Session 2aAO

Acoustical Oceanography and Underwater Acoustics: Environmental Effects on Acoustic Propagation

Roger M. Oba, Cochair
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Ying-Tsong Lin, Cochair
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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 Badiey (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 Badiey (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 initative 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 321OA.]

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 321OA.]

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, $I_{tot}(t) = \langle |E(t)|^2 \rangle$, particularly the incoherent or scattered intensity, $I_{incoh}(t) - I_{total}(t) - I_{coh}(t)$. 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
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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. Geoffry 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 piezolectricity. 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 at 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 loose 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=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 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. Comaparison 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, bissingerg@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 Dünnwald [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 Interdisplinary Musicology (2009)] to study the relationship between human perception and these acoustical modifications, we ascertained results that partially conflict with Dünnwald'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 psychophysical 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 destiny, 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 Calrleen 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 pedictions 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 convectional 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 (Temsa R&D & Technol. Corp., Tubitak MAM Technol. Free Zone, 41470 Gebze-Kocaeli, Turkey, okan.tandogan@temsa-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 exciters' 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. Niranjan G. Humbad, Dan Silaghi, and Matthew Morris (Behr America, Inc., Behr America, Inc. 2700 Daley Dr., Troy, MI 48083, niranjan.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 ventury 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 Systémes Intrieurs 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.

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 a 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 $B_n\psi_n(y,z)e^{\pm is_n}x$, $n=0,1,2,\ldots$, 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.

10:00—10:25 Break

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.

Contributed Paper

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, daniele.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 μPa), 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 ICBEN, the World Health Organization and UIA-PHG, and other features such as blogs and wikis.

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
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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 Breazele 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 approach 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 $\beta\propto$limA1→0(A1/A22k2a), 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 Psychoacousticss 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-numbers 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 colocated 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 hemifield. 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, Ontorio 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 callsign 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 callsigns 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-μs/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-μs interphase-gap, separated by 8 μs—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). EURASIP 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

2a TUE. AM

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. Prelingually 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 & Enviromental 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 [T 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.]

2a TUE. AM

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. Gaumond, 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., Sonetech 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-ouput communication testbed. Aijun Song, Mohsen Badiey, 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 underway 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 321OA.]

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 (HELSs) 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, Daejon 305-701, Korea, godspd@kaist.ac.kr) and Yang-Hann Kim (KAIST, Sci. Town, Daejon 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 manuver 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 321OA.]

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 Badiey (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.]