construction work to proceed.

2:45

2pNCa8. How good is the Roadway Construction Noise Model for predicting highway construction noise impacts? Marc Wallace (Tech Environ., Inc., 303 Wyman St., Ste. 295, Waltham, MA 02451, mwallace@techenv.com)

The reconstruction of the I-93/Route 110/Route 113 in Methuen, MA will be conducted over a 4-year period. As part of the project, a new I-93 bridge superstructure will need to be built. This bridge work will consist of constructing the center portion of new superstructure, which will include pile driving. The work will require shifting I-93 traffic lanes several times to accommodate bridge construction over a several months. The design engineers are also considering constructing the superstructure over a weekend by shutting down I-93 and re-routing over local roads to reduce construction impacts. The Federal Highway Administration's (FHWA) Roadway Construction Noise Model (RCNM) was used to assess potential construction noise impacts on nearby residential neighborhoods. The RCNM is a good screening tool to conservatively assess potential construction noise impacts. There are more sophisticated models, such as the Cadna A comprehensive three-dimensional (ISO 9613-2) model that takes into account topographic and atmospheric conditions. The goals of this paper will be to compare the RCNM and Cadna A model results to assess the accuracy of the RCNM model and to also assess the potential benefits or impacts of constructing the superstructure over a shorter period of time using the Cadna A model.

3:00

2pNCa9. Proactive regulation engenders creative innovation: Quietening the jack hammer. Eric Zwerling (Dept. of Environ. Sci., Rutgers Noise Tech. Assistance Ctr., 14 College Farm Rd., NB, NJ 08901, Zwerling@rutgers.edu), Charles Shamoon (New York City Dept. of Environ. Protection), and Stephen Szulecki (The Noise Consultancy, LLC)

In 2002, the New York City Department of Environmental Protection undertook to completely revise the city's Noise Code; it went into effect in 2007. The Mayor's Office made the reduction in construction noise a priority. Incentives were provided for the application of best management practices and the use of quieter equipment. DEP staff proposed the development of a remediation strategy for pneumatic pavement breakers. Working with KeySpan Energy (now National Grid), they approached Zo-Air Company, an industrial tool distributor, who developed the "No Racket Jacket." The key to developing a successful engineered solution is understanding the noise source. Most of the noise emissions from a pavement breaker come from internal components, not the tool bit hitting the pavement. Enclosing the source significantly reduces the noise emissions. The jacket *al.so* serves to insulate the operator from the exhaust air of pneumatic units, and the oil often spewed onto their legs. This paper discusses the development process which included four prototypes, and the results, with data from source sound level measurements and operator dosimetry. Three scenarios were tested: unremediated pneumatic hammer, pneumatic hammer with the "No Racket Jacket," and an electric pavement breaker.

3:15—3:30 Break

3:30

2pNCa10. Designing partial enclosures to minimize airborne sound transmission. D. W. Herrin, A.E. Carter (Dept. of Mech. Eng., Univ. of Kentucky, 151 RGAN Bldg., Lexington, KY 40506-0503, dherrin@engr.uky.edu), J. Shi, and D. C. Copley (Caterpillar, Inc.)

Enclosures are a very common way to reduce noise emissions from machinery. However, enclosures display complex acoustic behavior that is difficult to predict. Boundary element simulation was used to better understand the airborne transmission path for a partial enclosure. Sources considered included a point source and a diesel engine. Insertion loss was used as the performance measure to evaluate the effect of several different design considerations. Results indicate that the most important factors affecting enclosure performance are the opening size, amount of absorption, and the source-to-opening distance. It is notable that insertion loss was relatively insensitive to enclosure or source size.

3:45

2pNCa11. New York City western rail yard construction noise study. Erich Thalheimer (Parsons Brinckerhoff, 75 Arlington St., Boston, MA 02111, thalheimer@pbworld.com)

This paper will describe the technical approach, regulatory environment, and outcome of a construction noise study performed on behalf of a \$2.5 billion private development to be built over the Western Rail Yards in Manhattan. Community noise modeling for the 9 year project was performed in accordance with New York City Environmental Quality Review (CEQR) Manual requirements using the CADNA-A and FHWA TNM models. The on-site noise analysis accounted for the staggered schedule of excavation, foundation laying, erection, and finishing of eight high rise residential and commercial buildings. Construction equipment source emissions used in the CADNA-A model were taken from the FHWA Roadway Construction Noise Model (RCNM) and the NYC DEP Construction Noise Regulations (Local Law 113, Chap. 28). Project-related trucking (mobile) noise was evaluated using the TNM model. Seventeen multi-story noise receptor locations surrounding the development site were evaluated with ambient noise measurements and predictive noise modeling. A construction vibration assessment for the elevated High Line park was also conducted. The results indicated that noise exceedance conditions were expected from pile driving, hoe ramming, and jackhammering operations during daytime, nighttime, and weekend periods, so noise mitigation measures were developed and incorporated into the project's construction permit.

4:00

2pNCa12. Report on mitigation of nighttime construction noise. Cliff Schexnayder (Arizona State Univ., P.O. Box 6700, Chandler, AZ 85246, cliff.s@asu.edu) and Jennifer S. Shane (Iowa State Univ.)

Nationally, the amount of road work being done at night appears to be significant. Data taken during the summer of 2001 suggests that one-third of the work zones present on National Highway Systems roadways were active primarily at night. Night work can have impacts on adjacent neighborhoods and businesses that would not be experienced or as disturbing if conducted during the daytime. Construction equipment is a major noise and nuisance generator on nearly all nighttime construction projects. This paper will report on an ongoing Transportation Research Board study that is seeking to identify best practices in terms of source control, path control, and receptor control of noise on transportation projects. Individual DOTs having specific experience and contractors are being queried about their experiences in mitigating nighttime construction nuisances.

4:15

2pNCa13. Noise control evaluation for a concrete batch plant. Robert D. O'Neal (Epsilon Assoc., Inc., 3 Clock Tower Pl., Ste. 250, Maynard, MA 01754, roneal@epsilonassociates.com)

As a result of neighborhood complaints, an Administrative Consent Order was issued to a concrete batch plant for violating the Massachusetts noise policy. The fact that the batch plant had been in business long before the residential subdivision was built did not factor in the State's decision. A detailed noise control study was conducted for the facility. All mobile and stationary sources of noise were measured in addition to measurements at

the receiving property. The major sources included mixer trucks receiving concrete and prepping their load, concrete trucks climbing a hill, batch plant dust collector fans, and bulk cement truck deliveries. The measurements were analyzed, and a series of noise control measures was evaluated and recommended, some of which were implemented by the facility. Several rounds of controls and associated sound level testing were conducted to determine if the batch plant met the State noise requirements, and compliance was eventually achieved. This paper will discuss the key sources of noise at the facility and the techniques used to reduce noise at the source.

4:30

2pNCa14. Road construction noise inside a radio station. Sean Connolly (Big Sky Acoust., LLC, P.O. Box 27, Helena, MT 59624, sean@bigskyacoustics.com)

Construction noise limits are typically presented as noise levels not to be exceeded at the exterior of a building or a property line. However, when a building houses noise-sensitive uses, the possible construction noise impacts must be assessed inside the building. In 2004, a contentious legal dispute erupted between a radio station owner and the state highway department over the reconstruction of the highway directly adjacent to the station. The owner claimed that excessive road construction noise would disrupt live broadcasts from his studio. Therefore, the construction noise levels needed to be assessed inside the studio during the entire construction process, but legal maneuvering severely limited access to the station. This paper discusses the criteria and methods used to assess construction noise impacts inside the studio, the relationship between the measured noise levels inside and outside of the studio during various phases of road construction, and of course, if the construction noise interfered with radio broadcasts.

2p TUE. PM

TUESDAY AFTERNOON, 20 APRIL 2010

GRAND BALLROOM III/IV, 1:00 TO 3:30 P.M.

Session 2pNCb

NOISE-CON and Noise: Information Technology Noise

David Nelson, Cochair

Nelson Acoustics, P.O. Box 879, Elgin, TX 78621

Terry W. Baird, Cochair

Hewlett-Packard Company, 11311 Chinden Blvd., Boise, ID 83714

Marco Beltman, Cochair

Intel Corp., 2111 NE 25th Ave., Hillsboro, OR 97124

Contributed Papers

1:00

2pNCb1. German information technology-equipment noise categorization concept: Study on several data projectors. Gregor Feneberg (Mueller-BBM GmbH, Robert-Koch-Str. 11, 82152 Planegg, Germany, gregor.feneberg@muellerbbm.de)

An investigation project tendered by the German Federal Institute for Occupational Safety and Health (BAuA), with a planned duration of 1 1/2 years, has recently been awarded to Mueller-BBM. The existing concept of the quality classification of household appliances, where the energy efficiency categories A-F each represents a determined range of power consumption, is to be transferred to a system for the categorization of IT-equipment with regard to its noise emission. Currently it is almost impossible for a layperson to compare noise declarations provided by manufacturers even though there are standardized and very detailed specifications in ISO 9296 for the measurement and declaration of the noise emission of IT-equipment. One of the central issues of the project is the consolidation of

existing procedures for noise emission measurements on IT-equipment with consideration of the impact of psychoacoustic parameters. A categorization proposal based on ISO 7779 will be presented. In order to check the necessity of additional psychoacoustic metrics like tonality and transient loudness, 11 data projectors were measured according to the BAuA proposal. In addition, the annoyance of the operational noise was evaluated by a panel test. The results will be compared and contrasted with the current ISO /ECMA standard measurements.

1:15

2pNCb2. Development of the standard ISO 7779 3rd edition: Revised and new contents. Ikuo Kimizuka (IBM Japan, Ltd., 1623-14, Shimotsuruma, Yamato-shi, Kanagawa-ken 242-8502, Japan, kmzk@jp.ibm.com)

ISO 7779 is a noise test code for airborne noise emission of information technology and telecommunications equipment (so called, ITTE). Since its 1st edition of 1988, ISO 7779 has stated, in detail, the installation and op-

erating conditions for specific equipment categories. In nature of ITTE, however, changing of the product characteristics is much faster than the pace of standard document updates. For instance, most models of copy machines and printers (except some dedicated high speed machines) are almost being combined into new category, so called, multi-functional devices. To catch such rapid changes up timely, it becomes impractical to state specific condition within this International standard. Therefore, ISO/TC43/SC1/WG23 which is responsible for ISO 7779 decided to propose new framework so that 3rd edition should refer to ECMA-74, annex C, which has been the industrial counterpart and also pilot standard of ISO 7779. This paper introduces background of key changes from 2nd edition and new technical contents for ITTE noise emission measurement methodologies.

1:30

2pNCb3. Printer impulsive noise metrics: Sensitivity and repeatability analysis. Terry Baird and Katy Ferguson (Hewlett-Packard Comp., 11311 Chinden Blvd., Boise, ID 83714, terry.baird@hp.com)

The major cause of annoyance from impulsive noise is the impulsive amplitude. Many methods can characterize impulsive amplitude, such as time-averaged measures, peak amplitudes or real-time levels. For example, the "impulsive noise index" described by ISO 11201 or the "impulsive parameter" described in ISO 7779 are time-averaged metrics. Real-time metrics might include peak sound pressure level or peak loudness in the time domain. Impulsive metrics must be evaluated for effectiveness. That is, the metric must agree with the perceived impulsive amplitude. Impulsive metrics must also be practical. They must be insensitive to sample-to-sample variation while still providing the granularity between passing and failing results. This paper focuses on evaluating the sensitivity, measurement repeatability, and result granularity to determine the practicality of various effective impulsive noise metrics.

1:45

2pNCb4. Development of schemes to gauge the annoyance of time-varying sound. Menachem Rafaelof (Seagate Technol., 389 Disc Dr. Longmont, CO 80503, menachem.rafaelof@seagate.com)

Hard disk drives (HDDs) nowadays have use in many diverse applications. Examples include various types of consumer electronic devices that operate in extremely quiet environments and personal computers that integrate the HDD and other peripherals within the display very close to the user. Higher exposure to the HDD sound, either due to the lack of masking sound or close proximity to the user, require establishment of a scheme to identify features within the sound to decide on its acceptability. This paper describes an effort aimed at developing a scheme for identification of attention-grabbing or annoying features within time-varying sound that would point to its unacceptability by a typical user. In this first part, sample sounds were examined based on their perceived strength (loudness), duration, repetition rate, and higher order statistics to acceptability based on a jury study.

2:00

2pNCb5. Impact of personal video recorder/digital video recorder on hard disk drive acoustic requirements. Dave Ali (Western Digital, 5863 Rue Ferrari, San Jose, CA 95138, dave.ali@wdc.com)

In recent years the impact of the personal video recorder/digital video recorder market has grown substantially. This growth has a direct impact on the acoustic test laboratories that make sure the systems meet the manufacturer's specification for noise emissions. To date the standards have not addressed this growth and subsequently have not provided guidance as to standardized methods of test for these systems. This paper will show the impact of the market and provide a proposed addition to the appropriate acoustic test standards.

2:15

2pNCb6. Fan noise prediction for design. Charles Oppenheimer (Hewlett-Packard, 18110 SE 34th St., Vancouver, WA 98683, charles.oppenheimer@hp.com)

This paper develops a fan noise prediction model intended for use in the design of systems for cooling and drying. Airflow through a system is described by a dimensionless resistance parameter. Fan noise is predicted by a

scaling relationship obtained from a dimensional solution of Lighthill's acoustic analogy. The scaling relationship contains unknown proportionality factors, which are determined empirically and compared for axial fans and centrifugal blowers. The dependence of fan and blower noise spectra on non-dimensional scaling parameters is also investigated.

2:30

2pNCb7. Constant acoustic fan curves for 92 mm axial fans. Willem Beltman (Intel Labs, Intel Corp., M/S JF2-86 2111 NE, 25th Ave., Hillsboro, OR 97124, willem.m.beltman@intel.com)

The noise emission of fans changes as a function of the operating point. Airflow and acoustic fan plenum experiments can be used to characterize these effects to estimate noise under installed conditions. Recently, the concept of iso-acoustic fan curves was introduced by various authors, and results were presented for small radial blowers and some axial fans. The current paper presents an analysis of the iso-acoustic fan curves for an extensive set of 92 mm axial fans. An analysis is presented of the airflow and acoustic performance of these fans, and expressions for the iso-acoustic fan curves are derived and presented. This allows a direct comparison between the various fans based on operating point conditions.

2:45

2pNCb8. Aero-acoustic simulations of small scale electronic cooling blowers. Jessica Gullbrand and Willem M Beltman (Intel Labs, Intel Corp., 2111 NE 25th Ave., MS JF2-86, Hillsboro, OR 97124, jessica.gullbrand@intel.com)

The noise emission of small radial blowers is important for the cooling of notebook systems. It has been demonstrated that airflow simulations for these radial blowers can be performed to predict the blower performance, even in the presence of inlet restrictions. The current paper presents an extension of this work to include predictions of the radiated acoustic noise. The calculated blower flow field is used as an input for a second stage aero-acoustic prediction. This paper outlines two methods. The first method uses a free field approach based on the Ffowcs-Williams and Hawkings analogy. The second approach constructs a rotating fan source, the magnitude of which is determined by integrating the pressure across a number of blade segments. The radiated field is then determined in a boundary element calculation, both for a free field and in the presence of the fan enclosure. Results for both methods are presented and compared to experimental data.

3:00

2pNCb9. A relation between flow, noise, and system impedance for notebook computers. Eric Baugh (Intel Corp., 15400 NW Greenbrier Pkwy., M/S CO5-166 Beaverton, OR 97006, eric.baugh@intel.com)

Performance of notebook computers is strongly limited by the ability to reject heat from the system. In a typical mainstream design, the airflow driven by the internal fan is responsible for about 80% of the maximum cooling capability, and this airflow impacts all the internal components (CPU, memory, radios, etc.) as well as the top and bottom skin temperatures. Fan flow, in turn, is limited by acoustics. It has been shown that the fan installation conditions inside a notebook computer have an adverse effect on both flow rate and noise. The market trend is toward thinner systems, which generally have a higher system impedance. As impedance increases, system airflow must decrease to maintain a fixed acoustic limit, holding all other variables constant. In order to demonstrate the effect of impedance, this paper describes a simple, two parameter model relating the flow rate in a notebook computer at a given operator position sound pressure level to the impedance index of the system. This relation is shown to reasonably approximate the behavior of about 20 different notebook computers.

3:15

2pNCb10. The dB[EQL]: An alternative sound pressure weighting according to the equal loudness contours of the international standard ISO 226-2003. Wade Bray (HEAD Acoust., Inc., 6964 Kensington Rd., Brighton, MI 48116, wbray@headacoustics.com)

The conventional spectral weightings applicable to sound pressure [dB(A), etc.] are fixed in spectral shape and intended for use over certain ranges

of unweighted sound pressure level: for example, the *A*-weighting from threshold of hearing up to 65 dB(SPL). In general, these weightings' spectral shapes match the sensitivity of hearing as a function of frequency within "use-level" ranges, although the strong effect of the cavum conchae resonance of the ear, apparent in the equal-loudness contours of ISO 226, is not considered except in the rarely Used *D*-weighting. Particularly for sounds with tonal content and within the general level range of the *A*-weighting, the authors propose a new spectral weighting assembled from the Phon values

of the complete set of equal-loudness contours calculated for each frequency within the human auditory range. It will be shown that although giving values similar to those of the *A*-weighting, the dB[EQL] or equal-loudness weighting is situation dependent rather than fixed, and better represents subjective impressions at all frequencies. Although based on perceived loudnesses, the dB[EQL] sound pressure weighting is not a specific loudness measurement, does not consider critical band formation, and does not yield masking or psychoacoustic loudness data.

TUESDAY AFTERNOON, 20 APRIL 2010

GRAND BALLROOM III/IV, 3:45 TO 5:00 P.M.

Session 2pNCc

NOISE-CON and Noise: Consumer and Industrial Product Noise

Charles S. Hayden, Chair

NIOSH, 4676 Columbia Pkwy., Cincinnati, OH 45226-1998

Contributed Papers

3:45

2pNCc1. Geothermal heat pump noise control (cont.): Its all in the details. Jeffrey Fullerton (Acentech, Inc., 33 Moulton St. Cambridge, MA 02138, jfullerton@acentech.com)

The installation of a geothermal heat pump in the basement of a 1900 era home introduced significant noise and vibration to a bedroom above, which was discussed in detail in a previous conference. After mitigating the noise and vibration using piping isolation products, lower levels of noise and vibration still remained. This paper will discuss pursuits that were implemented and the very effective resolution for mitigating the noise and vibration that was generated by the heat pump.

4:00

2pNCc2. Improving screw compressor housing design using simulation. D. W. Herrin, Z. Tao (Dept. of Mech. Eng., Univ. of Kentucky, 151 RGAN Bldg., Lexington, KY 40506-0503, dherrin@engr.uky.edu), A. J. Graybill, J.E. Bender, P.C. Marks, and C. Eichelberger (JCI Bldg. Efficiency)

The sound radiation from a rotary screw compressor was simulated using structural finite element and acoustic boundary element analysis. The modes of the housing were measured with and without the screws. Finite element modal analysis results were compared to the experimental modes with good agreement. Results demonstrated that the screws affected the bending modes but not the more important cylinder modes of the compressor housing. The sound radiation from the compressor housing was then predicted using acoustic boundary element analysis. The results indicated that the second and fourth pumping frequencies were close to two structural modes. Subsequent simulations investigated the effect of design changes on the housing. These changes included using a thicker cylinder and adding ribs.

4:15

2pNCc3. Vibro-acoustic model of an enclosed electric power generator. David C. Copley and Harvind Raman (Caterpillar Inc., Tech. Ctr. Bldg. E, P.O. Box 1875, Peoria, IL 61656, copley_david_c@cat.com)

A vibro-acoustic model of an enclosed diesel-powered electric generator was built using commercial vibro-acoustics software (primarily SEA) and

empirical methods. The vibro-acoustic sources, including the engine, generator, cooling fan, and exhaust were characterized experimentally for several different operating conditions. The vibro-acoustic performance of the enclosure and sound suppression features were modeled and experimentally validated. Good agreement was achieved between experimental and simulated results.

4:30

2pNCc4. Passive noise control methods applied to a household icemaker. Richard Ruhala (Southern Polytechnic State Univ., 5735 Forest Ave., Evansville, IN 47712, rruhala@usi.edu)

Two icemakers were provided to the engineering program at the University of Southern Indiana for educational testing and development. Three students worked sequentially over 1 year to quantify and localize the noise sources using sound pressure and sound intensity methods, and then implement various design modifications in order to reduce the overall sound power by a goal of 3 dBA. Noise reduction was achieved during some machine operating cycles, although the overall noise reduction goal was not obtained. The students did learn one important finding in noise control engineering: that reducing the noise of a well developed and quiet product is a difficult task that requires expertise in noise assessment and control. This paper will also focus on findings from these studies that are pertinent to the development of quieter refrigerators, freezers, and icemakers.

4:45

2pNCc5. Acoustics of energy efficient hand dryers: Is this progress? Jeffrey Fullerton and Gladys Unger (Acentech, Inc., 33 Moulton St., Cambridge, MA 02138, jfullerton@acentech.com)

Recently there have been innovations in the technology of bathroom hand dryers with a particular focus on energy efficiency. The results have included some rather interesting methods for drying wet hands in different ways than previous generations of hand dryer models. The focus of this study will be to compare the sound levels from these different energy efficient models with previous models that have been used for years. The overall sound level will be compared to the rated efficiency of each unit.

Session 2pNCd

NOISE-CON, Noise, and Physical Acoustics: Aircraft Exterior Noise

Eric R. Boeker, Chair

USDOT Volpe NTSC, 55 Broadway, Cambridge, MA 02142

Contributed Papers

1:00

2pNCd1. FAA noise impacts research roadmap. Raquel Girvin (FAA, 800 Independence Ave., SW Washington, DC 20591, raquel.girvin@faa.gov)

The Federal Aviation Administration's (FAA) continuing mission is to provide the safest, most efficient aerospace system in the world. Improving efficiency through airspace redesign, airport capacity expansion, and other initiatives of the FAA Next Generation Air Transportation System (NextGen) may be hampered without an aggressive program to address the environmental consequences of aviation. FAA's work toward mitigating such environmental consequences includes research to advance the state of knowledge on noise impact analysis. This paper describes the status of FAA's ongoing research efforts and the comprehensive research roadmap addressing critical noise impacts research needs.

1:15

2pNCd2. Overview of aircraft en-route noise prediction. Bill (Hua) He (Federal Aviation Administration, Office of Environment and Energy (AEE) —Noise Div., 800 Independence Ave., NW, Bldg. 10A, 900West Washington, DC 20591, hua.he@faa.gov) and Eric Boeker (US Dept. of Transportation Volpe Ctr.)

Community noise near airports is a major aviation noise concern, prompting the development of various modeling tools to predict the noise changes due to airport expansion and fleet changes. Once aircraft leave the terminal area and reach high altitudes, the aircraft noise received on the ground is assumed to be insignificant relative to urban ambient noise and is often neglected. However, this might not be the case, under either of the following conditions: (1) when aircraft fly over areas with very low ambient noise levels, such as in wilderness areas, or (2) when an aircraft is powered by advanced propulsion systems that produces high noise levels. There is an increasing need to address both of these issues. Over the past 2 decades, some limited noise modeling capabilities and related research have begun to quantify the high-altitude aircraft noise or "en-route" noise. The purpose of this paper is to review and summarize related noise prediction research conducted through direct technical exchange with researchers, some simple data analysis as well as through a literature review. The authors will draw on experience in developing current aircraft community noise tools and formulate key future research questions and options.

1:30

2pNCd3. Application of atmospheric absorption models for aircraft enroute noise. Kieran Poulain, Victor Sparrow, Kenneth Brentner (201 Appl. Sci. Bldg., Univ. Park, PA 16802, kxp252@psu.edu), and Louis Sutherland (Los Angeles, CA)

Propagation of aircraft enroute noise for flights above 18 000 ft above ground level (AGL) (5.49 km AGL) is important for estimating noise impact in U.S. National Parks and other quiet areas. One key to the accurate estimation of noise impact is atmospheric absorption. In our atmosphere, it turns out that the absorption coefficient is dependent on altitude as well as frequency. The altitude dependent parameters include mean pressure, temperature, and relative humidity as well as the concentration of molecular species. This altitude effect can be important for certain frequency bands, and this point will be emphasized in this paper. In addition, an improved atmospheric absorption model for high altitudes of Sutherland and Bass [J. Acoust. Soc. Am. **115**, 1012–1032 (2004)] shows slightly different values of atmospheric absorption in important frequency bands for aviation noise at altitude compared to the typical ANSI Standard S1.26-1995 (R 2009). For

certain situations, the updated atmospheric absorption coefficient could produce different enroute noise predictions on the ground. [Work supported by FAA/NASA/Transport-Canada PARTNER Center of Excellence.]

1:45

2pNCd4. Modeling propagation of aviation noise under range-dependent conditions, using a hybrid parabolic equation-fast field program method. Joyce Rosenbaum, Anthony Atchley, and Victor Sparrow (Graduate Program in Acoust., The Penn State Univ., 202 Appl. Sci. Bldg., University Park, PA 16802, jer262@psu.edu)

Based on the Parabolic equation method, a model for sound propagation over uneven terrain, with mixed ground impedance and range-dependent meteorological conditions, has been developed to improve prediction of the impact of aviation noise on communities. Intended to enhance the noise prediction capabilities for the Federal Aviation Administration's Aviation Environmental Design Tool in support of the Next Generation Air Transportation System (NextGen), the model uses a hybrid parabolic equation-fast field program approach for computing aviation noise contours. These methods ensure that the low-frequency content, a factor in community impact, is propagated accurately. The model can accommodate a broadband, moving sound source, traveling along a user-specified path, over three-dimensional terrain. A test case, designed to represent simple but realistic propagation conditions, is described, and results for a source traveling along a simple flight path are presented and discussed.

2:00

2pNCd5. Acoustic repropagation technique and practical source characterization for simulation noise model databases. Christopher Hobbs and Juliet Page (Wyle Labs., 241 18th St. South, Ste. 701, Arlington, VA 22202, chris.hobbs@wyle.com)

By utilizing the acoustic repropagation technique in conjunction with an open-source, platform-independent database, a robust noise source can be constructed that is well suited for use in simulation noise models. This paper describes a widely applicable methodology for creating a three dimensional noise source from measurements that can account for source directivity. It will be shown how the resulting data, known as noise spheres, may contain a variety of noise metrics representing the source at a fixed distance for all spherical angles. Such spheres can be combined to represent complex noise sources and depending on their metric content can be used to accurately predict the near and far field environment about the source. A case utilizing narrow band spectra with phase information will be explored. It will be shown how a collection of spheres can be used to represent different operating states of a source and how the use of an independent parameter such as speed or power can be used to interpolate non-exact states in a noise model. By way of example, an approach to incorporating non-linear sound propagation from a high amplitude noise source is explored in conjunction with the advanced acoustic model.

2:15

2pNCd6. Modeling of noise from aircraft taxiing operations. Juliet Page and Christopher Hobbs (Wyle Labs., 241 Eighteenth St. South, Ste. #701, Arlington, VA 22202, juliet.page@wyle.com)

This paper documents a study performed for the Airport Cooperative Research Program, investigating noise from aircraft taxi operations. A holistic approach is utilized, applying an understanding of the operations, the detailed aircraft motions, and the acoustic sources. A series of acoustic sensi-

tivity studies was conducted in order to develop a physical understanding of the relative importance of the various modeling elements within the framework of AEDT. The study is based on decoupling the modeling of taxi noise into the following three areas and exercising each element independently. (1) Engine source noise (level, spectra, and directivity); (2) aircraft movements and operating states (location, duration, and power setting); (3) environment/propagation (lateral attenuation, terrain, shielding, and ground impedance). Aircraft taxi noise sensitivity studies exercised each element independently. Limited opportunistic commercial aircraft taxi operation acoustic measurements were conducted. Independent taxi flight data recorder information was queried to determine statistical engine and aircraft operational parameters. The sensitivity studies reveal that the primary weakness for taxi noise modeling is related to item (#1), engine source noise modeling. This paper will discuss the sensitivity study findings and present an approach for development of a nominal taxi state noise-power-distance, spectral class, and directivity database for AEDT.

2:30

2pNCd7. Investigation of the use of lower frequency acoustic data in helicopter environmental noise modeling. Eric Boeker and Noah Schulz (USDOT Volpe Natl. Transportation Systems Ctr., 55 Broadway RVT-41 Cambridge, MA 02142, eric.boeker@dot.gov)

The current Federal Aviation Administration (FAA) aircraft noise modeling tools, Aviation Environmental Design Tool (AEDT) and Integrated Noise Model (INM), do not consider noise below 50 Hz in their computations. This paper describes a preliminary study to determine the effect of including low-frequency data on the accuracy of AEDT/INM results. Expanded aircraft noise spectra containing one-third octave band data to 12.5 Hz were analyzed using methods adapted from AEDT/INM. Results

from expanded spectral data are compared with results from the historical AEDT/INM spectral data (one-third octave band data from 50 Hz to 10 kHz). This comparison showed a range of differences, from increases in overall un-weighted sound pressure levels, to negligible changes in A-weighted and time audible metrics. These changes may be particularly important for helicopters, with dominant low-frequency rotor noise below 50 Hz. Following the comparison, the potential implementation of expanded spectral data in AEDT/INM is discussed.

2:45

2pNCd8. Examination of the low frequency limit for helicopter noise data in aviation environmental design tool and integrated noise model. Eric Boeker and Noah Schulz (USDOT Volpe Natl. Transportation Systems Ctr., 55 Broadway RVT-41, Cambridge, MA 02142, eric.boeker@dot.gov)

The current Federal Aviation Administration (FAA) aircraft noise modeling tools, Aviation Environmental Design Tool (AEDT) and Integrated Noise Model (INM), do not consider noise below 50 Hz in their computations. This paper describes a preliminary study to determine the effect of including low-frequency data on the accuracy of AEDT/INM results. Expanded aircraft noise spectra containing one-third octave band data to 12.5 Hz were analyzed using methods adapted from AEDT/INM. Results from expanded spectral data are compared with results from the historical AEDT/INM spectral data (one-third octave band data from 50 Hz to 10 kHz). This comparison showed a range of differences, from increases in overall un-weighted sound pressure levels, to negligible changes in A-weighted and time audible metrics. These changes may be particularly important for helicopters, with dominant low-frequency rotor noise below 50 Hz. Following the comparison, the potential implementation of expanded spectral data in AEDT/INM is discussed.

TUESDAY AFTERNOON, 20 APRIL 2010

GRAND BALLROOM VII/VIII, 3:15 TO 4:45 P.M.

Session 2pNCe

NOISE-CON, Noise, and Structural Acoustics and Vibration: Vibrations and Structureborne Noise in Buildings

Jeffrey P. Kwoikoski, Chair

D. L. Adams Assoc., Inc., 1701 Boulder St., Denver, CO 80211

Contributed Papers

3:15

2pNCe1. Flanking noise transmission in condominiums via steel roof deck. Jim X. Borzym (Borzym Acoust. Consulting & Eng., LLC, 2221 Columbine Ave., Boulder, CO 80302, acoustics@columbine.net)

Noise was found to transmit horizontally via a structure borne path. A lightweight roof consisting of ribbed steel deck spanned two adjacent loft-style condominium residential spaces. This noise transmission path significantly flanked the demising partition wall with known high-transmission loss. Two remedies were attempted to reduce transmission, by adding concrete on top of the roof deck. One remedy was successful.

3:30

2pNCe2. Flanking paths associated with exposed roof decks in condominiums. Jeff Kwoikoski (D. L. Adams Assoc., Inc., 1701 Boulder St., Denver, CO 80211, jkwoikoski@dlaa.com)

An award winning residential/commercial condominium project featured two-story loft condominiums with an industrial look. The building was constructed with steel structure, exposed concrete on metal deck floors, and an exposed metal deck roof. Condominiums were separated by gypsum board on metal stud walls. A number of owners in the building complained of poor sound isolation from adjacent units. Field tests showed that sound isolation of the walls fell below expectations for the design. Subsequent investigation

showed that a variety of structureborne and airborne flanking paths contributed to poor sound isolation. Improving sound isolation required a combination of conventional and creative methods.

3:45

2pNCe3. Concrete pier foundation and building design for electron microscope installations with vibration disturbances from train, truck, and bus traffic. Chad Himmel (1705 W. Koenig Ln., Austin, TX 78756, himmel@jeacoustics.com)

A university microscopy and imaging center housing nine instruments including scanning electron microscopes and transmission electron microscopes was proposed for a site near an active railroad and roadways. Low-frequency sound and vibration from trains, trucks, and buses can affect the performance and use of electron microscopes, by disturbing the stability and resolution of the specimen's image. On-site observations and spectrum measurements were conducted to determine the ambient and transient ground borne vibration conditions. Measurements showed that disturbances from trains as well as bus and truck traffic on campus roads significantly exceeded allowable floor vibration tolerances for many of the instruments. Soils exhibited response to disturbances at dominant frequencies around 5 and 10 Hz. This paper addresses the potential for concrete pier-supported floor vibration and structural resonance contributions to ground borne vibration levels and solutions proposed to mitigate the disturbances for sensitive

instrument installations. Graphic diagrams of structural design concepts and results from on-site measurements prior to construction will be discussed. The designs and effectiveness of the microscope rooms, structural treatments, and actively damped, pneumatic vibration isolation systems are discussed, along with post-construction vibration measurement results and photographs.

4:00

2pNCe4. Tri-axial measurement of roadway vibration in multiple research buildings located throughout an urban college campus. Scott Harvey and Josh Curley (Phoenix Noise & Vib., 5216 Chairmans Ct., Ste. 107, Frederick, Maryland 21701, sharvey@phoenixnv.com)

Plans are currently underway to run a mass transit rail line through the center of a major university. Concerns over this significant change to the school's infrastructure were raised by members of the university's faculty, particularly professors questioning whether their research projects and vibration sensitive equipment will be disturbed from vibration generated by all the new train passbys. A study was arranged to investigate the campus existing vibration levels and analyze the potential for a substantial increase in vibration once the rail line is constructed. This was primarily a measure of vibration from traffic sources, especially university buses which operate frequently throughout the campus during all hours. The project involved recording simultaneous tri-axial vibration measurements in numerous locations throughout the campus. This required designing methods to manage equipment, make measurements, organize data, and present results in an efficient, timely, and coherent manner. The data indicate the level of ambient ground and structural vibration already present from daily university activities while serving as a basis for comparison to the increased vibration levels, if any, the newly constructed train will cause.

4:15

2pNCe5. A method for determining the force spectrum produced by mechanical equipment. Andrew Gorton and Chris Papadimos (Papadimos Group, 818 Fifth Ave., Ste. 207, San Rafael, CA 94901, andrew@papadimosgroup.com)

Currently there is no standardized method for determining the force spectrum produced by a piece of mechanical equipment. This information is

useful for conducting detailed vibration analysis in buildings, specifying proper vibration isolation for mechanical equipment, and comparing mechanical equipment vibration performance. This paper presents a method for estimating the force spectrum for a piece of mechanical equipment supported on spring isolators. The force spectrum is calculated from the measured vibration response of the sprung mass and the frequency response function of the system. For validation, this procedure was used to determine the force spectrum of a 21 500 CFM air handling unit externally mounted on nominally 1-in. static deflection spring isolators. The calculated force spectrum was then used as a force input to a finite element model of a mass-spring-damper system with the same properties as the sprung air handler. The predicted results compare favorably with the measured vibration response and tend to support that this is a viable method for characterizing the force produced by mechanical equipment.

4:30

2pNCe6. Shipboard structureborne noise by resonant interactions of two torsional vibration modes. Darrell Milburn (J&A Enterprises Inc., 16 Broadway, Salem, MA 01970, milburn@jandaenterprises.com)

In recent years several occurrences of a similar engine-firing rate structureborne noise resonance were observed in the passenger areas of different ships. All were 16V four stroke high speed diesel propulsion systems, including engine vibration isolation mountings and remote reduction gears. The engine firing rate noise path came through the reduction gear mounting feet. Interactions of two torsional vibration modes with nearly equal natural frequencies caused the noise: one mode with the maximum amplitude in the engine crank shaft, and the other in the propeller shaft/reduction gear. Over a narrow range of engine speed, the firing rate energy excited both modes. The resonant torsional vibrations were transmitted to the structure through the reduction gear bearing reactions. In the observed cases, the amplitudes of the torsional stresses at resonance were well within the safe limits of the driveline components, raising no concern to vibration analysts assessing only the risk for mechanical failure. This paper outlines the system parameters related to the occurrences of these resonances, permitting noise control engineers to diagnose this problem or to identify "at risk" installations during the design phase. Noise control recommendations are presented as case studies with examples of the achieved noise reduction results.

TUESDAY AFTERNOON, 20 APRIL 2010

DOVER C, 1:20 TO 4:45 P.M.

Session 2pNSa

Noise, Architectural Acoustics and INCE: Ventilation, Fan, and Duct Noise Control II

Lixi Huang, Cochair

The Univ. of Hong Kong, Dept. of Mechanical Eng., Pokfulam Rd., Hong Kong, P.R. China

Kirill V. Horoshenkov, Cochair

Univ. of Bradford, School of Eng., Great Horton Rd., Bradford, BD7 1DP, UK

Jian Kang, Cochair

Univ. of Sheffield, School of Architecture, Western Bank, Sheffield, S10 2TN, UK

Invited Papers

1:20

2pNSa1. Reduction of blade passing noise of a vertical take-off and landing (VTOL) aircraft. Jie Pan (School of Mechanical Eng., The Univ. of Western Australia, 35 Stirling Hwy., Crawley, WA, pan@mech.uwa.edu.au), Hongmei Sun (The Univ. of Western Australia, Crawley, WA, Australia, hongmei@mech.uwa.edu.au), B. S. Walsh, K. D. Do, P. O'Neill, and J. Ranasinghe (The Univ. of Western Australia, Australia)

Rotor-stator interaction has been identified as the dominant noise source of a vertical take-off and landing (VTOL) aircraft developed by Entecho. This paper reports field measurement results of blade passing noise of the VTOL air-craft together with its analysis and control. The blade passing event was simulated in a wind tunnel experiment. The flow speed, rotor position, and rotor-stator spacing were varied with the chord-wise pressure distribution of the lead-ing surface of the rotor blade being measured by an array of six flush

mounted microphones. Results show useful features of the pressure distribution on the rotor blade influenced by an upstream stator, which are used for the analysis and prediction of the sound radiation from the VTOL aircraft. The preliminary result of the reduction in the blade passing noise using angled stator blades is also presented.

1:40

2pNSa2. Prediction of axial fan noise by linearized perturbed compressible equations with large eddy simulation. Young J. Moon, Hoyeon Kim, Youngmin Bae (Dept. of Mech. Eng., Korea Univ., Seoul 136-701, Republic of Korea, yjmoon@korea.ac.kr), Florent Ravelet, Sofiane Khelladi, Hussain Nouri, and Farid Bakir (ParisTech, Paris, France)

The seek for an efficient aerodynamic and aeroacoustic design of axial-flow fans is an important field of investigation for both academic and applied research. Improvements can only be made with a better understanding of the physical mechanisms arising in these machines that combine strong interactions between rotating and non-rotating parts of highly complex geometries. One way is to couple well-suited experimental investigations and innovative computational methods that overtake the weaknesses of methods based, for instance, on aeroacoustic analogy. In this paper we study an axial fan using a new numerical method based on large eddy simulation /linearized perturbed compressible equations with Brinkman penalization. This method is developed in the Department of Mechanical Engineering at Korea University. The experimental tests and validations are performed in the Laboratory of Fluid Dynamics at Arts & Métiers ParisTech in Paris. Detailed analysis of numerical and experimental results is in progress within the two partner teams.

2:00

2pNSa3. The near-field of spinning sources: Why source identification is hard. M. J. Carley (Dept. of Mech. Eng., Univ. of Bath, Bath BA2 7AY, England, m.j.carley@bath.ac.uk)

An asymptotic analysis is presented for the near field of spinning sources, based on a transformation into cylindrical coordinates centered on a line at a fixed radius from the source axis. This transforms the circular source into an equivalent finite length line source with a source distribution made up of “modes” given as Chebyshev polynomials of the second kind $U_n(s) = \sin[(n + 1)\cos^{-1} s]/\sin \cos^{-1} s$. These modes play a role like that of modes in ducts and the analysis shows that their acoustic field propagates or decays depending on whether $n + 1 < k$ or $n + 1 \geq k$, respectively, with k wavenumber, similar to the cut-on/cut-off behavior of duct modes. The analysis is used to examine the problem of identifying a source from field measurements. This has a wide range of applications in many industrial fields but is well known to be (very) ill conditioned. Using the information supplied by the analysis of the source near field, the reasons for this ill conditioning are explained and some possible methods to mitigate the problem are outlined.

2:20

2pNSa4. Toward the prediction of fan noise: From low-speed to high-speed turbomachineries. Stphane Moreau (Dept. of Mech. Eng., Univ. de Sherbrooke, 2500 Boulevard de l'Université, Sherbrooke, QC J1K2R1, Canada stephane.smoreau@gmail.com)

Environmental concern and comfort trigger more and more acoustic specifications and regulations on aircraft impact on airports and on ventilation systems in buildings or transportation systems (cars, trains, or airplanes), which in turn impose lower and lower maximum noise levels to such systems. For instance, turbofan engines have increased their bypass ratio in order to improve the aircraft performance while diminishing the nominal speed of rotation. The jet noise is then reduced, and the fan noise becomes a dominant source of noise, especially at approach. A quick calculation of the overall noise generated by a given fan geometry, either low speed or high speed, would be a valuable asset to any fan design designer and manufacturer prior to any installation in a building or an airplane, for instance. However, an accurate prediction of the sound by any full turbomachinery still remains a challenging goal and a daunting task to be achieved by a direct computation. In the present study, the noise predictions will then rely on a strip theory combined with an acoustic analogy based on the wall pressure fluctuations. For the low-speed fans, the model is an extension of the development by Schlinker & Amiet for helicopters. Turbofan engines induce two additional difficulties to noise modeling: the cascade effects and the duct configuration which are presently modeled.

2:40

2pNSa5. Sound generation mechanism in low speed axial fans. Stefano Bianchi, Alessandro Corsini (Dept. of Mech. and Aeronautics, “Sapienza” Univ. of Roma, Via Eudossiana 18, 00184 Roma, Italy, bianchi@dma.ing.uniroma1.it), and Anthony G. Sheard (Flakt Woods Ltd., Colchester, Essex CO4 5ZD United Kingdom)

Because of the global civil regulation concerning acoustic emission, the goal of the manufacturers is to substantially decrease the noise radiated by the low-speed fans, without degrading their aerodynamic performance. One of the main goal to accomplish this target is the evaluation of the unsteady aerodynamic sources in the fan rotor responsible for the noise emission. An experimental analysis, based on a cause-effect method, was carried out and the results are presented in this paper. The process focuses on the experimental comparison between the novel prototypes of a family of low-speed fans and enabled the evaluation of the noise sources in the rotor near-field domain. The near-field aerodynamic sources accounted in this work correspond to the zones with high pressure fluctuations amplitude, located in the wake region of the exhaust flow. The sources were measured along the rotor radius and then correlated with the noise emitted by the fan at the far-field domain, using a Fourier based methodology. The analysis was carried out looking at the cross-spectra polar form using the phase shift as revealing tool for aerodynamic noise sources. The results were compared to evaluate the merit of the aero-acoustic performance of each prototype.

3:00—3:15 Break

3:15

2pNSa6. Quiet, efficient fans for space flight: An overview of NASA's technology development plan. Danielle Koch (NASA Glenn Res. Ctr., 21000 Brookpark Rd., MS 54-3, Cleveland, OH 44135, l.danielle.koch@nasa.gov)

NASA has developed a Technology Development Plan for improving the aerodynamic and acoustic performance of spaceflight fans. The intent is to make broader use of the technology developed at NASA Glenn to improve the efficiency and reduce the noise of aircraft engine fans. The goal is to develop a set of well-characterized government-owned fans nominally suited for spacecraft ventilation and cooling systems. NASA's Exploration Life Support community will identify design point conditions for the fans in this study. Computational Fluid Dynamics codes will be used in the design and analysis process. The fans will be built and used in a series of tests. Data from aerodynamic and acoustic performance tests will be used to validate performance predictions. These performance maps will also be entered into a database to help spaceflight fan system developers make informed design choices. Velocity measurements downstream of fan rotor blades and stator vanes will also be collected and used for code validation. Details of the fan design, analysis, and testing will be publicly reported. With access to fan geometry and test data, the small fan industry can independently evaluate design and analysis methods and work toward improvement.

3:30

2pNSa7. Aerodynamic design and computational analysis of a space flight vehicle cabin ventilation fan. Daniel L. Tweedt (AP Solutions, Inc., 22526 Tammy Circle, Council Bluffs, IA 51503, daniel.tweedt@ap-solutions.com)

Quieter working environments for astronauts are needed if future long-duration space exploration missions are to be safe and productive. Ventilation and payload cooling fans are known to be dominant sources of noise, with the International Space Station being an important case in point. To address this issue in a cost effective way, early attention to fan design, selection, and installation has been recommended. Toward that end, NASA has begun to investigate the potential for small-fan noise reduction through improvements in fan aerodynamic design. Using tools and methodologies similar to those employed by the aircraft engine industry, the aerodynamic design of a new cabin ventilation fan has been developed, and its aerodynamic performance predicted using computational fluid dynamics (CFD) codes. The design, intended to serve as a baseline for future work, is discussed along with selected CFD results.

3:45

2pNSa8. Surface integration method used on semi-solid surface for fan noise prediction. Yoon-Shik Shin and J. Stuart Bolton (Ray W. Herrick Labs., Purdue Univ., 140 S. Martin Jischke Dr., West Lafayette, IN 47907-2031, shin31@purdue.edu)

As a continuation of previous work, the Ffowc-Williams and Hawkings model was applied to an axial fan that was operated under unfavorable inflow condition. A thin aluminum foam screen that is used as a treatment for disturbed inflow condition was included as a noise source in computational aeroacoustic simulation. The incompressible URANS model was used for the CFD simulation and the inflow foam was modeled as a homogeneous porous zone with permeability and inertial resistance factors found experimentally. Since the homogeneous porous zone defined in this way does not have a solid structure that could be used as a dipole noise source,

the control surface integration method was applied to the surface of the homogeneous porous zone: i.e., to a surface right between the normal fluid zone and the porous zone. The unsteady pressure data on the control surface under the incompressible assumption was treated as representing pressure fluctuations on a semi-solid surface rather than as a representation of acoustic sources within the volume surrounded by the control surface.

4:00

2pNSa9. Cooling fan noise reduction by optimizing spacing of blades. John Wang (Volvo Construction Equip., 312 Volvo Way, Shippensburg, PA 17257, john.wang@volvo.com)

Cooling fan noise theories are briefly reviewed. A simplified model is derived from a rotating point source theory. Optimization was done using the simplified model. A fan with unevenly spaced blades was prototyped, tested, and compared with a fan with evenly spaced blades. Noise reduction data are presented.

4:15

2pNSa10. Suppression of ducted dipole noise by an expanded segment around the source. Lixi Huang (Dept. of Mech. Eng., Univ. of Hong Kong, Hong Kong, lixi@hku.hk)

Low-speed fan noise mainly features dipole sound radiation. For such a fan operating in a duct, the low- to medium-frequency radiation derives mainly from the unsteady, axial forces acting on the rotating blades. Recently, we have found, both theoretically and experimentally, that self-cancellation of sound can be achieved by placing the dipole in an expanded segment of the duct which provides a reverberating environment with a reduced radiation impedance. The insertion loss spectrum shows a very broadband appearance similar to but is better than the transmission loss spectrum of an expansion chamber silencers used for reflecting an incident plane wave. Parametric analysis reveals that the expansion ratio controls the peak insertion loss, while the chamber length controls the single-lobe bandwidth. Simple plane wave analysis implies that short chamber would give wider stopband with a chamber of vanishing length yielding the best result. It is shown here that such analysis fails when the chamber is too short. For a given finite frequency band, there exists an optimal chamber length. Spectral element analysis reveals details of such optimal length and the result is explained in terms of the known fundamentals of dipole radiation and duct acoustics.

4:30

2pNSa11. Active control of axial fan noise using a multiple-input multiple-output feedback controller. Cole V. Duke, Scott D. Sommerfeldt, and Kent L. Gee (Dept. of Phys. and Astron., Brigham Young Univ., N283 ESC, Provo, UT 84602, coleduke@gmail.com)

In the past, significant progress has been made in actively controlling the tonal noise of axial cooling fans using a digital, feed-forward controller. Progress has also been made in controlling broadband fan noise using an analog feedback controller due to the nondeterministic nature of the noise. Current work focuses on the control of broadband noise with a multiple-input multiple-output (MIMO) feedback controller using four control sources and four error sensors. To ensure stability of the closed-loop control system, the phase and magnitude characteristics of the controller must be chosen to produce attenuation in the target frequency band without augmenting the noise outside the band. A practical analog feedback controller will be presented, and performance results will be compared between a single-input single-output controller and a MIMO controller.

Session 2pNSb

Noise, Animal Bioacoustics and INCE: Effects of Noise on Humans and Non-Human Animals II

Ann E. Bowles, Cochair

Hubbs Sea World Research Inst., 2595 Ingraham St., San Diego, CA 92109

Brigitte Schulte-Fortkamp, Cochair

Technical Univ. Berlin, Einsteinufer 25, Secr TA 7, Berlin, 10587, Germany

Invited Papers

1:05

2pNSb1. Determining the effects of long-range missile noise on wildlife. Timothy Lavallee (LPES, Inc., 14053 Lawnes Creek Rd., Smithfield, VA 23430, tlavallee@lpsinc.com)

This is presentation on determining the sound levels from missile test and the potential effects on wildlife. The discussion will include an estimate noise from the launch, rocket motor splashdown, sonic boom, and detonation of long-range missiles. An overview of calculations and modeling to determine exposure to wildlife will be incorporated. The discussion will include overview of the ongoing development of noise threshold criteria for marine mammals and terrestrial wildlife, particularly, dolphins, pinnipeds, and avian species. A literature review and detailed review of the effects threshold will be included. A discussion of the benefits and the shortcoming associated with both the noise modeling and the limitations with respect to the existing effects thresholds will be included.

1:25

2pNSb2. Addressing concerns about ultrasound noise during expansion of a laboratory animal facility. Gary M. Glickman (Wilson, Ihrig & Assoc., Inc., 65 Broadway, Ste. 401, New York, NY 10006, gglickman@wiai.com)

Since 2002 the mouse has become the preferred model for biomedical research largely due to the ease with which it can be genetically engineered. The growth in genetic research has led many institutions to expand their animal holding and research facilities. Managing expectations of investigators during the process of construction and minimizing potential for noise impacts to research are critical for a successful expansion project. Noise in the ultrasound frequency range is often a concern among researchers due to limited data on its causes and response of mice to it and due to the fact that humans cannot perceive ultrasound and are therefore unable to detect when it occurs. Consequently, prior to and during construction, issues involving ultrasound are best given careful consideration so as to minimize it and at the same time address researcher's concerns. Work will be presented from a case study of an expansion project at a medical research facility involving transgenic mice. Issues surrounding the potential for construction noise to impact mice holding rooms will be examined. Particular attention will be paid to measurement and evaluation of noise at ultrasound frequencies and approaches for addressing the concerns of investigators.

Contributed Papers

1:45

2pNSb3. Animal magnetic resonance imaging and acoustic noise. Amanda M. Lauer (Dept. of Otolaryngol.-HNS, Johns Hopkins Univ., 521 Traylor Bldg., 720 Rutland Ave., Baltimore, MD 21205), AbdEl-Monem M. El-Sharkawy, Dara L. Kraitchman, and William A. Edelstein (Johns Hopkins School of Medicine, Baltimore, MD 21205)

MRI scanner noise can cause discomfort and potential damage to human hearing. MRI is used extensively in animal research and, increasingly, as a diagnostic tool in veterinary medicine, generally with no hearing protection. Effects of noise could be detrimental for research animals that will be used after scanning or for companion animals that will be returned to owners. Though safe exposure standards have not been determined for most animals, it is important to consider that animals exposed to scanner noise may suffer hearing damage. To study potential animal MRI adverse hearing effects, we measured sound levels produced by several animal scanning protocols in a commercial 3T scanner. Using hearing threshold data for research or companion animals, we estimate weighted sound pressure levels and compare these to levels at which damage occurs in humans. SPLs were above 90 dB on many scans and often exceeded 100 dB. Animals may be exposed to

these levels intermittently for 1 h or more. Exposure to these sound levels could result in temporary or permanent hearing loss. Thus, it is important to use hearing protection, quieter pulse sequences, or quieter scanners for animal research or companion animal veterinary MRI.

2:00

2pNSb4. Comparison of impulse noise damage risk criteria using the chinchilla impulse noise exposures. William J. Murphy, Amir Khan (Hearing Loss Prevention Team, CDC/NIOSH, 4676 Columbia Parkway, MS C-27, Cincinnati, OH 45226, wjm4@cdc.gov), and Peter B. Shaw (CDC/NIOSH, Cincinnati, OH 45226)

The 1968 CHABA recommendations to limit impulsive noise exposure to levels below 140 dB sound pressure level form the basis of current United States occupational and military standards. The U.S. military standard, MIL-STD 1474D, estimates the number of allowable shots to which a person may be exposed using peak level, B-duration, for varying levels of hearing protection usage. The French Weapons Noise Committee has uses the 85 dBA A-weighted equivalent level, $L_{Aeq8 hr}$, as the limit for allowable exposures.

The U.S. Army sponsored a series of noise exposures with chinchillas to investigate the effects of level, number of impulses, and interstimulus interval. Several types of impulses were created ranging from acoustic shock tubes to narrow band impacts reproduced by a loudspeaker. The goodness-of-fit and the discrimination of five noise exposure metrics were evaluated in this study: MIL-STD 1474D, AHAH model, $L_{Aeq8 hr}$, Pfander's C-duration metric, and Smoorenburg's D-duration metric. Goodness-of-fit was evaluated with a logistic regression and discrimination was evaluated using the area under the receiver operator characteristic curve. The $L_{Aeq8 hr}$ was found to best predict the temporary threshold shifts and the AHAH model was found to best predict the permanent threshold shifts. [Partial funding provided by US Army Aeromedical Research Laboratories MIPR8J07586218].

2:15

2pNSb5. Extraction of hearing loss threshold levels and equal auditory risk metric curves of chinchillas from existing exposure data. Wonjoon Song, Steve Goley, and Jay Kim (Dept. of Mech. Eng., Univ. of Cincinnati, Cincinnati, OH 45221-0072)

An existing set of chinchilla noise exposure data has been re-analyzed using an advanced signal analysis tool and a statistical regression analysis. The data are comprised of 23 digitally recorded noise files and auditory damage indicators of the chinchillas measured at six frequency points, 0.5, 1, 2, 4, 8, and 16 kHz. From the relationship between the equivalent sound pressure level (Leq) of the six 1/3 octave frequency components of the noise and PTS of chinchillas, the Leq level that starts to cause auditory damage to chinchillas is identified as a function of frequency. These threshold levels of chinchillas, obtained for the first time in this study, will be useful information to design chinchilla exposure studies. The levels that will cause the same amount of PTS at the six frequencies are also obtained, from which equal auditory risk metric (EARM) curves are constructed. It is shown that a chinchilla version of an advanced noise guideline which enables quantitative, frequency-by-frequency assessment of the risk of a noise can be developed using the EARM curves. Possibility as well as difficulties of developing a human version guideline similar to this chinchilla prototype are discussed.

2:30

2pNSb6. Survey of ambient noise in aquariums. Colin W. Jemmott (Penn State Grad. Program in Acoust., P.O. Box 30, State College, PA 16804, cwj112@psu.edu)

Owning and maintaining an aquarium is a common hobby, but the ambient noise in aquariums resulting from pumps, filters, bubblers, and other equipment is not well studied. Elevated ambient noise levels have been shown to adversely affect wild fish and marine invertebrates, and anecdotal evidence suggests that this may be a problem in aquariums as well. Aquariums designed to maintain coral reefs require high water flow and pristine water conditions, which in turn require pumps and protein skimmers that contribute to underwater noise. A survey of ambient noise in both fresh and saltwater aquariums ranging in size from 5 to 550 gal was conducted. The aquariums differed in construction material, number, size and type of pumps, and presence of other equipment. The ambient noise broadband lev-

els, spectrum levels, and time-frequency representations are compared, and general conclusions are drawn about noise characteristics of equipment and aquariums.

2:45

2pNSb7. A new tranquility rating tool and the role of audio and visual interactions. Kirill Horoshenkov, Robert Pheasant, Greg Watts (School of Eng., Univ. of Bradford, Bradford BD7 1DP, United Kingdom, k.horoshenkov@bradford.ac.uk), David Whitaker (Univ. of Bradford, Bradford BD7 1DP, United Kingdom), and Jian Kang (Univ. of Sheffield, Sheffield S10 1FL, United Kingdom)

Restorative environments which enable individuals to recover the sense of well-being are becoming increasingly important. These environments are characterized by an enhanced level of tranquility. Therefore, obtaining the optimum balance between the landscape and soundscape characteristics and measuring the resultant tranquility of these spaces are essential for their design and maintenance. In order to understand the key factors which affect the tranquillity construct, a large volume of audio and visual data has been collected across a representative range of landscapes in the UK. These data have been analyzed objectively by studying the temporal and spectral characteristics of the recorded sounds and the proportion of the natural and contextual features present in the video clips. The tranquility rating of these landscapes has been obtained from subjective experiments on 44 subjects to whom uni- and bimodal stimuli have been presented in a separate experiment. The results of these experiments make possible to objectively measure the key components of the tranquillity construct and determine their relative importance in the design of a restorative space. On this basis new relations between the tranquility rating, sound pressure level, loudness characteristics, and the visual quality of the scene have been derived.

3:00

2pNSb8. The effect of anthropogenic noise on the calling behavior of amphibians and birds in urban areas of Puerto Rico. Maria Isabel Herrera-Montes and Mitchell Aide (Dept. of Biology, Univ. of Puerto Rico, P.O. Box 70377, San Juan, PR 00936-8377, isahemontes@yahoo.com)

The bird and amphibian communities of Puerto Rico have high levels of diversity and endemism. Although Puerto Rico has a very high density of roads and high levels of noise pollution, presently 40% of the island is covered in forest. I will address the following questions: Do high levels of anthropogenic noise change species composition of amphibians and birds? Are amphibians and birds modifying the time of calling and/or characteristics of their vocalization in response to anthropogenic noise? I collected sound recordings in two habitats. For recordings I used the automated recording devices developed by our research group. Each recording device was programmed to record for 7 consecutive days; 1 min every 20 min. Previous results showed that noise influences the bird community composition in secondary lowland forest sites, but not in karst forest. Another possibility is that bird and amphibian use other strategies to minimize the effect of noise. This can involve evolutionary changes in signal characteristics, as a long term adaptation or the species may be adjusting signal traits in response to variations in noise levels as a short term adaptation. I will answer these questions when I do analyzing all data.

3:15—3:45 Panel Discussion

Session 2pNSc**Noise, Animal Bioacoustics, and INCE: Acoustics and Public Policy**

Nancy S. Timmerman, Chair
 25 Upton St., Boston, MA 02118

Invited Papers**1:20**

2pNSc1. Hearing protection devices: The regulatory journey to change. Catrice Jefferson (U.S. Environ. Protection Agency, 1200 Pennsylvania Ave., NW, Washington, DC 20460)

The regulation for the labeling of hearing protection devices (40 CFR Part 211, Subpart B) was promulgated in 1979 and has been the directive for 30 years. During this time, there have been significant changes in the industry with respect to newly developed products, which has resulted in the demand for various types of hearing protectors and an increased awareness of hearing loss in both occupational and non-occupational environments. The foundation of the newly proposed regulation of August 5, 2009 is based on these observations and those that were presented by industry, hearing conservation professionals, and the general public during the EPA Workshop on Hearing Protection Devices that was held in March 2003. While these observations will serve as the backdrop of the presentation, the scenery will be the key elements of the proposed regulation, such as the testing methodology, noise reduction rating range, label format and content, recurrent testing, and reporting requirements of testing data. The audience will journey through these scenic elements but will be delivered to its final destination, which are regulatory change and the prospects of withstanding another 30 years.

1:40

2pNSc2. Proposal for assessing the effectiveness of noise policies. Lawrence Finegold and Michiko So Finegold (Finegold & So, Consultants, 1167 BournemOUTH Ct., Ctr.ville, OH 45459, lsfinegold@earthlink.net)

During the last half of the 20th century, most of the developed countries of the world have been addressing noise as an environmental and occupational problem, and have been working to develop noise control and exposure policies and related noise control technologies. Since the 1970s, the United States has been an international leader in these efforts, although more recently significant advances have been made elsewhere, particularly by the European Commission. However, little is known about how effective various noise policies and regulations have been in controlling the noise exposure of the populations which they are intended to protect. This paper proposes a national program to address this issue. The first phase involves the collection and cataloging of noise policy regulations and related guidelines and, for the most part, has already been completed by the International Institute of Noise Control Engineering, including a comprehensive index of U.S. noise policies. The second phase would involve developing a baseline of noise exposure estimates for various community environments. The third phase would determine the long-term effectiveness of noise policies in controlling noise exposure by examining the changes in various noise exposures and noise effects over time.

2:00

2pNSc3. Integrating disturbance ecology and public policy: A better framework for managing the effects of anthropogenic noise. Ann E. Bowles (Hubbs-SeaWorld Res. Inst., 2595 Ingraham St., San Diego, CA 92109) and Amy R. Scholik-Schlomer (Natl. Marine Fisheries Service, Office of Protected Resources, 1315 East-West Hwy., Silver Spring, MD 20910)

The Marine Mammal Protection Act and the Endangered Species Act are statutes designed to conserve and recover marine mammal species. In addition to language that restricts activities that could "take" individuals via death, they restrict activities that could harass marine mammals. By framing the problem in terms of harassment, these statutes acknowledge the potential for significant behavioral effects. However, predicting biologically significant effects of disturbance is often difficult. A better theoretical and technical framework is needed that accounts for the adaptive scope (both physiological and behavioral) of large, intelligent, wide-ranging, and long-lived marine mammals. While ASA members have debated policy statements on the inappropriate use of acoustic concepts in controversies over marine mammal exposure to noise, they have not effectively addressed the need for better integration of biological theory. Of particular importance are theories on cognitive function, adaptive defensive strategies, and allostasis that help explain why marine mammal responses can be intelligent and flexible in some cases and injurious in others. By integrating such biological concepts into its policy statements, the Society could foster more effective use of research resources and help stakeholders support regulatory instruments for noise that would effectively balance human needs with those of marine mammals.

2:20

2pNSc4. Disastrous decisions: Costs to natural quiet of Grand Canyon National Park. Dickson J. Hingson (275 S. River Run, Flagstaff, AZ 86001)

Noisy intrusions from air tour rides and other aircraft increasingly, adversely affect highly significant areas of the Grand Canyon National Park. These continue in the heart of the park, even though Theodore Roosevelt told us to "do nothing to mar its grandeur." Preserving the Canyon's numinous aura requires maintaining its emblematic silence. The requisite policy would be to severely restrict aircraft noise. Belated efforts for an aggressive noise policy began here soon after 1965 airport completion near the Park. Hard won Congressional mandates for restoration of natural quiet were nonetheless frustrated by an accelerating aircraft boom fed by still more bad decisions, FAA stonewalling/bullying, and industry selfishness. The NPS/FAA environmental outcome—being finalized now, under a "change" administration—is coming into focus via a detailed EIS under public comment. The still unmet national direction to curb this

noisy, commercial park “take-over” illustrates serial failures in group decision-making. A “road map” of contributing reasons for a veritable acoustic/soundscape “collapse” here gains strength from insights in Jared Diamond’s recent book “Collapse”:, i.e., failures to anticipate, to perceive, and to behave or obey. To right these wrongs, the concerned public must assertively guide the current Administration’s implementation of substantial, enduring restoration of the Park’s natural quiet.

2:40

2pNSc5. Sex, money, religion, and politics: Understanding noise policy in terms of what is left unsaid. Leslie D. Blomberg (Noise Pollution Clearinghouse, Box 1137, Montpelier, VT 05601-1137)

Sex, money, religion, and politics. We are not supposed to talk about these things. And when it comes to discussions of creating a comprehensive noise policy, we generally do not, and perhaps we should not. But we should at least understand their contribution to the mxe9;lange of laws and regulations that is our current noise policy. This paper examines the elephant in the room that people are not talking about.

3:00—3:20 Break

3:20

2pNSc6. Would Western noise policies be effective in developing countries? Lawrence Finegold (Finegold & So, Consultants, 1167 Bournemouth Ct, Ctr.ville, OH 45459, Lsfinegold@earthlink.net) and Dieter Schwela (Swedish Environ. Inst.)

This paper examines the question of whether current and evolving Western noise policies would be effective and appropriate for use in developing countries. Differences in noise sources, available finances and noise control technologies, cultural norms, climate, views concerning the role of the government, etc., make it likely that different approaches might be needed in developing countries for their noise policies to be effective. Thus, it recommends that an international consortium of scientists and engineers, government representatives, and key stakeholders be organized to address this important topic. The noise policy needs of developing countries require an awareness of their typical noise problems, along with appropriate, affordable, and technologically feasible solutions, and how these might be incorporated into emerging Global Noise Policy concepts. This paper presents a concept for a Strategic Approach to Environmental Noise Management in Developing Countries, developed by the Swedish Environmental Institute (SEI), to form a foundation for the proposed international consortium. The next step is to now formalize an international consortium to provide additional technical input, to encourage participation by international agencies, professional societies and stakeholders, to organize workshops and special sessions at acoustics congresses, and to promote the evolving SEI concept with the governments of developing countries.

3:40

2pNSc7. There ought to be a law! Nancy S. Timmerman (25 Upton St., Boston, MA 02118, nancy.timmerman@alum.mit.edu)

This paper will examine some of the logical consequences of laws and policymaking as they relate to acoustics. In acoustics and noise control, “public policy” is used to try to legislate good behavior. Those who behave well do not need the laws. Those who do not will not obey the laws. Examples will be drawn from aviation and planning, examining the use of measures such as DNL. In particular, this paper will address how laws create unintended results, frequently not addressing the issue at all.

Contributed Papers

4:00

2pNSc8. New York City: The (E) designation and restrictive declaration. Benjamin Sachwald (AKRF, 440 Park Ave., South 7th Fl., New York, NY 10016, bsachwald@akrf.com)

Urban areas are typically associated with high-noise levels. With a population of more than 8×10^6 and growing, New York City is one of the largest cities in the world. In New York City, because of the population density and the wide variety of land uses (for example, residential, commercial, transportation, manufacturing, and industrial) that exist in close proximity, it is common for a residential building to be located in an area with ambient noise levels that would be considered unsuitable for residential use. To protect inhabitants of a planned building to be located in an area with high ambient noise levels, an (E) designation or a Restrictive Declaration may be used to ensure that the building’s interior environment meet a certain acoustical design criterion. In this paper, the (E) designation and Restrictive Declaration are defined, their regulatory process is outlined, and specific case studies are discussed.

4:15

2pNSc9. Recommendations for the improvement of the management of noise from helicopter operations. Paul Kendrick, David C. Waddington, Geoff Kerry (School of Computing, Sci. and Eng., Univ. of Salford, Salford M5 4WT, United Kingdom, p.kendrick@salford.ac.uk), Matthew Muirhead, and Ray Browne (QinetiQ Ltd., Farnborough GU14 0LX, United Kingdom)

This paper results from recent research commissioned by the Department for Environment, Food and Rural Affairs (Defra, UK) with the objective of identifying possible opportunities for the improvement of the management of noise from helicopter operations in the UK. Effected populations are not just those living close to heliports but include those exposed to noise from helicopters used by the military, emergency services, and commercial companies. A detailed report has been produced addressing the improvement of the management of helicopter noise. Also produced was a short non-technical guide including the means of redress for perceived disturbance. The report also makes clear recommendations for future work and identifies particular areas where specific research is required. This paper discusses these recommendations and suggests a way forward for research into the improvement of the management of helicopter noise. [Work funded by the Department for Environment, Food and Rural Affairs (Defra), UK].

Session 2pPA

Physical Acoustics and Engineering Acoustics: Ultrasonics, Nonlinear Acoustics, Acousto-Optics, and Engineering Acoustics in Honor of Mack Breazeale II

Laszlo Adler, Cochair

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Chair's Introduction—12:55

Invited Papers

1:00

2pPA1. Highlighting the career of Mack Breazeale. James G. Miller (Dept. of Phys., Washington Univ., 1 Brookings Dr., St. Louis, MO 63130, james.g.miller@wustl.edu)

Mack Breazeale's creative work was respected and recognized by professional organizations including the Acoustical Society of America, the Institute of Acoustics (Great Britain), and the IEEE Ultrasonics, Ferroelectrics, and Frequency Control Society (UFFC), each of which honored him with the rank of Fellow. Breazeale served as the UFFC Distinguished Lecturer in 1987–1988, during which time he traveled extensively to deliver highly regarded presentations. In recognition of Breazeale's many contributions to the IEEE UFFC, including his many-year participation in the Technical Program Committee that sets the theme for the Annual IEEE Ultrasonics Symposium, Breazeale was honored with the Achievement Award in 2008, the highest Society-wide award presented to a member in special recognition of outstanding contributions. I had the pleasure of highlighting Breazeale's many contributions that led to this prestigious honor during the awards ceremony at the 2008 Ultrasonics Symposium in Beijing. When I wrote to Breazeale to express my congratulations on this award and to discuss how I might summarize his many achievements in a career spanning decades, Breazeale responded by saying, "The most important aspect of my career is the fact that I have served as Major Professor to 15 MS candidates and 16 doctoral candidates."

1:10

2pPA2. Training graduate students: The teaching philosophy of Mack Breazeale. Michael S. McPherson (Dept. of Phys. and Astronomy, 1906 College Heights Blvd. #11077, Bowling Green, KY 42101, mike.mcpherson@wku.edu)

Mack Breazeale's last Ph.D. student will comment on Dr. Breazeale's contributions to solid state acoustics, nonlinear acoustics, and acousto-optics. In particular, this talk will focus on Dr. Breazeale's approach to, and methodology for, training and teaching graduate students how to conduct research and mature into capable scientists. This talk will speak to Dr. Breazeale's unique gifts as a teacher and mentor and his approach to conducting scientific work in general.

1:20

2pPA3. Research on acoustical memory discovered in the laboratory of Mack Breazeale. Igor Ostrovskii (Dept. of Phys., Univ. of Mississippi and NCPA, University, MS 38677, iostrov@phy.olemiss.edu)

Research at Mack's Laboratory at NCPA has discovered that a LiNbO₃ single crystal can memorize an ultrasonic pulse for tens of microseconds, the frequency range was 17–28 MHz. Supposedly, the Acoustical Memory (AM) effect is connected to the oxide internal microstructures. This conclusion is based on frequency, thermal, and ultrasound attenuation data. Mack Breazeale's leadership, scholarly character, and gentle nature were clearly revealed through these experimental investigations, various meetings, and scientific discussions in his office and laboratory. However, a basic connection between the ceramic microstructures and AM formation is not revealed yet. Further development of AM investigations is done in the inversely poled ferroelectric wafers of LiNbO₃ and LiTaO₃. The speeds of different acoustic modes are altered in the presence of periodically poled ferroelectric domains. Calculations for a bulk periodically poled LiNbO₃ yield a similar result despite the uniform acoustic properties along the specimen. Distortions in the dispersion curves of ultrasound in non-homogeneous ferroelectrics with specific internal microstructures may be a physical cause of acoustical memory. Some applications are discussed. [Work supported by UM.]

1:40

2pPA4. Theoretical and experimental investigation of the backward beam displacement. Nico F. Declercq (UMI Georgia Tech, CNRS 2958, 2 Rue Marconi, 57070 Metz France and Woodruff School of Mech. Eng., Georgia Tech, Atlanta, GA 30332-0405)

The discovery of a backward beam displacement of ultrasound interacting with a periodically corrugated surface, dates back from 1976, when Breazeale and Torbett reported it [Appl. Phys. Lett. **29**, 456 (1976)]. Since 2002 new investigations have been undertaken partially motivated by Breazeale's enthusiasm. An overview is presented of how the phenomenon was first discovered in 1976, how a theoretical explanation was found since 2002 [Declercq *et al.*, Appl. Phys. Lett. **82**, 2533 (2003); Declercq *et al.*, J. Appl. Phys. **96**,

6869–6877 (2004)] and what further verifications and discoveries have been made since then [Teklu *et al.*, *J. Appl. Phys.* **97**, 084904-1 (2005)]. The report also highlights new research showing the ubiquitous presence of the phenomenon and its importance for nondestructive applications [Herbison *et al.*, *J. Acoust. Soc. Am.* In press].

2:00

2pPA5. Breazeale legacy in Gaussian acoustics. Dehua Huang (NUWC, Newport, RI 02841)

Professor Breazeale is a great educator, an outstanding researcher, and scientist in many acoustic fields. This talk focuses professor efforts and leadership in pioneer Gaussian acoustics theory and practice. In 1992, instead of going by KZK or Helmholtz equations for theoretical Gaussian acoustic wave solution, a new path was discovered based on the far field approximation from the Rayleigh integral. By which, the mathematical solution matches with the other Gaussian wave equation results. Further, the design of the Gaussian acoustic transducer technologies is evolved. It was only possible for 2 MHz and above frequency band operating Gaussian transducers. This frequency band barrier was crossed over, and a set of kilohertz Gaussian transducers was fabricated in the laboratory by a new design. The definition of a qualified Gaussian transducer was also finalized. The propagation of a Gaussian wave in an ocean environment was also simulated.

2:20

2pPA6. Acoustic parametric oscillators: In the footsteps of Mack Breazeale. Lev A. Ostrovsky (Zel Technologies and NOAA Earth System Res. Lab., 325 Broadway, Boulder, CO 80305)

Parametric oscillators have been studied in physics since the 18th century. In acoustics, the effect of transformation of pump energy to a half-frequency subharmonic (or, more generally, to any lower-frequency signals) began to be studied in 1960s by Korpel and Adler and in the laboratory of Breazeale. They observed basic parametric effects such as half-frequency generation and generation of lower-frequency doublets. In early 1970s, our group in Russia began studies of acoustic parametric phenomena in dispersive systems (waveguide resonators), in which most of extra harmonics and subharmonics are suppressed so that generation at the desired frequencies occurs much more efficiently. Indeed, in these experiments the subharmonic mode was generated so intensively that it can even damp the pumping mode in resonance. Also the later experiments with “non-classical,” strongly nonlinear solid materials, such as ferroelectrics and metals, should be mentioned. The more recent works include parametric phenomena in bubbles; moreover, the group of Lauterborn in Germany has demonstrated that the period-doubling scenario of transition to chaos can be realized in cavitation noise. This presentation outlines a brief history of acoustic parametric oscillators and relevant effects in the context of pioneering works of Professor Mack Breazeale.

2:40

2pPA7. Close encounters of the distinguished kind. Koen Van Den Abeele (Wave Propagation and Signal Processing, K.U. Leuven Campus Kortrijk, E. Sabbelaan 53, B-8500 Kortrijk, Belgium, koen.vandenabeele@kuleuven-kortrijk.be)

Mack Breazeale entered my life at the very start of my research career in Kortrijk, Belgium. At the 20th anniversary of the university campus, he was co-organizer together with Oswald Leroy of the Physical Acoustics Meeting. I had the honor to edit and compile the book containing all contributed papers. Since that moment we have met each other regularly in Kortrijk or at an international meeting. Later he was a member of the jury at the defense of my doctoral thesis, and the same evening he managed to persuade my wife and me to come to NCPA for a year as a post-doc. We will always remember Ole-Miss and the parties at the gentle Breazeale family. Upon returning to Belgium I immediately got an email from Paul Johnson, triggered by Mack Breazeale, asking me to apply for a post-doc at Lanl. Thanks to Mack (and Paul), we had an unforgettable experience there. I consider it the birth of nonlinear acoustics and ultrasonics in Belgium. Oh, and by the way, we have performed some great science together in acousto-optics and nonlinear acoustics.

3:00—3:20 Break

Contributed Papers

3:20

2pPA8. Frequency dependence of the ultrasonic parametric threshold amplitude for a fluid-filled cavity. Alem A. Teklu (Dept. of Phys. & Astronomy, College of Charleston, 66 George St., Charleston, SC 29424, teklu@cofc.edu), Michael A. McPherson (Western Kentucky Univ., Bowling Green, KY 42101), Mack A. Breazeale (Natl. Ctr. for Physical Acoust., University, MS 38677), and Nico F. Declercq (Georgia Tech Lorraine, 57070 Metz-Technopole, France)

The excitation of fractional harmonics in a liquid-filled cavity by ultrasonic waves was described previously as a parametric phenomenon [L. Adler and M. A. Breazeale, *J. Acoust. Soc. Am.* **48**, 1077–1083 (1970)]. That is, by driving a transducer at one end of a fluid-filled cavity parallel to

a rigid plane reflector at the other end, standing ultrasonic waves can be generated. Variations in the cavity length resulting from transducer motion lead to the generation of resonant frequencies lower than the drive frequency (known as fractional harmonics). The system was modeled by using a modified Mathieu equation whose solution resulted in the prediction of critical threshold drive amplitude for the excitation of parametric oscillation. The apparatus used by Adler and Breazeale was recently refined for accurate measurements of the threshold amplitude for parametric excitation at frequencies ranging from 2 to 7 MHz. The measurements showed that in this range the threshold amplitude increases with increasing drive frequency in apparent discrepancy with the results of Adler and Breazeale. Analysis of the theory indicates, however, that both past and current results lie in two different stability zones, and each is in agreement with the existing theory.

3:35

2pPA9. Optical Bragg imaging of acoustic fields after reflection. Michael S. McPherson (Dept. of Phys. and Astronomy, Western Kentucky Univ., 1906 College Heights Blvd. #11077, Bowling Green, KY 42101, mike.mcpherson@wku.edu), M. A. Breazeale (Univ. of Mississippi, University, MS 38677), Nico F. Declercq (Georgia Tech Lorraine, UMI Georgia Tech, 57070 Metz-Technopole, France), and Alem Teklu (College of Charleston, Charleston, SC 29424)

If visible light is incident at the Bragg angle on an acoustic wave of the appropriate frequency, the light will undergo Bragg diffraction from the acoustic wavefronts in a manner analogous to the Bragg diffraction of x-rays in a crystalline lattice. If the acoustic beam has previously passed through an object before having light diffracted from it, the structure of that object will be imaged in the optical diffraction orders. In traditional Bragg imaging applications, the sound is first transmitted through the object. In this presentation a new reflection-based method of Bragg imaging (where light is reflected from rather than transmitted through the object) will be discussed. This technique is capable of imaging structures at or near the surface of the object and may have potential industrial and biomedical NDT applications.

3:50

2pPA10. Infrasound-convection instability. Konstantin Naugolnykh (Dept. of Phys., Univ. of Colorado, 325 Broadway, Boulder, CO)

Heated below temperature stratified atmospheric layer is unstable and convection flow can be developed in such an area. Convection happens because warm less dense air goes up while cooler air comes down. Infrasound can initiate convection flow and being amplified by the flow. The process of infrasound-convection interaction is considered in the present paper. The model of convection in a compressible fluid is considered and analyzed. The conditions of infrasound amplification are obtained.

4:05

2pPA11. Ultrasonic imaging and characterization of objects submerged in highly attenuating fluids. Bart Raeymaekers, Cristian Pantea, Curtis F. Osterhoudt, and Dipen N. Sinha (Los Alamos Natl. Lab., MPA-11, P.O. Box 1663, MS D429, Los Alamos, NM 87545)

An ultrasonic scanning tool to image objects with high-spatial resolution submerged in highly attenuating fluids is described. The attenuation coefficient of the fluids used range over two orders of magnitude. For deeper penetration into the attenuating fluids, a parametric array with a difference frequency between 200 and 500 kHz was used. The primary frequencies that generated this difference frequency varied between 2 and 8 MHz. This parametric array provides a collimated beam for imaging. Both simple tone burst and frequency chirp signals were used for the measurement. Using this apparatus, the quality and accuracy of the image as a function of frequency, fluid attenuation coefficient, and fluid type (clear and suspensions) were investigated. It was found that imaging was possible over a wide range of attenuating coefficients and fluid types.

4:20

2pPA12. Batchelor's fourth order velocity correlation. John M. Seiner (Natl. Ctr. for Physical Acoust., The Univ. of Mississippi, University, MS 38677)

The Lighthill theory for aerodynamic noise relates far field sound pressure to complex fourth order two point two-time space-time velocity correlations. The Mani-Gliebe-Balsa (MGB) jet noise prediction code fol-

lows Lighthill's theory in the development of its algorithm. Since fourth order turbulence correlations are not modeled in the MGBs flow solver, it is necessary to resort to some form of approximation. The MGB invokes Batchelor's fourth order decomposition, which is exact for homogeneous and isotropic turbulence. For anisotropic flow, like that of a jet, it is not known what the error is in adopting Batchelor's isotropic decomposition. The question here is can stereo particle image velocimetry (SPIV) provide a means to establish the error incurred with adoption of this decomposition. In an attempt to define the penalty for adopting the Batchelor's decomposition, a simple 2×94 ; diameter laboratory jet was operated at ambient temperatures at Mach 0.85. SPIV measurements were acquired of the fourth order two-point space correlation and all of the other second order correlations. A comparison was made to numerical plume simulations with an advanced numerical flow solver running in the RANS mode. This paper provides an explanation for observed variations between measured and predicted jet noise.

4:35

2pPA13. Multipole expansion for nonlinear acoustic scattering at the difference-frequency. Glauber T. Silva (Inst. of Phys., Univ. Fed. Alagoas, Maceio, Alagoas 57072-970, Brazil, glauber@pq.cnpq.br)

In this work, the scattering at the difference-frequency (DF) of two finite-amplitude plane waves by a rigid sphere in a fluid is studied. The DF-scattering is present in some acoustic imaging methods such as vibro-acoustography. This has motivated us to perform this study. We obtain for the first time the multipole expansion for the DF-scattered pressure in the farfield. Theory is based on the successive approximation and the Green's function methods. From the multipole expansion, we derive some quantities such as scattering cross-section and the total DF-scattered power. Furthermore, we show that the DF-scattered pressure increases with the observation distance r like $r \ln r$, while it varies almost linearly with the difference-frequency. Despite the amplification, we demonstrate that the DF-scattered pressure does not grow indefinitely because of absorption in the fluid. The theory is applied to the scattering in the Rayleigh limit (scatterer much smaller than the incident wavelengths). Only monopole and dipole moments are relevant in this limit. In conclusion, we believe that this theory can help to understand better the nonlinear scattering in acoustics.

4:50

2pPA14. A new form of Gaussian beam function as the base function for generic wave field analysis. J. J. Wen (VeroSound, Inc., 128 Powder Mill Rd., Sudbury, MA 01776)

Wave propagations are best described by three major approaches: the spherical wave approach represented by Rayleigh integral, the planar wave approach represented by spatial Fourier analysis, and the Gaussian beam approach. Spherical and planar wave approaches, both in form of double integrals, are analytically inconvenient and computationally expensive. In contrast, the Gaussian beam approach, with huge advantages in analytical operability and computational efficiency, has been quietly but firmly winning popularity. Dr. Breazeale's article: A diffraction beam field expressed as the superposition of Gaussian beams, co-authored with Wen, is not only a most representative work for Gaussian beam approach but also a most cited academic work in sound field analysis and computation. As a memory of Dr. Breazeale's pioneer role in the raise of Gaussian beam method, this article presents a new enhancement to the existing Gaussian beam method. A modified form of Gaussian beam function introduced as the base function set for wave field analysis greatly reduces the error rooted from the parabolic approximation, the number of Gaussian beams needed, and the run-time computation burden. The process for calculating the beam coefficients and beam parameters is also greatly simplified.

2p TUE. PM

Session 2pPP**Psychological and Physiological Acoustics and Engineering Acoustics: Application of Auditory Models to Hearing Aids**

Brent W. Edwards, Chair

*Starkey Hearing Research Center, 2150 Shattuck Ave., Ste. 408, Berkeley, CA 94704***Invited Papers****1:00**

2pPP1. The application of auditory models to hearing aid research. Brent W. Edwards (Starkey Hearing Res. Ctr., 2150 Shattuck Ave., Ste. 408, Berkeley, CA 94704, brent_edwards@starkey.com)

Audibility has been a focus of hearing aid research for much of the hearing aid field's existence. With modern hearing aids, audibility no longer limits performance for hearing aid wearers with moderate hearing impairment. Limitations in suprathreshold auditory-perception now need to be better understood in order to determine existing hearing deficits while aided. The ways in which hearing aids affect suprathreshold auditory function also need to be better understood in order to ensure that hearing aids provide maximal benefit in complex real-world situations. Auditory models can be used to address these needs. This talk will review current applications and future opportunities for the application of auditory models to hearing aid research.

1:20

2pPP2. Differentiating inner from outer hair cell damage: Measures and applications to model-based hearing aid design and fitting. Enrique A. Lopez-Poveda, Peter T. Johannesen, Patricia Prez-González, Jorge M. Mndez, and Almudena Eustaquio-Martín (Instituto de Neurociencias de Castilla y Leon, Univ. Salamanca, C/Pintor Fernando Gallego 1, 37007 Salamanca, Spain, ealopezpoveda@usal.es)

Hearing aids (HAs) are designed and fitted mainly to restore normal audibility and loudness growth. Unfortunately, after decades of research, many users still find them frustrating. We hypothesize that this is partly because HAs operate on the assumption that cochlear hearing loss is always associated with a reduction in cochlear gain and/or compression when this, in fact, need not be the case. Indeed, recent studies have shown that cochlear hearing loss frequently relates to selective or combined dysfunction of inner hair cells (IHCs) and outer hair cells (OHCs). Neither absolute thresholds nor loudness growth allows for a differential assessment of the type of hair cell damage, yet such assessment could be highly informative for optimizing HA design and fitting. In this presentation, we summarize our progress in designing a fast, universal method for inferring the degree of IHC and OHC dysfunction using distortion-product otoacoustic emission input/output functions. We also describe our progress in designing a HA based on these ideas and on a bank of dual-resonance non-linear filters inspired by computational models of healthy and impaired human cochlear responses. [Work supported by the MEC, Grant BFU2006-07536, and The Oticon Foundation.]

1:40

2pPP3. The use of a loudness model to derive initial fittings of multi-channel compression hearing aids. Brian C. J. Moore (Dept. of Experimental Psych., Univ. of Cambridge, Downing St., Cambridge CB3 9LG, United Kingdom, bcjm@cam.ac.uk)

A model of loudness perception applicable to hearing-impaired people has been used to develop three methods for the initial fitting of hearing aids with multi-channel compression, based on the audiometric thresholds. Method one, CAMEQ, has the goals of amplifying speech so that, on average, the specific loudness is the same for all frequencies within the range 500–4000 Hz, over a wide range of overall sound levels, giving about the same overall loudness as “normal” for speech covering a wide range of sound levels. Method two, CAMREST, determines the gains needed give normal specific loudness patterns for speech over a wide range of sound levels. Method three, CAMEQ2-HF, is similar to CAMEQ but differs from it in the following ways: (1) gains are recommended for center frequencies up to 10 kHz (as compared to 6 kHz for CAMEQ), (2) CAMEQ2-HF is based on the assumption that the user may wish to hear sounds from many directions and uses a diffuse-field-to-eardrum transfer function (as opposed to a free-field-to-eardrum transfer function for CAMEQ); (3) CAMEQ2-HF is based on recent wideband measurements of the average spectrum of speech. All of the methods can be applied to any multi-channel compression hearing aid.

2:00

2pPP4. Modeling of the auditory periphery for hearing aid evaluation and design. Ian C. Bruce (Dept. of Elec. & Comp. Eng., McMaster Univ., 1280 Main St. W., Hamilton, ON L8S 4K1, Canada, ibruce@ieee.org)

Amplification schemes and gain prescriptions for hearing aids have thus far been developed and evaluated based on perceptual criteria such as speech intelligibility, sound comfort, and loudness equalization. Finding amplification strategies that simultaneously optimize all of these perceptual metrics has proven difficult, despite decades of research. Furthermore, novel amplification schemes based on rough conceptual models of the normal and impaired auditory physiology have often proven to be unsuccessful. In this talk, I will describe studies directly employing physiologically accurate computational models to evaluate more rigorously what hearing aids are likely doing to the neural representation of speech. The results of these investigations indicate that (i) a physiologically accurate

auditory-nerve (AN) model can predict optimal linear and nonlinear amplification gains, (ii) optimal gains are dependent on both the spike-timing and mean-rate representations of speech in the AN, (iii) the proportion of outer hair cell and inner hair cell dysfunction in an impaired ear can affect optimal gains, and (iv) slow wide dynamic range compression (WDRC) or automatic gain control better restores the neural representation of speech than does fast WDRC.

2:20

2pPP5. Application of auditory models to assessing the sound quality of hearing aid signal processing algorithms. Volker Hohmann (Medizinische Physik, Universität Oldenburg, Oldenburg, Germany)

Advances in systems technology allow for increasingly complex processing algorithms in hearing systems addressing increasingly complex acoustic conditions. These developments have the potential of improving the rehabilitation of hearing impairment, but establishing reliable measures of benefit is quite difficult for these complex algorithms and conditions. Being the “gold standard” for algorithm evaluation, subjective testing of hearing-impaired subjects has some limitations in this context. It is very time consuming and the long acclimatization time needed when listening with new devices in complex acoustic conditions cannot easily be accounted for. Therefore, objective methods for estimating the benefit of an algorithm in a certain acoustic condition are desirable. They allow for identifying promising candidate algorithms and the acoustic conditions in which the algorithms might be applicable and thus identify critical acoustic conditions to be tested subjectively. In addition and in combination with technical measures such as segmental signal-to-noise ratio and distortion measures, perceptual measures based on auditory models might be useful for developing meaningful objective measures. This talk presents recent applications of auditory models to objective algorithm evaluation. Because of the growing importance of binaural and multi-microphone processing in hearing instruments, binaural/multichannel models for speech intelligibility and quality will be emphasized.

2:40

2pPP6. Auditory models of suprathreshold distortion in persons with sensorineural hearing loss. Ken W. Grant, Van Summers, Joshua G. W. Bernstein, Sandeep A. Phatak, Matthew J. Makashay, Elena Grassi, and Golbarg Mehraei (Army Audiol. and Speech Ctr., Walter Reed Army Medical Ctr., Washington, DC 20307-5001, ken.w.grant@gmail.com)

Hearing technology has advanced to where it is now reasonable to ask whether signal processing algorithms can be developed to compensate for an individual’s hearing loss, thus allowing them to hear functionally in a manner similar to persons with normal hearing. Clinically, the pure-tone audiogram is the primary tool used to represent the patient’s hearing status. However, it has been well established, experimentally and theoretically, that the audiogram cannot reflect fully all aspects of the hearing loss, most notably that part which pertains to suprathreshold distortion. Much has been written about the distortion component of sensorineural hearing loss, yet there is little agreement on estimating its importance to speech recognition, nor much consensus on which hearing factors (e.g., spectral and/or temporal resolution) most accurately represent the distortion. Recent attempts to use biologically inspired models of auditory processing to represent a patient’s internal auditory experiences have shown promise as a way to understand the role that suprathreshold distortions might play in speech recognition. This talk reviews recent work to develop auditory models of {it individual} hearing-impaired listeners to predict differences in the perception of speech and non-speech signals not readily explained by audiometric thresholds. [Work supported by the Oticon Foundation.]

3:00—3:10 Break

3:10

2pPP7. Real-world application of auditory models to hearing instrument design. Stefan Launer (Phonak AG, Laubisruetistrasse 28, 8712 Staefa, Switzerland)

Auditory models provide a powerful tool for designing and developing signal processing algorithms of modern digital hearing instruments and can even be integrated directly into these algorithms. Models for loudness perception have been applied directly in hearing instrument algorithms with two applications: to control the gain of a hearing instrument and to limit the maximum output power of a hearing instrument as a “loudness limiter.” Another example is the application of a very simple model in a hearing instrument to help resolve localization problems due to the microphone position in BTE hearing instruments, resulting in reduced front/back confusions. Examples of the application of models to the development and analysis of hearing instrument performance in real-life environments include the use of auditory scene analysis models to better understand human sound source classification and to improve the performance of automatic control systems in hearing instruments. A last example is the application of models describing human localization performance under realistic acoustic environments to understand the impact that hearing instrument signal processing has on the ability of hearing impaired people to localize sounds with hearing instruments. This talk will explore and discuss the potential of these different approaches for hearing instrument technology.

Contributed Papers

3:30

2pPP8. Moving fluid loading to the basilar membrane in passive nonlinear cochlear model. Svetlana Kovinskaya (Mechmath LLC, Prior Lake, MN 55372, mechmath@mechmath.com)

The mechanical response of the cochlea to sound is a nonlinear phenomenon. Existing models (with either passive or active outer hair cells) do not include effect of the motion of fluid loading to basilar membrane (BM). Being extremely slow this motion will play significant role only if the traveling wave in BM is close to the point of ceasing of propagation. Two fluid channels corresponding to the scalae vestibuli and tympani, each filled with incompressible liquid, provide the fluid loading to BM. Equation de-

scribing BM vibration with contra-directed moving fluid loading (from opposite sides of BM) becomes nonlinear equation because of an additional force proportional to loading velocity squared and the local curvature of BM (the second derivative of BM displacement). The velocity of this mass-like fluid motion is proportional to the applied sound power. In the case of sinusoidal excitation, the equation for BM vibration becomes the Mathieu equation with the characteristic equation determining stable and (parametrically) unstable regions. The nonlinear equation gives a multi-frequency quasi-steady state solution. Suggested mechanism (additional forces in the equation for BM vibration) explains the compression of strong stimuli, explains the generation of distortion products, and is in qualitative agreement with other experiments.

3:45

2pPP9. Predicting the effects of compression incorporating signal-adaptive band allocation. James M. Kates (GN ReSound, Dept. Speech Lang Hear Sci., Univ. of Colorado, 409 UCB, Boulder, CO 80309)

This paper considers the effects of dynamic-range compression on speech intelligibility and quality for hearing-impaired listeners. The paper focuses on a modification of compression in which the compression behavior for each frequency band is controlled by the peak amplitude within a frequency region surrounding the band. The stimuli are sentences in a background of multi-talker babble. Speech intelligibility and quality indices are used to model the listener responses for mild, moderate sloping, and moderate/severe hearing losses. The indices predict the effects of varying the compression parameters and the width of the frequency region, thus giving an indication of the expected benefit of the modified compression system and the optimal parameter settings as a function of hearing loss and listening conditions.

4:00

2pPP10. Prediction of binaural speech intelligibility when using non-linear hearing aids. Nicolas N. Ellaham (School of Inf. Tech. and Eng., Univ. of Ottawa, 800 King Edward Ave., Ottawa, ON K1N 6N5, Canada, nellaham@uottawa.ca), Christian Giguère (Univ. of Ottawa, Ottawa, ON K1H 8M5, Canada), and Wail Gueaieb (Univ. of Ottawa, Ottawa, ON, K1N 6N5, Canada)

A new objective measurement system is proposed to predict speech intelligibility in binaural listening conditions for use with hearing aids. Digital processing inside a hearing aid often involves non-linear operations such as clipping, compression, and noise reduction algorithms. Standard objective measures such as the articulation index, the speech intelligibility index (SII), and the speech transmission index have been developed for monaural listening. Binaural extensions of these measures have been proposed in the literature, essentially consisting of a binaural pre-processing stage followed by monaural intelligibility prediction using the better ear or the binaurally enhanced signal. In this work, a three-stage non-linear extension of the binaural SII approach is introduced consisting of (1) a stage to deal with non-linear processing based on a simple signal separation scheme to recover estimates of speech and noise signals at the output of hearing aids [Hagerman and Olofsson, *Acust. Acta Acust.* **90**, 356 (2004)], (2) a binaural processing stage using the equalization-cancellation model [Beutelmann and Brand, *J. Acoust. Soc. Am.* **120**, 331 (2006)], and (3) a stage for intelligibility prediction using the monaural SII [ANSI-S3.5, 1997 (R2007)]. Details of the new procedure will be discussed. [Research supported by NSERC (Canada).]

4:15

2pPP11. Predict the intelligibility of individual consonants in noise for hearing-impaired listeners. Feipeng Li, Woojae Han, and Jont Allen (Beckman Inst., Univ. of Illinois at Urbana, 405 N. Mathews Ave., Urbana, IL 61801)

The performance of hearing-impaired (HI) listeners to understand speech often deteriorates drastically in cocktail-party environments. To understand why a particular HI listener can hear certain sounds and not the others, it is critical that we take the prior knowledge about speech cues into consideration and investigate the effect of different types of cochlear hearing

loss on speech perception. In the last 2 years we have tested over 50 hearing-impaired ears on consonant identification in noise. To evaluate the impact of shift of hearing threshold on the intelligibility of individual consonants in masking noise, a consonant intelligibility index (CII) is developed to predict the perceptual scores of individual consonants. Results of preliminary study indicate that CII makes an accurate prediction for flat mild-to-moderate hearing loss, but fails for the cases of cochlear dead region and extremely unbalanced (e.g., severe high-frequency) loss, suggesting that audibility alone does not fully account for the hard of speech perception in noise for many HI listeners.

4:30

2pPP12. Evaluating hearing aid processing with an auditory model of modulation sensitivity. Frederick J. Gallun (Natl. Ctr. for Rehabilitative Auditory Res., U.S. Veterans Hospital Rd., Portland, OR 97239, Frederick.Gallun@va.gov), Pamela E. Souza, and Eric Hoover (Northwestern Univ., Evanston, IL 60208)

Amplitude modulation is one of the fundamental aspects of sound to which humans are sensitive. A number of auditory models based on this sensitivity have been successful at explaining other aspects of human behavior, both in the domains of psychoacoustics and speech perception. A constrained-parameter model of perceptual similarity based on modulation sensitivity (the spectral correlation index) has recently been developed for use in the prediction of perceptual confusions at the phonemic level [Gallun and Souza (2008)]. The relatively small parameter set of the model allows for efficient adaptation of the basic architecture to novel problems. With a single change in the range of one parameter, this model can be used for the evaluation of hearing aid processing by predicting the confusions that will accompany various choices of amplification parameters. This approach demonstrates the value of evaluating signal processing strategies in hearing aids through changes in perceptual similarity as predicted by auditory models. [Work supported by NIH NIDCD R01 DC0060014 and VA RRD CDA-2 C4963W.]

4:45

2pPP13. Modeling quality judgments of music stimuli reproduced by a simulated hearing aid. James M. Kates (Dept. Speech Lang Hear Sci., GN ReSound, Univ. of Colorado, 409 UCB, Boulder, CO 80309) and Kathryn H. Arehart (Univ of Colorado, Boulder, CO 80309)

The focus in hearing aids is generally on speech, but the ability to reproduce music is also important for listener satisfaction. This study uses a sound quality metric to predict judgments of music quality made by normal-hearing and hearing-impaired listeners. A simulated hearing aid was used to process music modified by (1) additive noise and nonlinear processing, (2) linear filtering, and (3) combinations of noise, nonlinear processing, and linear filtering. The music stimuli included jazz, classical, and vocalize selections. The quality metric incorporates a model of the auditory periphery that reproduces aspects of impaired hearing. The effects of noise and distortion on the signal are estimated by measuring the changes to the envelope time-frequency modulation pattern, and the effects of linear filtering are estimated by measuring the changes to the signal long-term internal spectral representation. The accuracy of the quality metric for music will be presented, and the performance of the metric for music will be compared to its accuracy in reproducing the quality ratings of speech stimuli processed under similar conditions.

Session 2pSA**Structural Acoustics and Vibration: Computer Modeling of Structural Acoustics and Vibration for Complex Structures**

James E. Phillips, Chair

*Wilson Ihrig and Associates, Inc., 5776 Broadway, Oakland, CA 94618-1531***Chair's Introduction—1:25*****Invited Papers*****1:30****2pSA1. A multiphysics model of hearing aid vibroacoustics using finite elements.** Thomas Burns and Dave Tourtelotte (Starkey Labs., Inc., Eden Prairie, MN 55344)

Electromagnetic balanced armature receivers [Hunt, *Electroacoustics*, Chap. 7] are used exclusively to generate acoustic output in hearing aids. These transducers are much more efficient than electrodynamic transducers and are capable of delivering upward of 140 dB of sound pressure to person. In an effort to maximize system gain in a hearing aid, vibroacoustical feedback paths originating from the receiver are modeled using finite elements. Given an electrical excitation, the electromagnetic-mechanical force on the armature is solved as a function of frequency. The force on this armature vibrates an internal diaphragm, which generates acoustic output while vibrating the entire hearing aid. Assuming that there are no acoustical leaks in the design, vibroacoustical coupling limits the usable gain of the aid. Using commercially available software, the fluid is modeled with full Navier–Stokes elements and is coupled to all structural boundaries. The armature is “kicked” with the aforementioned force, and spectral analysis is used to study mechanical transmissibility within and vibroacoustic reradiation around the hearing aid.

1:50**2pSA2. Finite element vibration analysis and measurement of a 255 ft vehicle ferry.** Jesse H. Spence and Ron E. Dempsey (Noise Control Eng., 799 Middlesex Tpke., Billerica, MA 01821)

Finite element analysis (FEA) methods have become part of the standard repertoire when designing a ship. This is particularly true for static analyses, although FEA methods are becoming increasingly utilized for analysis of low-frequency vibration as well. This paper discusses the FEA modeling process that was used for analyzing low-frequency (less than 80 Hz) habitability vibration on a 255 ft vehicle ferry. Deck vibration levels were measured after the vessel was built. Measured versus predicted vibration data show good correlation, particularly when the various sources of potential error and uncertainty are considered. The subject ferry was designed to meet the ABS Comfort Class (COMF) vibration criteria. Calculations of deck vibration were performed using NEiNastran to verify compliance with the criteria. Excitations due to propeller hull pressure excitation are highlighted here as these forces created the largest deck responses. However, it is noted that the main propulsion engines were resiliently mounted subsequent to initial analyses using the FE model. This paper discusses pertinent details of the model (element types, properties, etc.), application of hull pressures, damping, analysis methodology, and interpretation of results. Areas of uncertainty and impacts to the results are discussed.

2:10**2pSA3. An evaluation of residual vectors in the commercial finite-element program NASTRAN.** John B. Fahline (Water Tunnel Bldg., University Park, PA, 16802)

MSC NASTRAN introduced residual vectors in 2005. They are used to improve solution accuracy by representing the static contributions of truncated modes in transient and frequency response calculations. This formulation is convenient because the residual vectors are added to the modal vectors to form an augmented basis set, where the residual vectors are orthogonal to each other and to the original mode set. Modal frequency response calculations then proceed as usual only with a few extra vectors in the basis set. This talk will focus on evaluating the improvements in accuracy that can be achieved using residual vectors for modal frequency response problems where two structures with truncated mode sets are joined together.

2:30**2pSA4. Derivation of a Ritz series modeling technique for acoustic cavity-structural systems based on a constrained Hamilton's principle.** Jerry H. Ginsberg (School of Mech. Eng., Georgia Inst. of Technol., Atlanta, GA 30332-0405, jerry.ginsberg@me.gatech.edu)

Hamilton's principle for dynamic systems is adapted to describe the fully coupled interaction of a confined acoustical domain and a bounding elastic system in cases of imposed velocity or pressure fluctuations on the surface or forced excitation of the structure. A key part of the modified principle is the treatment of the surface traction as a Lagrange multiplier function that enforces continuity conditions at the fluid-solid interface. The structural displacement, fluid velocity potential, and traction are represented by Ritz series, where the usage of the velocity potential as the state variable for the fluid assures that the flow is irrotational. Designation of the coefficients of the potential function series as generalized velocities leads to corresponding series representations of the particle velocity, displace-

ment, and pressure in the fluid, which in turn leads to descriptions of the mechanical energies and virtual work. Application of the calculus of variations to Hamilton's principle yields linear differential-algebraic equations whose form is identical to those governing mechanical systems that are subject to nonholonomic kinematic constraints. Criteria for selection of basis functions for the various Ritz series are illustrated with an example of a rectangular cavity bounded on one side by an elastic plate and conditions that change discontinuously on other sides.

2:50

2pSA5. Acoustic color of viscoelastic objects in two-fluid environments: High-fidelity, high-speed, three-dimensional finite-element modeling. David S. Burnett (Naval Surface Warfare Ctr. Panama City Div., 110 Vernon Ave., Panama City, FL 32407, david.s.burnett@navy.mil)

NSWC PCD has developed a computer simulation system for modeling the acoustic color (target strength versus frequency and aspect angle) of realistic three dimensional (3-D) objects that are near to or straddling the interfaces between different fluids. It employs high-fidelity (fully 3-D physics throughout object and environment), finite-element (FE) modeling of acoustic scattering from viscoelastic objects. It is implemented in a scalable-architecture, multiblade rack system that automatically runs hundreds of thousands of FE models, changing the mesh resolution and outer fluid boundaries of the models as they sweep over frequency. This frequency-dependent modeling yields a frequency-independent modeling error for the acoustic color response. This, in turn, yields higher accuracy for subsequent signal processing of the acoustic color data. The first part of this paper will present a brief overview of the theoretical, numerical, and software issues. The second part will present verification and validation of the system: (1) ultra-high-precision correlation with T -matrix simulations for four canonical target/environment configurations and (2) experimental validation for objects in free space and straddling the interface between two different fluids. [Work supported by ONR and SERDP.]

3:10—3:25 Break

3:25

2pSA6. Energy finite element analysis of naval vehicles. Nickolas Vlahopoulos (Dept. of Naval Architecture and Marine Eng., Univ. of Michigan, 2600 Draper Rd., Ann Arbor, MI 48109, nickvl@umich.edu) and Kuangcheng Wu (Northrop Grumman Shipbuilding, Newport News, VA 23607)

Simulation methods can be used for meeting both the new acoustics and vibration initiatives for surface ship silencing. An energy finite element analysis (EFEA) formulation has been developed for computing efficiently the vibration of complex structures and the associated radiated noise at frequencies beyond the limits of conventional finite element methods. The EFEA provides a very attractive alternative to the statistical energy analysis (SEA) because it uses the same finite element model which is utilized for other simulations (i.e., structural analysis), while the SEA requires the development of a completely new model based on a lumped parameter approach. Further, the SEA requires estimates of modal densities for the various subsystems which introduce difficulties and approximations in the solution. Thus, the EFEA streamlines the structural-acoustic analysis within the ship design process due to modeling commonality with other disciplines. In this presentation, the main theoretical aspects of the EFEA will be presented first. Results presented for two 1/8th scale structures representing an advanced double hull design and a conventional hull design of a surface ship are analyzed. Results for the structural vibration induced on the outer bottom part of the structures are compared to available test data.

Contributed Papers

3:45

2pSA7. Engaging structural-acoustic simulations in multi-discipline design. Nickolas Vlahopoulos (Univ. of Michigan, 2600 Draper Rd., Ann Arbor, MI 48108, nickvl@umich.edu) and Jim He (Michigan Eng. Services, LLC, Ann Arbor, MI 48108)

In order to be effective and maximize the weight and cost savings, designing for noise and vibration attributes must be performed in parallel when designing a complex system for other disciplines (i.e., durability, crashworthiness, etc.). The system characteristics influence the performance in all the different attributes. Challenges arise when designing a system for improving mutually competing objectives, satisfying constraints from multiple engineering disciplines, and determining a single set of values for the vehicle characteristics. This presentation discusses an approach that conducts optimization analysis for a complex system by coordinating operations and exchange of data and information through a network of optimizations. A rotorcraft application is analyzed for structural-acoustic and crashworthiness. The hybrid finite element analysis (hybrid FEA) comprises the simulation method used for the structural-acoustic simulation. It combines conventional FEA with energy FEA for modeling the response of a system comprised by stiff, load bearing members, and flexible panels. The LS-DYNA solver is used for the crash-landing simulations. The weight of the gear box foundation is optimized while simultaneous improvements are pursued through parallel

driven optimizations that address the input power into the system from gearbox excitation and the dynamic stresses encountered in the gearbox foundation from crash landing.

4:00

2pSA8. SALINAS: A massively parallel finite element code for structural dynamics and acoustics analysis. Jerry W. Rouse (Dept. Analytical Structural Dynam., Sandia Natl. Labs., Albuquerque, NM 87185, jwrouse@sandia.gov), Timothy F. Walsh, and Garth M. Reese (Sandia Natl. Labs., Albuquerque, NM 87185)

This talk shall present an overview of SALINAS, a massively parallel finite element code for structural dynamics and acoustics analysis that is being developed at Sandia National Laboratories. SALINAS allows for prediction of both the time and frequency domain responses of complex structural, acoustic, and fully coupled structural acoustic systems having millions of degrees of freedom. An overview of SALINAS capabilities shall be presented including development history, solver and element types, quadratic eigenanalysis and frequency response, direct frequency response, nonlinear acoustics, implicit transient dynamic analysis, and infinite elements with focus given to structural acoustics capabilities. The application of SALINAS to structural acoustics problems shall also be presented as well as future directions of research for the development of the code.

2pSA9. Numerical study of structural intensity for a model airframe structure in healthy and damaged states. Micah R. Shepherd, Stephen C. Conlon (Appl. Res. Lab., Penn. State Univ., P.O. Box 30, State College, PA 16804), and Fabio Semperlotti (Penn. State Univ., State College, PA 16804)

Structural damage due to fatigue or material flaws presents a major challenge for many engineering applications. To address this challenge, structural health monitoring techniques have been developed in an effort to determine the type, severity, and location of the damage. Many of these techniques are based on the nonlinear contact phenomenon. In this paper, structural intensity methods are used as a damage-detection metric on a sim-

plified airframe structure. A finite-element model was created and the structural intensity field was mapped over the surface using individual element power flow estimates. Damage was then introduced into the model by removing connectivity between elements and including nonlinear stress-strain relationships and contact at the damage location. Structural intensity maps were then compared to the healthy case to correlate the changes with the damage state. More specifically, the increase in structural intensity at the sub- and super-harmonics was found to be indicative of the damage state. The results of this numerical study show the potential of surface structural intensity in autonomous damage detection for flight vehicles. The implications of this type of damage detection metric with respect to damage size and location will also be discussed.

TUESDAY AFTERNOON, 20 APRIL 2010

ESSEX A/B/C, 1:30 TO 4:30 P.M.

Session 2pSC

Speech Communication: Speech Production (Poster Session)

Benjamin R. Munson, Chair

Univ. of Minnesota, Dept. of Speech, Language, Hearing Science, 164 Pillsbury Ave., SE, Minneapolis, MN 55455

Contributed Papers

All posters will be on display from 1:30 p.m. to 4:30 p.m. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 1:30 p.m. to 3:00 p.m. and contributors of even-numbers papers will be at their posters from 3:00 p.m. to 4:30 p.m.

2pSC1. A procedure for estimating gestural scores from articulatory data. Hosung Nam (Haskins Labs., New Haven, CT 06511, nam@haskins.yale.edu), Vikramjit Mitra (Univ. of Maryland, College Park, MD), Mark K. Tiede (Haskins Labs., New Haven, CT), Elliot Saltzman (Boston Univ., Boston, MA), Louis Goldstein (Univ. of Southern California, Los Angeles, CA), Carol Espy-Wilson (Univ. of Maryland, College Park, MD), and Mark Hasegawa-Johnson (Univ. of Illinois at Urbana-Champaign, Urbana, IL)

Speech can be represented as a set of discrete vocal tract constriction gestures (gestural score) defined at functionally distinct speech organs [tract variables (TVs)]. Using such gestures as sub-word units in an ASR system, variation in speech arising from coarticulation and reduction can be addressed. Since there is a lack of test corpora annotated with gestural scores, we develop a semi-automatic procedure for estimating and annotating gestural scores from natural speech databases using the Haskins speech production model (TaDA). We first describe the procedure's application to 500 words with unique phone sets found in the Wisconsin x-ray microbeam database, generating both gestural scores and synthetic speech. Second, we perform dynamic time warping (DTW) to align the TaDA-generated speech signals with respect to the microbeam data. The DTW time-scaling pattern is then used to adjust the gestural score originally input to TaDA to generate new time-warped TaDA acoustics. Third, we fine-tune the gestural timing patterns within the new gestural score in a constrained manner to generate the score whose associated acoustics are minimally different from the database acoustics. Finally, we evaluate the optimized gestural score's validity by comparing its associated TV trajectories to those from the original microbeam database.

2pSC2. Prosodic interaction between speakers of American and British English. Jelena Krivokapic (Dept. of Linguist., Yale Univ., 370 Temple St., 302, New Haven, CT 06520-8366, jelena.krivokapic@yale.edu)

Previous work has found that interaction between speakers influences their production, in that phonetic similarity of the speakers to each other increases [Sancier and Fowler, *J. Phonetics* **25**, 421–426 (1995); Pardo, *J.*

Acoust. Soc. Am. **119**, 2382–2393 (2006); and Tobin, *J. Acoust. Soc. Am.* **125**, 2757 (2009)]. These studies have mostly focused on segmental properties of speech. Little is known, however, about how prosodic properties of speech are affected in conversational interaction. An experiment is presented that examines the interaction between native speakers of British English and American English. Using the synchronous speech paradigm [Cummins, *ARLO* **3**, 7–11 (2002) and Zvonik and Cummins, in *Proceedings of the Eurospeech 2003* (2003), pp. 777–780] where two speakers read sentences at the same time, we examine how speakers influence each other at the prosodic level. Eight subjects (4 dyads), each consisting of one British and one American speaker, read the following: a short story that contained four words where the two dialects differ in stress pattern, 13 target words where the dialects differ in vowels, and 84 sentences with varying intonation patterns (question, statement, and focus). The goal was to examine how speakers adjust to each other's speech, and whether certain prosodic properties are more susceptible to adjustment to the co-speaker's production than others.

2pSC3. Prosodic boundaries in German: Final lengthening in spontaneous speech. Susanne Fuchs (ZAS, Schuetzenstrasse 18, 10117 Berlin, Germany fuchs@zas.gwz-berlin.de), Jelena Krivokapic (Yale Univ., New Haven, CT 06520-8366), and Stefanie Jannedy (ZAS, Schuetzenstrasse 18, 10117 Berlin, Germany)

Most theories of prosodic structure postulate at least two phrasal categories above the word level, a minor and a major one. One correlate of phrasal boundary marking is lengthening on the right edge of a phrase. To gain a theory neutral understanding of the nature of prosodic boundaries, a Gaussian mixed model (GMM) was applied to durational data, estimating the underlying clusters of a continuous distribution. Spontaneous speech data were collected with standardized interviews from 19 adolescent speakers of a multi-ethnolect (Kiezdeutsch, Hood German) spoken in Berlin, orthographically transcribed and added to a database that allowed for searches of the particle "so" (so or like) in different conditions. Acoustic durations of /z/ and /o/ of so in phrase final position (as determined by orthographical markings or by following pauses) were labeled and z-transformed per speaker. The

results show that a model including two clusters accounts best for the data. Nevertheless, there is an overlap between the two distributions supporting the gradient nature of boundaries. Further analyses with more tokens of so including word level junctures are currently under way. The impact of our data-driven results on theoretical models of prosodic boundaries is discussed. [Supported by a grant from BMBF.]

2pSC4. Prosodic analysis of subglottal pressure contours. Helen M Hanson (Dept. ECE, Union College, 807 Union St., Schenectady, NY 12308, helen.hanson@alum.mit.edu), Stefanie Shattuck-Hufnagel (MIT, Cambridge, MA 02139), and Angela McLelland (Union College, Schenectady, NY 12308)

In previous work the problems of identifying final fall and estimating declination of the subglottal pressure contour were addressed [Hanson *et al.*, *J. Acoust. Soc. Am.* **118**, 2027 (2005); **120**, 3090 (2006)]. Additional methods for identifying final fall have since been explored. These new methods will be described, and the identification of final fall will be compared across several of the methods. The use of these additional methods has meant that the complete data set for three subjects can be analyzed. Therefore it is worth revisiting the work presented in 2006. Objective measures of subglottal pressure declination will be evaluated with respect to subjective ratings of declination. The effects of phrase and boundary tones on declination rate will be re-examined, as will the effects of pitch accent distribution. [Work supported by NIH Grants DC00075 and DC04331 and by a Union College summer research grant to A.M.]

2pSC5. Effects of metrical structure on syllable timing II. Megan Tinius and Marios Fourakis (Dept. of Communicative Disord., UW-Madison, Madison, WI 53706)

Fourakis and Monahan [J. Linguist. (1988)] examined the effects of two different types of metrical foot (iamb and anapest) on syllable timing in 3 ft utterances. These utterances were comprised of three iambs, three anapests, and every possible combination of the two types. The present research follows the same paradigm but uses trochees and dactyls instead of iambs and anapests. A set of eight 3 ft long sentences was constructed. At one end there was the three trochee sentence: Auctions rescue business. At the other end was the three dactyl sentence: Auctions may rescue his businesses. The intermediate six sentences were formed by different combinations of the metrical feet. Five male and five female native speakers of Midwestern American English were recorded producing five repetitions of each sentence at a normal rate of speech. Segments were assigned syllable membership based on the maximum onset principle. Results will be presented on syllable durations as a function of foot type and sentence type. [Research supported by the University of Wisconsin Madison Graduate School.]

2pSC6. Temporal structure in the speech of persons with dementia: A preliminary study. Linda Carozza (Dept. of Commun. Sci. and Disord., St. John's Univ., 300 Howard Ave., Staten Island, NY 10301), Margaret Quinn, Jacqueline Palladino, and Fredericka Bell-Berti (St. John's Univ., Jamaica, NY 11439)

Although cognitive and language processes in dementia have been studied extensively, the question of motor speech degeneration in the course of dementing illness is a relatively unexplored area. The potential for early dissociation of motor functions of language at the level of speech production has not been explored; an interaction between motor speech and language production and perception changes should inform our understanding of the deterioration in dementia. If early motor declines are found, they may provide an additional early marker and contribute to early detection, giving further insight to the character of neurological deterioration in this disease. This study explores the temporal structure of speech in individuals in the mild-to-moderate stages of the disease. In this preliminary study, participants were asked to produce a series of four-word phrases containing target word in phrase-medial or phrase final position. The target words contained the vowel in "heed," "head," or "had," and began with a fricative or voiced or voiceless stop consonant and ended with either /t/ or /d/. Thus, we are able

to examine four aspects of the temporal patterns of speech in these persons: intrinsic vowel duration, VOT, phrase-final lengthening, and the effect of final consonant voicing on vowel duration.

2pSC7. Japanese language learners' production of short/long vowel and single/geminate consonant contrasts: A longitudinal development. Yukari Hirata and Ian C. McNally (Dept. of East Asian Lang. and Lit, Colgate Univ., 13 Oak Dr., Hamilton, NY 13346, yhirata@colgate.edu)

This longitudinal study examined the development of native English speakers' production ability to distinguish between single/geminate consonant and short/long vowel pairs in Japanese over 4 months abroad in Japan. Seven participants, who had formal Japanese language classes for 2 years in the US, recorded rika(a) and ka(k)ko pairs in a carrier sentence three times each, at three speaking rates, before and after the study abroad (pre-/post-Japan). Seven measures, including the duration of contrasting segments, words, and sentences, were taken and various ratios were also calculated. Of the native Japanese speaker criteria used to evaluate participants' data, the vowel-to-word and closure-to-word duration ratios of 0.38 and 0.35 were used as boundary ratios between short and long segments [Hirata (2004); Hirata and Whiton (2005)]. Using these criteria, participants' data were classified at 66% accuracy at pre-Japan but improved to 85% at post-Japan for vowel distinction. In contrast, they improved from 73% to 74% accuracy for consonant distinction. Analyses of other durational measures also supported that participants made significant improvements in vowel distinction, but not in consonant distinction. These results support the claims of Toda (1997) and Han (1992) and extend to include intermediate language learners into their respective studies.

2pSC8. Modifying speech to children based on perceived developmental level: An acoustic study of adults' fricatives. Hannah M. Julien, Benjamin Munson (Dept. of Speech-Lang.-Hearing Sci., Univ. of Minnesota, 164 Pillsbury Dr., Minneapolis, MN 55455 julie006@umn.edu), Jan Edwards (Univ. of Wisconsin, Madison, WI 53705), Mary E. Beckman, and Jeffrey J. Holliday (Ohio State Univ., Columbus, OH 43210)

Recent acoustic studies have shown that children learning English gradually differentiate between the sounds /s/ and /ʃ/ [Li *et al.* (2009)]. Other studies have shown that adults can perceive different degrees of accuracy in these phonemes when given a non-categorical response modality, such as a visual-analog scale [Urberg-Carlson *et al.* (2008)]. This presentation combines a VAS rating task with a production task. Adults were presented with children's productions of target fricatives. They rated the accuracy, after which they were instructed to produce the target sequence as if they were responding to the child whose speech they had just rated. Ongoing analyses examine whether adults respond differently to productions that they have rated as intermediate between /s/ and /ʃ/ and whether adults produce hyperarticulated version of fricatives in response to productions that they rated as not canonically /s/ or /ʃ/, as opposed to ones that they rated as adult-like. The results of this project will be used in supervised learning models of the acquisition of /s/ and /ʃ/ and will inform us on the extent to which adults calibrate their speech to children depending on the perceived level of speech mastery. [Work supported by NIDCD Grant 02932 and NSF Grant BCS0729277.]

2pSC9. Palatalization in infant-directed speech. Jenesia McCammon and Nan Bernstein Ratner (Dept. of Hearing and Speech Sci., Univ. of Maryland, 0100 Lefrak, College Park, MD 20742, jmcammon518@gmail.com)

Adult-directed speech (ADS) is characterized by frequent use of phonological rules such as palatalization ("did you" becomes "didju"). Use of such rules should blur word boundaries, making it more difficult for infants to identify words in the input. To date, there are conflicting data on the degree to which infant-directed speech (IDS) makes more or less frequent use of these rules [Shockey and Bond (1980) and Bernstein Ratner (1986)]. We are following a cohort of children from 7 months to 2 years. We examined whether rules such as palatalization are used more or less frequently in IDS by 20 mothers to their children and to an adult listener. To date, there are more opportunities for palatalization in the mothers' IDS than ADS to pre-verbal children, and the rule is more frequently applied in IDS. In this case, IDS is not more acoustically clarified than ADS. Research suggests that IDS

clarification may be a function of the child's language stage. We would therefore not expect reduction in palatalization, which primarily affects front vowels, until a later stage of the child's language development.

2pSC10. Voice onset time in infant-directed speech at 7.5 and 11 months.

Anna Synnøstvedt, Nan Bernstein Ratner, and Rochelle Newman (Dept. of Hearing and Speech Sci., Univ. of Maryland, 0100 Lefrak Hall, College Park, MD 20742)

Studies have reported differences between infant-directed speech (IDS) and adult-directed speech (ADS), suggesting that mothers adjust speech to their infants in ways that may help children better parse the incoming acoustic signal. One aspect of IDS that has been examined is voice onset time (VOT). Results have been inconsistent, revealing longer VOT in IDS [Englund (2005); Malsheon (1980)], shorter VOT in IDS [Sundberg & Lacerda (1999)], or no difference in VOT between ADS and IDS [Baran *et al.* (1977)]. Characteristics of IDS may also depend on the language maturity of the child and, therefore, clarity of speech may vary across stages of development [Englund & Behne (2006)] as well as vary among mothers. The present study examines 15 mothers' VOT in IDS to children at 7.5 months old and again at 11 months as compared to their VOT in ADS. Words with initial stop consonants that occurred in both IDS and ADS conditions are analyzed using PRAAT. Variability was observed among mothers and age effects: preliminary data suggest that VOT in IDS at 11 months is better differentiated than at 7.5 months. [Work supported by NSF BCS0745412.]

2pSC11. Mandarin and English initial stops produced by Mandarin-English bilinguals. Jia-Shiou Liao (School of Appl. Foreign Lang., Chung Shan Medical Univ., No. 100, Sec. 2, Daching St., South District, Taichung City 402, Taiwan, jsliao@csmu.edu.tw)

This study investigates the relationship between the way Mandarin-English bilinguals and native speakers of English pronounce English initial stop consonants, and between the way Mandarin-English bilinguals pronounce Mandarin and English stops, with respect to voice onset time. Twelve Mandarin subjects who had learned English as their second language and were studying at a US university were asked to read randomized lists of 72 Mandarin characters and of 72 English words; twelve English subjects read only the list of English words. The syllable structure of each Mandarin character and of each English word in this study was CV. The Mandarin consonants were initial aspirated and unaspirated /p, t, k/; the English consonants were initial /p, t, k/ and /b, d, g/. The vowels used in both lists were high vowels. Significant differences are found between the Mandarin-speaking subjects' articulation of all the Mandarin stops and their minimally paired English equivalents produced by the English-speaking subjects. Significant differences are found between Mandarin and English speakers' articulation of the same English stops, except for that of /t/. When comparing the Mandarin-speakers' articulation of Mandarin stops and their minimally paired English counterparts, a significant difference is found only between English and Mandarin aspirated /p/.

2pSC12. Phonetic adaptation in non-native speech: Insights from a distributional analysis of long-lag voice onset time. Marisa Pineda and Meghan Sumner (Dept. of Linguist., Stanford Univ., Margaret Jacks Hall, Bldg. 460, Stanford, CA 94305-2150, middy@stanford.edu)

Speakers encountering long-lag voice onset time (VOT) for the first time in their L2 produce VOTs between their L1 and L2 values. Native-like long-lag productions are conditioned by speaker competency factors such as age of acquisition and experience, showing significant production differences between late bilinguals, early bilinguals, and native L2 speakers. Thus far, analyses have focused on average VOTs across speaker groups. This project investigates the full distributional properties of VOT in bilinguals (e.g., variation and skewness) in addition to averages, providing a more informative picture of bilingual acquisition. VOT production data were collected from French-English bilinguals (age of English onset 0–15 years) and submitted to a distribution-based analysis. Results show that while speaker groups differ predictably in mean VOT, the analysis discovers subgroups based on common production behaviors, eliminating the gross categorizations of early and late bilinguals and moving toward gradual, predictable shapes of VOT that are correlated with English schooling, time in an

English-speaking country, and age of acquisition. Ultimately, these results may fill in some missing blanks between perception and production, suggesting, for example, that differences in perceived accent and comprehensibility diverge due to degree of overlap with native L2 VOTs.

2pSC13. A Bayesian model of voice-onset time production. Mark VanDam (Boys Town Natl. Res. Hosp., 555 N 30 St., Omaha, NE 68131, mark.vandam@boystown.org) and Noah Silbert (Indiana Univ., Bloomington, IN 47405)

Talkers performed a listen-and-repeat task to investigate temporal detail in voice-onset time (VOT) productions of American English word-initial stop consonants. Experimental factors included linguistic context (isolation, carrier phrase, unfamiliar phrase, and familiar phrase), usage frequency (high and low), lexical status (word and non-word), training (baseline and posttraining), and posttraining generalization (test words and novel words). For each context, frequency, and lexical status, baseline VOT production estimates were collected, then a naive training regimen conducted, then post-training estimates of both test words and novel words were obtained. Testing novel words explored whether the effect, if obtained, generalized throughout the lexicon. A Bayesian linear model (analogous to analysis of variance) was used to model VOT means as a function of these factors. Posterior distributions of modeled VOT means were compared across six talkers, with a focus on probing the relationships between lexical frequency and status, linguistic context, and training. Preliminary results suggest that a number of these experimental factors influence fine-grained control of VOT production. As expected from speech production data obtained from multiple talkers and thousands of productions, the overall model fit was imperfect, but the results indicate that a Bayesian model can be productively deployed for data exploration into temporal aspects of speech production.

2pSC14. Priming at the level of phonetic detail: Evidence from voice onset time (VOT). Susannah Levi and Jennifer Bruno (Dept. of Communicative Sci. and Dis., NYU, 665 Broadway, 9th floor, New York City, NY 10012, slevi@nyu.edu)

Priming studies have shown effects on the production of target words when the target and prime are identical and when the target and prime share some but not all phonological features. Variation at the level of phonetic detail has also been found to affect production in word shadowing tasks where speakers produce longer VOTs when the VOT of the sample word is artificially lengthened. In this study, we examine the effect of naturally produced, yet systematic within-category variability in the VOT of the prime on the production of a target word for visually presented words. Primes either contained a voiceless stop with a long VOT ("keen") or a short VOT ("pan"). The target words contained a voiceless alveolar stop followed by five vowels which varied in height and backness (e.g., "tune" and "ten"). Results showed that the same target word was produced with longer VOTs when the prime contained a naturally long VOT than when the prime contained a naturally short VOT. These results suggest that speakers are sensitive to natural within-category variability, and that this sensitivity affects production across different segments.

2pSC15. Why are Korean tense stops mastered early: Evidence from production and perception. Eun Jong Kong (Waisman Ctr., Univ. of Wisconsin at Madison, 1500 Highland Ave. #493, Madison, WI 53705, ekong@wisc.edu) and Mary E. Beckman (The Ohio State Univ., Columbus, OH 43210)

Transcription-based studies have found that Korean-acquiring children master tense stops earliest among the three different types of homorganic stops (tense vs lax vs aspirated) despite its phonologically marked status. Tense stops in Korean have a short-lag VOT, so this finding is consistent with previous cross-linguistic research on order of acquisition of stop phonation types. However, Korean tense, lax, and aspirated stops are also differentiated by the fundamental frequency and the voice quality at the vocalic onset in addition to VOT. This study examined how correctly these multiple acoustic cues (VOT, f₀, and H1-H2) predict the mastery pattern of Korean stops. The effect of these acoustic parameters on 20 Korean adult listeners' assessments of children's (aged 2–6) productions of /tʰ/, /t/, and /th/ was analyzed. Listeners were asked to label the stimuli as one of three stop cat-

2p TUE. PM

egories and then to rate their goodness using visual analog scaling. Both categorical and gradient assessments of children's stops were analyzed based on the three acoustic parameters in a mixed-effect regression model in order to quantify the relative effects of acoustic parameters on adult perception of phonation. [Work supported by NIDCD 02932 and NSF Grant 0729140.]

2pSC16. "Probably, OK, whatever!": Variability in conversational speech stops and flaps. Natasha Warner (Dept. Linguist., Univ. of Arizona, Box 210028, Tucson, AZ 85721-0028, nwarner@u.arizona.edu) and Benjamin V. Tucker (Univ. of Alberta, Edmonton, AB T6G 2E7, Canada)

In casual, conversational speech, speakers often do not fully realize all segments one would normally expect in a given word. For example, "better" may be pronounced with an approximant instead of a medial flap (if the tongue fails to make a complete closure at the alveolar ridge), the expected flap may be entirely deleted, or the segments may be so heavily overlapped that it is almost impossible to say which segments are present and which not. The current work investigates how much reduction occurs (including both changes and deletions), comparing casual conversations, connected reading, and isolated word-list reading. It compares the phonemes /p, t, k, b, d, g/, post-stress vs inter-unstressed, and six segmental environments. The results represent 13 native English speakers and comprise a dataset of more than 9000 tokens, with several acoustic measures. This provides very thorough data on phonetic variability in both natural and "lab" speech. The work finds relatively little effect of stress and complex effects of segmental environment attributable to places of articulation. It also provides insight into how variable speakers are in how much they reduce, both within a speaker and within the population, a topic not often addressed in phonetic research.

2pSC17. Functional anatomy of the human tongue: Review and implications for the development of speech production. Margaret Denny and Richard S. McGowan (CReSS LLC, 1 Seaborn Pl. Lexington, MA 02420)

To acquire speech children must attempt to reproduce adult acoustic models using immature vocal tracts which differ from those of adults in complex ways. Control of the tongue is crucial for acceptable speech, yet the tongue is an understudied articulator because it is anatomically complex and difficult to record from using kinematic or electrophysiological methods. Nevertheless recent advances in tongue anatomy and physiology, combined with knowledge of vocal tract development, promise to shed light on the problems that children must solve in learning to speak as well as the solutions available to them. This review aims to integrate and present these findings to an audience interested in the development of speech production. A striking finding is that the tongue shows an anterior-to-posterior gradient in muscle fiber sizes and types. In multiple muscles, faster fibers are more common anteriorly while slower fibers are more predominant in the posterior tongue. Within fiber type (fast or slow) smaller fibers are seen anteriorly and larger ones posteriorly. This implies that rapid fine positioning and shaping may be particularly challenging when it involves the posterior tongue. [Work supported by NIDCD-0001247 to CReSS LLC.]

2pSC18. Tracking muscle deformation during speech from tagged and diffusion tensor magnetic resonance imaging. Xiaofeng Liu (Dept. of Comput. Sci., Johns Hopkins Univ., 1600 N. Charles St., Baltimore, MD 20017, xiaofeng.liu@gmail.com), Sudarshan Ramenahalli (Johns Hopkins Univ., Baltimore, MD 20017), Hideo Shinagawa, Maureen Stone (Univ. of Maryland Dental School, Baltimore, MD 21210), Jerry L. Prince (Johns Hopkins Univ., Baltimore, MD 20017), Emi Z. Murano (Johns Hopkins Hospital, Baltimore, MD 21287), Jiachen Zhuo, and Rao Gullapalli (Univ. of Maryland Med. School, Baltimore, MD 21210)

Oral head and neck cancer is the sixth most common cancer worldwide. These tumors are usually treated by surgical removal of the affected tissue. The result of the surgery is a loss of muscle tissue, accompanied by scarring, reduced strength, and often reduced function. This paper explores the relationship between tongue muscle orientation and muscle deformation pattern in normal and post-cancer surgery speakers. Our motivation is the need to better understand the mechanisms that underlie tongue motion, in order to better interpret clinical observations and to provide data that can help predict optimal surgical outcomes. This study uses diffusion tensor imaging (DTI)

to extract muscle fiber orientation, but it is not possible to image the muscle deformation in real-time using DTI. On the other hand, tagged MRI can track muscle fiber bundles in real time, but the original fiber bundle is not visible. Therefore, this work develops a method to overlay muscle bundles from the DTI onto the corresponding locations in tagged MR images and track muscle deformation. The mechanical properties of the muscles are calculated, such as translation and bending. Motion pattern differences between the normal and glossectomy tongue are detailed. [NIH R01-CA133015 and K99-DC9279.]

2pSC19. Dependency of compensatory strategies on the shape of the vocal tract during speech perturbed with an artificial palate. Jana Brunner (Res. Lab. of Electron., MIT, 50 Vassar St., Cambridge, MA 02139, jbrunner@mit.edu), Phil Hoole (Ludwig-Maximilians-Universität München, 80799 München, Germany), Frank H. Guenther (Boston Univ., Boston, MA 02215), and Joseph S. Perkell (MIT, Cambridge, MA 02139)

When adapting to an articulatory perturbation, different speakers may use different strategies. This study investigates the hypothesis that the choice of strategy depends on vocal tract shape. Vowel production was perturbed with prostheses that changed palatal shape. Three speakers had a prosthesis that effectively moved the alveolar ridge posteriorly; three others had a prosthesis that effectively flattened the palate. EMA recordings of speakers' adaptive behavior showed that during the production of the vowel /y/, the speakers with the alveolar prosthesis compensated by using increased lip protrusion. The speakers with the flat prosthesis used about the same amount of lip protrusion as without perturbation. We hypothesize that this is because for the first group of speakers the main constriction in the vocal tract was moved forward due to the changed palatal outline: the front cavity became too short, which was compensated for by more lip protrusion. This hypothesis is being explored by determining whether the DIVA speech production model will produce compensations like the subjects when the model's vocal tract is adapted to fit individual vocal tract shapes using detailed information about the subjects' vocal tracts from MRI recordings. Complete results will be reported and discussed. [Work supported by DAAD, NIH.]

2pSC20. Can an ultrasound tongue be overlaid on an MRI (magnetic resonance imaging) vocal tract? Jim Lee (jima@yahoo.com) and Maureen Stone (Dept. of Neural and Pain Sci. and Dept. of Orthodontics, Univ. of Maryland Dental School, Baltimore, MD 21210)

Ultrasound image sequences and cine-MRI time-frames provide intersecting but not identical data about the articulation of speech. It would be valuable to be able to combine ultrasound and MRI datasets in a way that maximizes their strengths. Midsagittal MRI images the entire vocal tract; ultrasound images the tongue surface. However, MRI is not portable, requires multiple repetitions, and supine subject positioning. Thus insertion of the tongue from ultrasound movies onto the MRI vocal tract would provide tongue motion in a head coordinate system. This study aligns supine ultrasound tongue contours with MRI tongue contours of the same tasks using (1) rigid alignment of the tongue contours frame-by-frame and (2) rigid alignment of palates, whose alignment parameters are then applied to tongue contours. rms differences compare the error between methods. The value of this study is that if palatal alignment yields a comparable error to tongue alignment, then palatal alignment could be used to insert an ultrasound data set into a vocal tract outline that has no tongue motion data associated with it. [Funding support: R01CA133015.]

2pSC21. An articulatory, acoustic, and perceptual study of the effect of a short lingual frenulum on flaps in late Korean-English bilinguals. Kyung-Ah Kim (Dept. of Linguist., Cornell Univ., 203 Morrill Hall, Ithaca, NY 14853, kk339@cornell.edu)

Having a short lingual frenulum is known to limit tongue mobility, often resulting in nonstandard pronunciations of certain coronal sounds. While previous studies have largely focused on the physical and linguistic characteristics of the most extreme cases and on the effects of surgical intervention [Hagiwara *et al.* (2002); Lee *et al.* (2008)], the current study investigates the articulatory, acoustic, and perceptual characteristics associated with Korean speakers with a relatively short lingual frenulum but not short enough to

greatly impede communication. The study focuses on the flap—the single most problematic sound for Korean speakers with short lingual frenula—produced in both Korean and English by both “short-tongued” and normal late Korean-English bilinguals. Flaps are studied through palatographic, linguographic, and spectrographic analyses, while a perception test is performed to pinpoint the categorical boundary between the normal and short-tongued flaps. Preliminary results suggest that the short-tongued speakers are better able to accurately produce the English flap than the Korean flap, due to language-specific phonetic and phonological characteristics. This finding bears on whether surgical intervention is necessary to help normal (and mildly short-tongued) speakers of Korean improve their English pronunciation [see Ko (unpublished)].

2pSC22. Resonance-based source-filter separation for speech. Ivan Selesnick (Dept. of Elec. and Comput. Eng., Polytechnic Inst. of New York Univ., 6 Metrotech Ctr., Brooklyn, NY 11201) and Chin-Tuan Tan (New York Univ., New York, NY 10016)

This paper proposes the application of resonance-based signal analysis to the decomposition of speech into “source” and “filter” waveform components. Resonance-based signal analysis, developed by Selesnick,¹ is a novel analysis method that decomposes a signal into high-resonance (oscillatory) and low-resonance (transient) components. The original signal is given as a sum of the two components. The method utilizes sparse signal representations in the domain of two invertible constant- Q transforms, having low- and high- Q -factors, respectively. The constant- Q transforms are implemented as rational-dilation wavelet transforms. Applied to speech, the method provides a separation that cannot be achieved with other known filtering techniques. Current source-filter modeling of speech assumes a known linear filter which sufficiently estimates vocal tract characteristics and can be derived from a segment of stationary speech. The source glottal waveform is then extracted by inverse filtering. In contrast, resonance-based signal analysis directly decomposes speech into its source and filter waveforms without such assumptions. The outcome may provide a new perspective in speech analysis, synthesis, recognition, and coding. This paper illustrates the efficacy of this signal analysis technique with vowels; further investigation with other speech elements is in progress.

2pSC23. Speech separation based on multi-level segmentation. Jose J. Lopez and Maximo Cobos (ITeam, Univ. Politecnica Valencia, Camino de Vera, 46022 Valencia, Spain, jjlopez@dcom.upv.es)

Speech separation is still a challenging issue in acoustic signal processing. To deal with this problem, sparse methods are usually employed when there are more sources than sensors (underdetermined problem). In this paper, a two-microphone separation method is presented. The proposed algorithm is based on grouping time-frequency points with similar direction-

of-arrival (DOA) using a multi-level thresholding approach. The thresholds are calculated via the maximization of the interclass variance between DOA estimates and allow to identify angular sections wherein the speakers are located with a strong likelihood. Separation is finally performed by means of time-frequency masks. Several experiments conducted under different mixing scenarios are discussed, showing the benefits of the proposed approach.

2pSC24. Classification of gender based on cepstral coefficients and spectral moments. Laura Spinu and Jason Lilley (Univ. of Delaware, Newark, DE 19716)

Several gender classification methods based on acoustic information were compared. The data came from 31 native speakers of Romanian (10 males, 21 females). A subset of fricatives and vowels (7348 tokens) was divided by hidden Markov model training into three acoustically uniform regions. For each region, (a) a set of cepstral coefficients, specifically c_0 – c_4 and (b) two sets of spectral moments, specifically bark-transformed and linear moments 1–4 plus rms, were extracted. The acoustic data were then used in a linear discriminant analysis to classify the tokens by gender. The findings show that cepstral coefficients perform better than spectral moments in gender classification, and that spectral moments are more successful than linear moments. The overall correct classification was 90% for the cepstral analysis, 78% for bark spectral moments, and 72% for linear moments. When the data from fricatives and vowels were examined separately, it was found that in all cases the vowel information yielded more accurate gender classification (e.g., 84% correct classification for cepstral moments based on vowels only), but the fricative information alone could account for 78% correct classification. These analyses provide insight into the differential distribution of acoustic features related to gender over segment types.

2pSC25. Manipulating the short time Fourier transform and cross-spectrum to estimate speech parameters. Douglas J. Nelson (Natl. Security Agency, 9800 Savage Rd., Fort Meade, MD 20755)

Presented are methods for automated estimation of speech parameters, including pitch, formant frequencies, and the instants of excitation of voiced speech. These methods are based on reassignment applied to the short time Fourier transform (STFT) and to the cross-spectrum computed as the product of the STFT and the complex conjugate of the STFT delayed in time. The cross-spectrum is a complex-valued surface whose magnitude is the conventional spectrogram and whose phase is the phase of the STFT differentiated with respect to time. It is demonstrated that this representation may be easily manipulated to detect and extract information, such as pitch and formant frequencies. Based on the cross-spectral representation, we present a peak representation that may be used to efficiently encode speech spectral features, and we demonstrate applications of this representation.

Session 2pUW

Underwater Acoustics, Acoustical Oceanography, and Signal Processing in Acoustics: Acoustic Particle Velocity and Vector Fields: Signal Processing and Communication Applications II

Mohsen Badiey, Cochair

Univ. of Delaware, College of Earth, Ocean, and Environment, Newark, DE 19716-3501

Ali Abdi, Cochair

New Jersey Inst. of Technology, 323 King Blvd., Newark, NJ 07102

Chair's Introduction—1:00

Invited Papers

1:05

2pUW1. Modeling the acoustic channel for mixed pressure and particle velocity sensors. Paul Hursky and Michael B. Porter (HLS Res. Inc., 3366 North Torrey Pines Court, Ste. 310, La Jolla, CA 92037)

Looking at our repertoire of acoustic modeling techniques, ray tracing and Gaussian beams provide a good match to modeling the underwater channel for acoustic communications, given their fundamentally broadband nature. We will discuss how we have augmented existing acoustic models, including Gaussian beam and normal mode codes, to produce particle velocity predictions. In both of these cases, the modifications consisted of manipulating the components used to represent the pressure field, as opposed to relying upon a finite difference approximation of the predicted field itself. We will discuss further modifications to these codes that enable the dynamic features of the channel to be modeled, including source and receiver motion, and motion of the ocean surface due to ocean surface waves. It is important to model realizations of the channel, rather than averaged representations, because many of the high-speed, coherent modulation schemes that are the focus of continuing research use tracking loops in their design. These tracking loops must be stressed with realistically time-evolving channels. Acoustic channel predictions will be compared with experiment data from acoustic communications experiments off the coast of Kauai in Hawaii.

1:25

2pUW2. Geo-acoustic inversion with vector sensors in shallow water. Fenghua Li and Renhe Zhang (Natl. Lab. of Acoust., Inst. of Acoust., Chinese Acad. of Sci., Beijing 100190, China, lfh@mail.ioa.ac.cn)

Vector sensors have attracted much attention in recent years. However, relative few papers have been published on the geo-acoustic inversion with vector sensors. The data recorded with vector sensors in several shallow water acoustic experiments have been studied in this paper. The experimental data and theoretical analyses indicate that (1) the intensity of the vertical particle velocity decreases faster than that of the horizontal particle velocity; (2) the same normal modes of the horizontal particle velocity and the vertical particle velocity have similar transmission losses, but different amplitudes and phases; (3) the difference of the mode amplitudes between the horizontal particle velocity and the vertical particle velocity is dependent on the eigen-value and receiver depth. With those properties, two geo-acoustic inversion schemes employing vector sensors have been developed. The first inversion scheme uses a combination of matched field processing and the difference of transmission losses between pressure and particle velocity. The second estimates the bottom sound speed and attenuation from the amplitudes of normal modes of particle velocities. Experimental results show that the developed geo-acoustic inversion methods with vector sensors can decrease the uncertainty of inversion in comparison with that by hydrophones.

Contributed Papers

1:45

2pUW3. Modulated helicity for acoustic communications and helicity-selective acoustic receivers. Timothy M. Marston and Philip L. Marston (Dept. Phys. and Astronomy, Washington State Univ., Pullman, WA 99164 2814, marston@wsu.edu)

Helicoidal or vortex beams have an azimuthal phase gradient and an axial null in amplitude. The sign and magnitude of the azimuthal phase gradient determine the helicity and topological charge of the beam and the associated handedness. Beams with a unit-magnitude charge have been generated with a four-sector array, adjacent quadrants being driven with a 90 deg phase offset [B. T. Hefner and P. L. Marston, *J. Acoust. Soc. Am.* **106**,

3313–3316 (1999)]. The present work shows that the helicity can be rapidly modulated with an appropriately timed reversal of the excitation to a single pair of diagonal sectors. This reversal was demonstrated with an electronic commutator that inverts (at a zero crossing of the sine-wave excitation) two of the channels prior to amplification. While the wave helicity may be inferred with processed signals from a detection array, it may also be detected with simple superposition of outputs from a four-element detector using appropriately offset elements. Digital communications was demonstrated where signals of the wrong helicity were suppressed. Scattering by symmetric objects on the axis of the beam preserves the helicity of the radiation [P. L. Marston, *J. Acoust. Soc. Am.* **124**, 2905–2910 (2008)]. [Work partially supported by ONR.]

2:00

2pUW4. Amplitude, phase, location, and orientation calibration of an acoustic vector sensor array, Part I: Theory. Hans-Elias de Bree (Microflown Technologies, P.O. Box 300, 6900 AH Zevenaar, debree@microflown.com)

An acoustic vector sensor array consists of multiple sound pressure microphones and particle velocity sensors. A pressure microphone usually has an omni-directional response, yet a particle velocity sensor is directional. Currently, acoustic vector sensor arrays are under investigation for far field source localization and visualization. One of the major practical issues in these applications, however, is to determine the accurate position, orientation, and complex (phase and amplitude) sensitivity of each sensor within the array. In this study, a calibration method is developed to determine each of those crucial parameters based on a limited number of measurements with a reference sensor and multiple sound sources located at known locations. The calibration method is also designed to be robust to mistakenly switched cable connections. Ideally, the calibration process should take place in an anechoic environment, but efforts have been made to compensate the effects of moderate background noise and reflections.

2:15

2pUW5. Amplitude, phase, location, and orientation calibration of an acoustic vector sensor array, Part II. Experiments. Tom Basten (Microflown Technologies B.V., Ruitenberglaan 26, 6800AH, The Netherlands, basten@microflown.com)

An acoustic vector sensor array consists of multiple sound pressure microphones and particle velocity sensors. A pressure microphone usually has an omni-directional response, yet a particle velocity sensor is directional and usually has a response pattern as a figure of number eight. Currently, acoustic vector sensor arrays are under investigation for far field source localization and visualization. One of the major practical issues in these applications, however, is to determine the accurate position, orientation, and complex (phase and amplitude) sensitivity of each sensor within the array. In this study, a calibration method, which is discussed in Part I, is verified with experiments. The method determines all the crucial parameters based on a limited number of measurements with a reference sensor and multiple sound sources located at known locations. The calibration method shows also to be robust to mistakenly switched cable connections. The experiments are performed in different acoustic environments including an anechoic room. The results will be presented in this paper.

2:30

2pUW6. Experimental study of vector array time reversal communication. Zhongkang Wang, Hongtao Zhang, and Xiaofan Yang (Hangzhou Appl. Acoust. Res. Inst., 96 Huaxing Rd., Hangzhou City, P.O. Box 1249, China)

The cooperative array performance experiment (CAPEX) was performed in Lake Washington near Seattle in September 2009 [Rouseff *et al.*, this session]. The experiment included a passive time-reversal communications component. A variety of communications signals in the 1.5–4 kHz band were transmitted including BPSK, QPSK, DSSS, and OFDM. The signals were recorded on an eight-element vertical vector array used as the receiving array at ranges up to 4 km. Results are given by using combined scalar and vector sensors. [Experiment supported by ONR.]

2:45

2pUW7. Proof of principle for geo-acoustic inversion of vector sensor array data. Robert A. Koch (Appl. Res. Labs., Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, koch@arlab.utexas.edu)

Acoustic data collected from an array with elements comprised of omni-directional hydrophone sensors co-located with acceleration vector sensor triplets are analyzed for seabed geo-acoustic information. The data were collected during the passage of a surface vessel, the R/V Montague, in an August 2006 experiment conducted in Monterey Bay. Geo-acoustic inversion with the vector sensor acceleration components parallel to the array line of bearing produced a solution approximately identical to the solution obtained from the inversion of the hydrophone array data. [Sponsored by ONR.]

3:00—3:30 Break

3:30

2pUW8. Concerning the null contours of vector sensors. Dean J. Schmidlin (2629 US 70 E, Apt. E2, Valdese, NC 28690-9005)

A common use of a vector sensor is to maximize the sensing of a plane wave coming from one direction while creating a null in regard to a plane wave coming from another direction. It is shown that when this is accomplished, there is a null not only in the desired direction but a whole contour of nulls. The shape of the contour is a function of a single parameter and is described by means of a contour plot in the azimuthal angle–elevation angle plane. Additional insight is found by mapping the contours onto a unit-sphere in the spatial domain. One byproduct of the notion of null contours is a method for selecting the coefficients of the vector sensor for the case of two interfering sources. Finally, it is shown that once the null contour is understood for the vector sensor, it is a simple matter to extend the concept to directional acoustic sensors of higher order.

3:45

2pUW9. Utilization of the pressure gradient along a sparse vertical line array to determine vertical arrival angle from partial reflections from a layered seabed. David R. Dall'Osto and Peter H. Dahl (Dept. of Mech. Eng and Appl. Phys. Lab., Univ. of Washington, Seattle, WA 98105)

Pressure signals measured by two four-element vertical line arrays, centered at depths 25 and 50 m, during an experiment off the New Jersey coast in Summer 2006 (SW06, depth 80 m) are combined to determine the pressure gradient of the received signal. The pressure gradient is converted to an estimate of vertical particle velocity, which combined with an estimate of the mean-squared pressure, and the plane wave impedance establishes a non-dimensional quantity. This quantity is inverted to determine arrival angle estimates from separate multipath arrivals of the transmitted signals (1 kHz). A geoaoustic model for the seabed [Choi *et al.*, J. Acoust. Soc. Am. 124 EL128–EL134 (2008)] includes a prominent sub-bottom reflector at depth 20 m below water sediment interface, which produces an energetic arrival time separated from the bottom arrival. The arrival angle estimates are examined as the source moves out in range from 50 to 300 m, and results are compared to arrival angle estimates derived from PE simulations. The performance of this gradient-based estimator is evaluated for various types of noise sources and levels, utilizing both real and PE-simulated time series. [Work supported by ONR.]

4:00

2pUW10. A characterization of scattered acoustic intensity fields in the resonance region. Robert J. Barton, III (Sensors & Sonar, NUWC Code 1522, 1176 Howell St., Newport, RI 02841, robert.barton@navy.mil) and Kevin B. Smith (Naval Postgrad. School, Monterey CA, 93943, kbsmith@nps.edu)

In this study, we investigate the properties of the scattered acoustic vector fields generated by simple rigid and fluid-filled spheroids. Analytical solutions are derived from acoustic target strength scattering models in the near field region. Of particular interest is the understanding of the characteristics of energy flow of the scattered acoustic vector field in the near- to far-field transition region. We utilize the separable active and reactive acoustic intensity fields to investigate the structural features of the scattered field components. Numerical and in-air measured results are presented for the near and transition region scattered acoustic vector field of simple spheroids. A qualitative method is developed for characterizing submerged scatterer properties based on boundary condition and features of the scattered vector field.

4:15

2pUW11. Three dimensional sound source localization and sound mapping using a p-u sensor array. Jelmer Wind (Microflown Technologies B.V., 6800AH Zevenaar, The Netherlands, wind@microflown.com)

An acoustic sensor array, which consists of pressure and particle velocity sensors, is an attractive alternative to phased pressure array because knowledge of the three dimensional (3-D) particle velocity directly characterizes

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the direction of the source. Hence, it is possible to localize sources in the low-frequency range, where the phase difference between pressure sensors is small, and in the high-frequency range, where aliasing occurs with pressure sensors. This article applies advanced source localization techniques from aeroacoustics to acoustic vector sensors. Several methods are simulated and tested with several configurations and validated using measurement data from an anechoic chamber. The setup to test the algorithms consists of 24 sources that are distributed in 3-D around the set ups in the anechoic chamber. The sources are uncorrelated driven with white noise.

Conventional beamforming methods have been compared with acoustic vector based deconvolution methods and MUSIC algorithms. Four configurations are tested: (1) a planar array of sound pressure microphones, (2) a planar array of pu probes (pu probes are collocated sound pressure and particle velocity probes); the velocity probes where the velocity probes are oriented in randomly, (3) a velocity gradient array, (4) a quad array of three dimensional velocity probes plus collocated sound pressure. The different methods and algorithms

TUESDAY EVENING, 20 APRIL 2010

8:00 P.M.

OPEN MEETINGS OF ASA TECHNICAL COMMITTEES

The Technical Committees of the Acoustical Society of America will hold open meetings on Tuesday, Wednesday, and Thursday evenings. On Tuesday and Thursday the meetings will be held immediately after the Social Hours. On Wednesday, one technical committee will meet at 7:30 p.m.

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

Committees meeting on Tuesday are as follows:

| | |
|---|-------------------------|
| Acoustical Oceanography | Laurel C/D |
| Engineering Acoustics | Laurel A/B |
| Musical Acoustics | Galena |
| Noise | Grand Ballroom III/IV |
| Physical Acoustics | Grand Ballroom VII/VIII |
| Psychological and Physiological Acoustics | Grand Ballroom I/II |
| Structural Acoustics and Vibration | Dover A/B |