Session 2aAAa

Architectural Acoustics, Noise, Physical Acoustics, and Engineering Acoustics: Microperforated Acoustical Absorbing Materials

Christian Nocke, Cochair
Akustikbuero Oldenburg, Katharinenstr. 10, 26121 Oldenburg, Germany

Ning Xiang, Cochair
Rensselaer Polytechic Inst., Architecture, 110 8th St., Troy NY 12180

Chair's Introduction—8:00

Invited Papers

8:05

2aAAa1. The perforated panel resonator with flexible tube bundle. Yadong Lu (Inst. of Acoust., Chinese Acad. of Sci., 17 Zhongguancun St., P.O. Box 2712, Beijing 100080, PROC, yadong@mail.ioa.ac.cn), Huide Tang (Tianjin Hearing-aid Factory, Tianjin 300161, PROC), Jing Tian (Chinese Acad. of Sci., Beijing 100080, PROC), and Hongqi Li (Beijing Univ. of Technol., Beijing 100022, PROC)

This paper deals with a new type of resonant absorber, i.e., the perforated panel resonator with flexible tube bundle. The acoustical properties of the new sound-absorbing structure are measured and analyzed. From the measurement results and theoretical analysis, the following conclusions can be made. (1) The incorporation of the tube bundle into the resonator absorber is helpful to improve its low-frequency sound absorption. Because the artificial prolongation of air column length inside a hole neck will increase acoustic mass, resonance frequency can be shifted to a lower frequency correspondingly. (2) Compared with traditional perforated panel absorber, the acoustic resistance of the sound-absorbing structure can be increased considerably due to the incorporation of the flexible tube bundle into the resonator absorber, so that higher sound absorption coefficients can be expected. (3) By reasonably utilizing the tube and cavity coupling resonances, the frequency bandwidth of sound absorption of the perforated panel resonator with flexible tube bundle can be expected to broaden. The perforated panel resonator with flexible tube bundle may be used in some cases with strict space limitations and with expectations of increasing the sound absorption at lower frequencies.

8:30

2aAAa2. Measurements and calculations on microperforated sound absorbers. Christian Nocke and Catja Hilge (Akustikbuero Oldenburg, Katharinenstrasse 10, 26121 Oldenburg, Germany)

The sound absorption of microperforated absorbers can be predicted with a high degree of precision. Microperforated absorbers have been made of various materials in many applications. In recent years different manufacturers have offered a variety of microperforated materials. This paper reviews the first microperforated sound absorbers made of foils introduced some 10 years ago. This paper compares measured sound absorption coefficients of the microperforated foil absorber with the theoretical simulations. Theoretical approaches towards layered sound absorbers with microperforated components will be suggested. Finally, future developments in the field of microperforation in room acoustics will be discussed.

8:55

2aAAa3. Practical applications of transparent microperforated and microslit absorbers. Peter D'Antonio (RPG Diffusor Systems, Inc., 651-C Commerce Dr., Upper Marlboro, MD 20774)

It has been over 100 years since absorption was introduced as a quantifiable tool in architectural acoustics. Currently, the architectural acoustic community faces ever-changing challenges that require innovative approaches to absorb sound. Due to the importance of day lighting in sustainable design certification, such as LEED, and the concern about particulate material from fiberglass and mineral wool, there has developed a need for transparent and translucent absorptive materials that do not require the use of fibrous porous absorption. One approach has been the development of microperforated and microslit foils, sheets, and panels with submillimeter openings that provide significant viscous losses when spaced from a boundary. This presentation will review the design theory and absorption mechanisms, describe the various commercial materials now available, and illustrate current projects and potential applications.

2aAAa4. Effect of honeycomb structure in the back cavity on the absorption characteristics of microperforated panel absorbers. Motoki Yairi, Atsuo Minemura (Kajima Tech. Res. Inst., 2-19-1 Tobitakyu Chofu, Tokyo 182-0036 Japan), Kimihiro Sakagami, and Masayuki Morimoto (Kobe Univ., 657-8501 Kobe, Japan)

Since the principle of microperforated panel absorbers (MPPs) was established, many studies have been made on their practical applications. MPPs are typically made of very thin metal or plastic panels to obtain high absorption. However, such a thin, limp panel is in many cases not suitable for an interior finish of room walls because of its insufficient strength. The authors have proposed to use a honeycomb structure attached behind an MPP to stiffen it without deteriorating its acoustical performance. The honeycomb behind an MPP can also be expected to enhance its absorption performance, considering previous studies on porous sound-absorbing layers with the back cavity completely partitioned by a honeycomb structure. In this paper, the effect of honeycomb structure in the back cavity on the absorption coefficient of MPP was experimentally studied. The results suggest that the reverberation sound absorption characteristics become close to those of an MPP without a honeycomb structure for normal incidence. This means that the characteristics are improved with the honeycomb structure to show a higher and broader absorption peak. MPPs can be used more efficiently in more various situations with applying elaborated configurations of backing structures. [Work supported by the Kajima Foundationfs Research Grant.]

Contributed Paper

2aAAa5. An ideal absorbing materials in architectural acoustics: Micro-perforated panel absorbers. Ke Liu, Jing Tian, Fenglei Jiao, and Xiaodong Li (Inst. of Acoust., Chinese Acad. of Sci., P.O. Box 2712, Beijing 100080, P.R. China)

Micro-perforated panel absorbers (MPA) can be used to reduce or better the noise in architecture. This absorber was developed in the 1960s, when a robust sound absorber was needed for severe environments, without porous materials. It has a simple structure and the absorption properties can be exactly calculated. The panel can be made of any materials from cardboard to plastic to plywood to sheet metal, for different pur-

poses. Theoretical and experimental investigations on MPA are reviewed in this paper. By reviewing recent research work, this paper reveals a relationship between the maximum absorption coefficient and the limit of the absorption frequency bandwidth. It has been demonstrated that the absorption frequency bandwidth can be extended up to 3 or 4 octaves as the diameters of the micro-holes decrease. This has become possible with the development of the technologies for manufacturing MPA, such as laser drilling, powder metallurgy, welded meshing, and electro-etching to form micrometer-order holes. In this paper, absorption characteristics of such absorbers in random fields are presented and discussed both theoretically and experimentally. This review shows that the MPA has potentials to be an ideal absorbing materials in architectural acoustics in the 21st century.

TUESDAY MORNING, 6 JUNE 2006　　　　　　　　　　　　　　　ROOM 555AB, 10:25 TO 11:50 A.M.

Session 2aAAb

Architectural Acoustics, Musical Acoustics, and Engineering Acoustics: Surround Sound Essentials I

Alexander U. Case, Cochair
Fermata Audio + Acoustics, P.O. Box 1161, Portsmouth, NH 03802

K. Anthony Hoover, Cochair
Cavanaugh Tocci Associates, 327F Boston Post Rd., Sudbury, MA 01776

Chair's Introduction—10:25

Invited Papers

2aAAb1. Loudspeakers and listening rooms, Part I: The physical sound field. Floyd E. Toole (Harman International Industries, Inc., 8500 Balboa Blvd., Northridge, CA 91329, ftoole@harman.com)

In contrast to the considerable effort invested in concert halls and auditoria for live performances, the design of domestic listening spaces and recording control rooms has evolved with little guidance from science. In this presentation, an examination of the physical aspects of sound in small rooms reveals that traditional acoustical measures, such as reverberation time, are of little value. The reasons have to do with the differentiating facts that loudspeakers are relatively directional, and the rooms are relatively small, relatively absorptive, and furnished with large sound-scattering objects. As a result, at frequencies above about 300 Hz, listeners are in a transitional sound field consisting of direct and early reflected sounds, and a much-diminished late-reflected sound field. Loudspeaker directional properties and the acoustical behavior of the surfaces involved with early reflections are the factors relevant to perceptions. At low frequencies, room modes dominate. Notions of optimum room-dimensional ratios are of value only when loudspeaker and listener locations are known. New solutions, employing multiple subwoofers in configurations that result in selective modal excitation, can yield excellent low-frequency sound for several listeners in both rectangular and nonrectangular rooms.

2aAAb2. Loudspeakers and listening rooms, Part II: The perception of reflections. Floyd E. Toole (Harman International Industries, Inc., 8500 Balboa Blvd., Northridge, CA 91329, ftoole@harman.com)

A review of the effects of reflections on perceptions of sound quality, direction, image size, spaciousness, speech intelligibility, etc., lending perspective to the matter of which elements might be perceptually important in multichannel sound reproduction systems. In general they appear to be less problematic than is commonly believed. This, combined with the considerable ability of listeners to adapt to certain auditory influences of small rooms, brings into question several popular practices in the design of loudspeakers and listening rooms. Anechoic measurements on loudspeakers that have been done in a manner that anticipates sound reproduction in rooms can be used to calculate a figure of merit. The correlation between the calculated ratings, and those resulting from double-blind listening tests, is high: 0.86. This has implications for the acoustical treatment of rooms, and to the necessity for controlling the sound field at low frequencies. Only then can there be any assurance that the art in recordings will be heard by consumers as it was created in control rooms.

11:10

2aAAb3. Surround monitoring challenges in acoustically small spaces. Russ Berger (Russ Berger Design Group, 4006 Belt Line Rd., Ste. 160, Addison, TX 75001)

Critical listening spaces used to monitor and produce surround sound product are usually required to provide an accurate listening environment for more than one listener. Decisions are made not only about source material content and veracity, but also about how best to process and present the multichannel surround event in a manner that translates the aural experience to a wider audience. Oftentimes this audience is listening in other, less-than-ideal acoustical environments. Practical criteria and solutions are presented here to help overcome the limitations of monitoring surround sound in acoustically small spaces, as well as a discussion regarding first reflection control through the application of space coupling diffusors.

11:30

2aAAb4. Design considerations for an idealized domestic surround-sound listening space. David Moulton (Moulton Labs., 39 Ames Rd., Groton, MA 01450, dmoulton@moultonlabs.com)

I have been actively involved in the production of music for surround-sound systems since the late 1960s, including composition, production, recording (classical, multichannel pop, and experimental), loudspeaker design, and acoustical design. Based on my experiences, I am going to present some design principles (and constraints) for domestic (and public) surround-sound playback spaces, possible loudspeaker configurations, and topologies. I will also suggest a possible ideal domestic music playback space.

TUESDAY MORNING, 6 JUNE 2006   ROOM 552A, 8:25 A.M. TO 12:00 NOON

Session 2aAO

Acoustical Oceanography, Underwater Acoustics, and Engineering Acoustics: Ultra-Low and Low Frequency Marine Seismo-Acoustics

Ralph A. Stephen, Cochair
Woods Hole Oceanographic Inst., Woods Hole, MA 02543-1542

Peter M. Mikhalevsky, Cochair
Science Applications International Corp., 4001 N. Fairfax Dr., Arlington, VA 22203

Invited Papers

Chair's Introduction—8:25

8:30

2aAO1. Hydroacoustic observations of the 2004 megathrust earthquake and other events. Catherine de Groot-Hedlin (Scripps Inst. of Oceanogr., 9500 Gilman Dr., La Jolla, CA 92093-0225)

Hydroacoustic signals (T-waves) generated by the 2004 Great Sumatra earthquake were recorded by a network of 5 small hydroacoustic arrays located in the Indian Ocean at distances of 2800 to 7000 km from the epicenter. The array configurations allow for accurate determination of the receiver-to-source azimuth given coherent arrivals. Analysis of a series of 10-s windows within the T-wave coda shows that the receiver-to-source azimuth varies smoothly as a function of time, suggesting a nonstationary T-wave source. Apparent motion in the T-wave source has also been observed for many small-scale earthquakes and, for those events, reflects T-wave excitation over a broad expanse of the seafloor. For small-scale earthquakes, it is shown that the variations in the source

azimuth are predictable using a seafloor scattering model. The azimuthal variations for the great Sumatra event are shown to be inconsistent with a small-scale source and are thus indicative of the rupture velocity. The data indicate that the rupture proceeded in two distinct phases; initially it progressed northwest at approximately 2.4 km/s along the Sunda trench. At 600 km from the epicenter, the rupture slowed to approximately 1.5 km/s.

8:50

2aAO2. Use of low-frequency marine seismo-acoustics in understanding earthquake processes: Applications to the Great Sumatra-Andaman earthquake. Maya Tolstoy and DelWayne R. Bohnenstiehl (Lamont-Doherty Earth Observatory of Columbia Univ., 61 Rte. 9W, Palisades, NY 10964-8000, tolstoy@ldeo.columbia.edu)

Low-frequency (1–100 Hz) hydroacoustic recordings can provide a wealth of information on earthquake characteristics in the shallow marine environment. This frequency band, considered low frequency for acousticians, is actually high frequency for regional or global seismologists who commonly work with frequencies of a few Hz and below. In some cases this higher frequency data may provide higher resolution of some earthquake characteristics than traditional seismic methods. The 2004 Great Sumatra-Andaman earthquake provides an example where detailed analysis of the T-wave arrival has allowed detailed characterization of changes in rupture velocity that have not proven easy to resolve in traditional seismic data. Our single-station interpretation of the T-wave data shows that the fault ruptured northward from the epicenter in at least two phases with lower rupture velocity in the north. The methods behind this single station approach will be discussed along with possible future uses of this type of data for improving our understanding of earthquake rupture processes, as well as providing a rapid assessment of the size, location, and geographic extent of large submarine earthquakes, and hence their tsunamigenic potential.

9:10

2aAO3. Fault dynamics of Indian Ocean earthquakes derived from the array analysis of T-waves. Jay Pulli and Zachary Upton (BBN Technologies, 1300 North 17th St., Ste. 400, Arlington, VA 22209)

The Great Sumatran earthquake of 26 December 2004 generated some of the largest and longest T-waves ever recorded, and certainly the largest T-waves recorded on modern high-dynamic range hydrophones. For example, the recordings at the Diego Garcia hydroacoustic array indicate a T-wave duration of approximately 15 min. Array processing of these signals by many investigators indicates that the T-wave generation started near the earthquake epicenter and propagated northwest along the trench for nearly 600 km. This corresponds to the faulting zone defined by the aftershock distribution. A dataset of T-wave recordings of Indian Ocean earthquakes has been examined, covering the period 2001 to the present, including events on the oceans ridges and trenches, as well as some interplate events. The recordings were made at Diego Garcia, Cape Leeuwin, and Crozet Island. Event sizes range from magnitude 4 to 8. Bearing versus time is derived via intersensor cross correlations with a time resolution of 0.010 s, corresponding to a bearing resolution of 0.4 deg. Examples of events along the Sumatran Trench are shown with extended T-wave source areas, from which estimates of fault length and rupture velocity are made. These studies further demonstrate the utility of T-phases for understanding seismic source dynamics.

9:30

2aAO4. Using T-phase spectra to estimate rupture depth and static offset of the seafloor. David Salzberg (SAIC/OSD, 1710 SAIC Dr, M/S 1-11-15, McLean, VA 22102, david.h.salzberg@saic.com)

Analysis of T-phases recorded from recent tsunamogenic earthquakes indicates that they contain significantly more high-frequency (>50 Hz) relative to low-frequency (>10 Hz) T-phase energy with longer durations (>30 s) than nontsunamogenic events. Modeling T-phase spectrum explains this observation, where the spectrum of the recorded T-phase is the product of the source spectrum, propagation loss in the solid earth, conversion from seismic to acoustic T-phase energy, propagation loss in the ocean, and the instrument response. The propagation loss in the ocean can be accurately modeled. Assuming the seismic-to-acoustic conversion is constant over frequency, the remaining unknown term is loss in the solid earth which is primarily due to anelastic attenuation. A refined depth estimate can be obtained by modeling the solid-earth attenuation. That, coupled with the earthquake source mechanism, allows for estimation of the static ocean floor deformation and thus tsunami height. This approach has been applied to two tsunamogenic earthquakes (26, Dec. 2004, 28, Mar. 2005), and one other event. Both tsunamogenic events had significant (>1 m) seafloor uplift. In addition, the peak static offset estimated for the 26 Dec. event is consistent with estimates both from other studies, and from observations of the seafloor deformation.

9:50

2aAO5. Sumatra tsunamis recorded on hydophones and seismic stations in the Indian Ocean. Jeffrey A. Hanson and J. Roger Bowman (Science Applications International Corp., 10260 Campus Point Dr., San Diego, CA 92121)

Tsunamis generated by the great Indonesian earthquakes of 26 December 2004 (Mw 9.3) and 28 March 2005 (Mw 8.6) produced high-frequency dispersed signals (5 to 60 MHz) recorded by hydrophone and seismic stations in and around the Indian Ocean. For the first and greater earthquake, the dispersed energy is frequently observed to 30 MHz and in one case to 60 MHz. The high-frequency signals are consistent with pointlike sources. The earliest-arriving tsunami signal is located to a source at 4.3N, 93.8E using event-to-station distances estimated by measuring dispersion and azimuth estimates using an array of hydrophones. Fine structure in the tsunami signal indicates a second high-frequency source just south of Great Nicobar Island. The dispersion of later-arriving energy matches predictions for reflections of the tsunami from bathymetric features in the Indian Ocean basin. For the 28 March 2005 tsunami, the high-frequency dispersion is observed at the Diego Garcia hydrophone station and the AIS seismic station, and tsunami signals without apparent dispersion are seen at four other seismic stations. Phase velocities are estimated at hydroacoustic arrays and agree with linear dispersion theory at frequencies above 12 MHz.

10:20

2aAO6. Surface and body waves from hurricane Katrina observed in California. Peter Gerstoft (Marine Physical Lab., Scripps Inst. of Oceanogr., 9500 Gilman, 92093 La Jolla, CA), Michael C Fehler (Los Alamos Natl. Labs., Los Alamos, NM), and Karim Sabra (Scripps Inst. of Oceanogr., La Jolla, CA)

Hurricane Katrina struck land on 29 August 2005 as one of the strongest storms in the United States. The recent availability of continuous seismic data from a large number of stations enables us to characterize features of hurricane-generated noise in detail. By beamforming noise recorded on a distributed seismic array in southern California, we observe and track both the surface and body P-waves generated by Katrina in the 4–20-s period (0.05–0.25 Hz) microseism band. As the hurricane made landfall, the longer-period surface waves weakened, indicating that air/ocean/land coupling was a major factor in their generation. We observed P-waves that have propagated deep (1100 km) inside the Earth. These P-waves can be back-propagated to the hurricane. These findings demonstrate that ocean microseisms can propagate quite far and open the possibility of further use of seismic noise, even at very low signal level.

10:40

2aAO7. Source areas of midocean microseismic noise. Peter D. Bromirski (Integrative Oceanogr. Div., Scripps Inst. of Oceanogr., 9500 Gillman Dr., La Jolla, CA 92093-0209, pbromirski@ucsd.edu)

Comparison of microseismic noise data collected at the ocean-bottom Hawaii-2 Observatory (H2O), located midway between Hawaii and California, with wind and wave data shows three frequency bands having different source locations. At frequencies from about 0.3 to 0.5 Hz, the noise spectrum shows a strong correlation with overhead wind speed and direction, implying that this energy is generated locally. Correlation of swell height above H2O with microseism energy at frequencies below 0.3 Hz is generally poor, implying that these signals originate at distant locations. The highest noise levels observed occur in the 0.2- to 0.3-Hz band, apparently resulting from the interaction of waves forced by pre- and postfrontal winds from relatively large coincident storms in the same vicinity. Correlation of the H2O microseism levels with NOAA buoy data, with hindcast wave height data over the North Pacific, and with seismic data from mainland and island stations, defines likely source areas of the 0.1- to 0.2-Hz signals. Most of the microseism energy at H2O between 0.08 and 0.2 Hz appears to be generated by high-amplitude storm waves impacting long stretches of coastline nearly simultaneously. [Work supported by the California Department of Boating and Waterways.]

Contributed Papers

11:00

2aAO8. Sound of hurricane. Konstantin Naugolnykh (ESRL/Zeltech, 325 Broadway, Boulder, CO)

The strong perturbation of atmosphere such as hurricane is accompanied by infrasound radiation. This radiation can be observed at a distance of thousands of miles from the hurricane. The effect of sound radiation is connected apparently to the interaction of the counterpropagating sea-surface waves that produces a sound radiation of the doubled frequency of the surface wave oscillation. [M. S. Longuet-Higgins, A theory of the origin of microseisms]. Theoretical model of the sound generation by the sea-surface waves is developed on the basis of the advanced theoretical approach [Wright *et al.*, "Hurricane directional spectrum spatial variation in the open ocean," J. Phys. Ocean. **31**, 2472–2488 (2000)]. The spectrum of sound radiation is presented as a function of the sea-surface wave features. Some estimation of the parameters of infrasound produced by hurricane is made using the data of Wright *et al.* on the hurricane-produced sea-surface waves spectrum. The diffraction effects are taken into account to evaluate the trapping of the radiation by a horizontal atmospheric waveguide, important for the long-range propagation of the noise. [K. A. Naugolnykh, and S. A. Rybak, "Sound generation due to the interaction of surface waves," Acoustical Physics **49**, 88–90 (2003)].

11:15

2aAO9. Anomalous transparency of the water/air interface for low-frequency sound. Oleg A. Godin (CIRES, Univ. of Colorado and NOAA/Earth System Res. Lab., 325 Broadway, Boulder, CO 80305, oleg.godin@noaa.gov)

Sound transmission through water/air interface is normally weak because of a stark density contrast between the two media. For a monopole sound source, at depth larger than a wavelength, the ratio R of the acoustic power transmitted into the air to the total emitted acoustic power is a small quantity of the order of the ratio of impedances of air and water. Acoustic transparency of the interface is shown to rise dramatically at low frequencies, where depth of the monopole source is small compared to the wavelength, with R increasing by up to 36 dB. Under certain conditions, almost all of the acoustic energy emitted by a low-frequency monopole source underwater is predicted to be radiated into the air rather than into the water. Physically, higher transparency at lower frequencies is due to increasing role of inhomogeneous waves emitted by the source and a destructive interference of fields in water due to the actual and image sources. The anomalous transparency of the water/air interface may have significant implications for generation of low-frequency acoustic noise in the air by bubbles collapsing underwater and for acoustic monitoring and detection of powerful underwater explosions for the purposes of the Comprehensive Nuclear Test Ban Treaty.

11:30

2aAO10. Low-frequency sound reflection and conversion to Darcy-type type diffusion waves at bottom interfaces with marine sediments. Jason D. Holmes, Allan D. Pierce, William M. Carey (Boston Univ., Boston, MA 02215), and James F. Lynch (Woods Hole Oceanograph. Inst., Woods Hole, MA 02543)

The upper medium is sea water, and the bottom medium is a marine sediment whose physical description is consistent with the theory advanced by Frenkel in 1944 and with the model proposed by Biot in 1961 for a porous medium at low frequencies. Within the sediment, three generic types (modes) of wavelike disturbances are possible. These three modes are here termed the acoustic, shear, and Darcy modes. The first two are governed by propagating wave equations and the third by a diffusion equation. The first and third are dilatational. All three are nominally uncoupled, although coupling occurs at interfaces. When an obliquely incident acoustic wave impinges from the water on the interface, all three types of disturbances are in principle excited in the sediment, and the details of the excitation are governed by interface conditions, whose derivation is reported in the present paper. Each mode has its own characteristic parameters, and the sediment reflection coefficient, considered as a function of angle of incidence and of frequency, implicitly depends on all of these parameters. The question is addressed as to how, given such reflection coefficient data, one might extract the parameters for the Darcy mode for representative realistic experimental circumstances.

11:45

2aAO11. The relation between received microbarom signals and the sea state under a hurricane. Roger Waxler, Kenneth E. Gilbert, and Carrick L. Talmadge (NCPA, University, MS 38677)

It is known that the peak at about 0.2 Hz observed in both seismic (the microseism peak) and infrasound (the microbarom peak) spectra have their origin in the standing wave component of the ocean surface wave spectrum. The largest source of microseism and microbarom radiation is known to be from the sea states built up under hurricanes. In 1963 Hasselmann published a relation between the ocean wave spectrum and the microseism signal produced by the ocean waves. We have recently derived an analogous expression for the microbarom signal. This expression is presented and possible implications for remote monitoring of hurricane intensity are discussed.

TUESDAY MORNING, 6 JUNE 2006   BALLROOM A, 7:45 A.M. TO 12:10 P.M.

Session 2aBB

Biomedical Ultrasound/Bioresponse to Vibration: Memorial Session for Frederic Lizzi

Jeffrey A. Ketterling, Cochair

Riverside Research Inst., Lizzi Ctr. for Biomedical Engineering Research, 156 William St., New York, NY 10038-2609

Ernest J. Feleppa, Cochair

Riverside Research Inst., Lizzi Ctr. for Biomedical Engineering Research, 156 William St., New York, NY 10038-2609

Chair's Introduction—7:45

Invited Papers

7:50

2aBB1. Spectral parameter imaging of the corneal stroma. Ronald H. Silverman, Omer Gal, Monica Patel, and D. Jackson Coleman (Weill Medical College of Cornell Univ., 1300 York Ave., Rm. K808, New York, NY 10021)

Frederic L. Lizzi was a pioneer in the fields of high-frequency ultrasound and tissue characterization. This report details use of spectral parameter imaging of very-high-frequency radio frequency data for characterization of the cornea. The corneal stroma consists of broad 1–2-micron-thick belts (lamellae) embedded in a proteoglycan matrix. The lamellae consist of collagen fibrils 30 nm in diameter with center-to-center spacing of about 60 nm. Any disturbance to this orderly arrangement will result in loss of optical transparency. Parameter images representing the slope, intercept, and midband fit to the calibrated power spectrum were generated by post-processing of radio frequency data of 35- and 70-MHz scans of the cornea. Stromal backscatter was significantly higher at 70 MHz than at 35 MHz. In an uncomplicated LASIK-treated cornea examined 6 years postsurgery, significant differences in backscatter between the stroma anterior versus posterior to the LASIK interface were observed. This may relate to alteration in lamellar organization, even in the absence of loss of optical transparency. Frequencies beyond 50 MHz offer a probe of tissue microstructure that could be employed in evaluation of stromal organization. [Work supported by NIH Grant EB000238 and Research to Prevent Blindness.]

8:10

2aBB2. Inverse scattering bioacoustics: Under the influence of Fred Lizzi. Michael Insana (Univ. of Illinois, MC 278, 1304 Springfield Ave., Urbana, IL 61801)

A long-standing problem in bioacoustics is the assessment of soft tissue health from an analysis of backscattered echo signals—an inverse problem known as tissue characterization. Pathologists view magnified images of tissue morphology to assess tissue function and grade disease states. Ultrasonic tissue characterization methods are attempts to accomplish the same assessment noninvasively. The goal is not to image the microstructure directly, but rather to diagnose tissues from measurements of statistical moments of the echo spectrum properly normalized to isolate tissue features. Fred Lizzi pioneered this approach and provided an analytical framework used widely. It will be shown how his analysis evolved from classic studies in atmospheric physics to give a clear picture of what is becoming a material science approach to diagnosis using ultrasonics. Dr. Lizzi's methods have been used by many to diagnosis and monitor treatment of cancer from a very small feature space. Others have expanded these methods to discover the sources of ultrasonic scattering in tissue systems and observe dynamic features of physiology. Coupled with elasticity imaging, inverse scattering approaches provide *in vivo* tools for studying the influence of molecular signaling on tissue structure during the progression of cancer and atherosclerotic diseases.

8:30

2aBB3. Stimulation of collagen production with low-intensity acoustics. James Greenleaf, Heather Argedine, and Mark Bolander (Mayo Clinic College of Medicine, 200 First St. SW, Rochester, MN 55901)

Low-intensity pulsed ultrasound is a commonly prescribed therapy for delayed unions and nonunions after fractures. Several prospective double-blind clinical trials have shown that a 1.5-MHz ultrasound signal at a 200-s tone burst repeating at 1 kHz shortens the time to clinically evaluated healing by 30%. We have shown *in vitro* that pulsed ultrasound increases aggrecan gene expression and aggrecan production in chondrocytes, a key cell in the fracture healing process. Because the square wave pulsed 1.5-MHz signal produces radiation force vibration at 1 kHz, we tested the hypothesis that dynamic radiation force, not ultrasound, is responsible for the biological effect of the signal. Experiments showed that a 1-kHz acoustic square wave, simulating the radiation force caused by the clinically used signal, induced chondrogenesis similarly to pulsed ultrasound treatment in ATDC5 cells. These results may have implications for remote ultrasound or acoustic stimulation of different types of strain-sensitive cells in addition to those responsible for fracture healing.

8:50

2aBB4. Noninvasive prostate surgery using high-intensity focused ultrasound. Narendra T. Sanghvi (Focus Surgery, Inc., 3940 Pendleton Way, Indianapolis, IN 46226)

The salient features of ultrasound are to provide both medical diagnostic and therapeutic applications due to its short wavelength in soft tissue. These combinations have motivated us to apply ultrasound for the noninvasive surgery of the prostate diseases for both benign and malignant tumors. Over a decade of engineering and clinical research a successful transrectal ultrasound device, the Sonablate, has been developed and clinically practiced for the treatment of localized prostate cancer as well as benign prostatic hyperplasia. The main advantages of using the Sonablate device are on-line imaging of the prostate in 2D and 3D, accurate therapy planning, and real-time monitoring of treatment. The long-term, over 6 years follow-up clinical results from two multicenter clinical studies indicate no severe long-lasting side effects and acceptable efficacy of the treatment. The primary benefits of this treatment are fewer complications, outpatient procedure, and very quick recovery. The presentation will include design of the device, characterization of therapy and imaging transducer, and treatment parameters for the effective clinical results.

9:10

2aBB5. Quantitative ultrasound techniques. Timothy Hall, James Zagzebski, and Ernest Madsen (Univ. of Wisconsin, 1530 MSC, 1300 University Ave., Madison, WI 53706)

The ultrasound research group at UW-Madison has been pursuing quantitative measurement of the acoustic properties of tissues for more than three decades. Much of that effort parallels the efforts pioneered by Fred Lizzi and colleagues at Riverside Research Institute (RRI). While the RRI group was blazing trails in data acquisition, analysis, and interpretation techniques, the UW group was investigating the limits of measurement accuracy using similar approaches. Wise selection of the tissues to investigate, great intuition regarding fruitful paths, and good hard work resulted in substantial progress and success in the RRI quantitative ultrasound efforts. Careful measurements with high-quality phantoms, in parallel with tissue measurements, resulted in substantial progress in determining limits on quantitative ultrasound measurement accuracy under various experimental conditions. This presentation will highlight some of the differences and similarities in the approaches to quantitative ultrasound by these two groups and how we can use that information in future investigation.

9:30

2aBB6. Therapeutic applications of ultrasound in intracellular drug delivery and tissue ablation. Cheri X. Deng (Dept. of Biomed. Eng., Case Western Reserve Univ., 10900 Euclid Ave., Cleveland, OH 44106-7207)

Biomedical application of ultrasound has been advancing beyond diagnostic imaging. Innovative strategies of therapeutic applications of ultrasound have been exploited that utilize both thermal and mechanical effects generated by ultrasound. For ultrasound mediated intracellular drug and gene delivery, ultrasound has been used to increase cell permeability transiently. We have developed novel techniques to investigate the mechanisms of ultrasound-induced cell porosity or sonoporation in real time at the single cell level. We studied the dynamics of sonoporation and correlation with delivery outcome in mammalian cells, bacteria, and tissues. Applications of high-intensity focused ultrasound (HIFU) include tissue thermal ablation and other applications. We have developed interdisciplinary techniques using real-time fluorescent imaging, acoustic radiation elastography, and optical coherent tomography (OCT) techniques to study the changes of electrophysiology, mechanical, and optical properties of cardiac tissues during focal HIFU cardiac ablation. [Work supported by NIH, ACS, and NSF.]

9:50

2aBB7. Enhanced drug delivery with ultrasound and engineered delivery vehicles. Katherine W. Ferrara, Mark A. Borden, Paul A. Dayton, Azi Kheirolomoom, Dustin E. Kruse, Michaelann Tartis, and Aaron F. Lum (Dept. of Biomed. Eng., UC Davis 451 East Health Sci. Dr., Davis, CA 95616)

We demonstrate that local drug delivery can be achieved by ultrasound, combined with engineered delivery vehicles, where the vehicles have a diameter on the order of nanometers to microns. Delivery vehicles can be created from microbubbles with a thickened shell or a lipid shell decorated with drugs, genes, or nanoparticles. Alternatively, liquid-filled nanoparticles can be employed to carry the desired compound. Ultrasound can deflect these vehicles from the center of the flowstream, can fragment the vehicle releasing its

contents, and may enhance the uptake of the particle or its contents by cells in the desired region. The ultrasonic mechanisms behind these changes are summarized. The addition of targeting ligands to the shell to improve target specificity is also explored. Methods to measure the effectiveness of local drug delivery, including correlative imaging modalities, binding assays, and cytotoxicity assays, will be described. [The support of NIH CA 103828 is gratefully acknowledged.]

10:10–10:25   Break

10:25

2aBB8. Tissue characterization using novel high-frequency models. Michael Oelze (Dept. of Elec. & Comput. Eng., Univ. of Illinois at Urbana-Champaign, 405 N. Mathews, Urbana, IL 61801, oelze@uiuc.edu)

Conventional models for ultrasound tissue scattering at conventional ultrasound frequencies (less than 15 MHz) were insufficient for differentiating between two kinds of solid tumors in mice that had clear morphological differences. F-tests on backscattered power spectra from the two kinds of tumors revealed significant differences between the two curves for frequencies above 16 MHz. New models were constructed to better reflect the ultrasonic interaction with underlying tissue structure. One model considered only the cell nucleus as a dominant source of scattering. A second model, the whole cell model, considered the nucleus and cytoskeletal structure. These models were used at high ultrasonic frequencies (16 to 70 MHz) to characterize the two kinds of tumors and compared with scatterer estimates using a conventional model, the spherical Gaussian model. The nucleus model revealed significant differences between the two kinds of tumors but scatterer estimates did not match the underlying structure as observed from optical photomicrographs. The whole cell model did not reveal significant differences but scatterer estimates matched underlying structure. [Work supported by NIH Grant F32 CA96419 and start-up funds from the Department of Electrical and Computer Engineering at the University of Illinois at Urbana-Champaign.]

10:45

2aBB9. Scattering of high-frequency ultrasound by cells and cell ensembles: In search of the dominant scattering source. Michael Kolios (Dept. of Phys., Ryerson Univ., Toronto, Canada)

High-frequency ultrasound (HFU) imaging (in the range of 20–60 MHz) has been recently enabled by advances in transducer technology and electronics. These high frequencies afford greater image resolution, on the order of 40 microns, at the expense of penetration depth, limited to a few centimeters. The associated ultrasound wavelengths of HFU are 25–75 microns, on the order of the size of cells. Therefore, it is hypothesized that at these frequencies ultrasound tissue characterization will provide more meaningful information about the physical characteristics of cells and any changes that occur in their structure during cancer treatment. The late Dr. Frederic Lizzi pioneered ultrasound tissue characterization and the seminal papers he published provided the impetus for this work. Here, backscatter from individual cells and tightly packed cell aggregates of the same cells (emulating tumors) are examined and compared to theoretical models of ultrasound backscatter. Recent work with a series of cells lines with different sizes and physical characteristics will be presented. It will be shown that for *in vitro* and *in vivo* experiments, the backscatter signal characteristics best correlate with nuclear size.

11:05

2aBB10. Automated coronary plaque characterization with intravascular ultrasound backscatter: *In vivo* and *ex vivo* validation. D. Geoffrey Vince, Anuja Nair, M. Pauliina Margolis, Stacy L. Kirby, Jennifer J. E. Kuznicki (Volcano Corp., Advanced Technology Lab., 9500 Euclid Ave., Cleveland, OH 44195 gvince@volcanocorp.com), Xiaofeng Wang, Barry D. Kuban (The Cleveland Clinic Foundation, Cleveland, OH 44195), and James R. Margolis (Miami International Cardiology Consultants, Miami, FL 33137)

Atherosclerosis is considered both a systemic and local disease. Current diagnostic tools do not allow adequate *in vivo* characterization of lesions. Intravascular ultrasound (IVUS) backscatter analysis has displayed the potential for real-time plaque characterization by advanced spectral analysis. One hundred-fifteen plaques from 51 coronary arteries were imaged *ex vivo* at physiologic pressure, using 20-MHz IVUS transducers. After imaging, the arteries were pressure fixed and matching histology was collected. Regions were selected from histology and corresponding IVUS data were used to build the ''VH'' plaque classification system using spectral analysis and classification trees. VH images were validated *ex vivo* by comparison with histology via 407 selected regions that comprised 162 fibrous, 84 fibro-fatty, 69 necrotic-core, and 92 dense-calcium regions. The overall predictive accuracies were 92.9%, 93.4%, 94.3%, and 99.3% respectively, with sensitivities and specificities ranging from 84% to 99%. Reproducibility was assessed *ex-vivo* and *in vivo* with two pullbacks of IVUS catheters in each. A semiparametric regression model and the *t*-test were used to quantify reproducibility, which was high in each case. VH is a robust technique that provides accurate and reproducible information on plaque composition, with considerable potential for assessment of plaque vulnerability in real time.

Contributed Papers

11:25

2aBB11. Discovery of the ring resonant frequency for noninvasively estimating local elastic modulus of arterial wall. Xiaoming Zhang and James F. Greenleaf (Mayo Clinic College of Medicine, 200 First St. SW, Rochester, MN 55905, Zhang.xiaoming@mayo.edu)

Increased stiffness of arteries has recently gained acceptance as an independent risk factor for cardiovascular and many other diseases. Pulse wave velocity (PWV) is widely used for estimating the stiffness of an artery. However, PWV is an average indicator of artery stiffness between

the two measuring points, and therefore does not identify local stiffness. In addition, the thickness of the artery is needed to calculate artery elastic modulus. We have discovered that there is a series of ring resonant frequency in arteries that can be used for estimating local arterial elastic modulus. These resonances can be excited by remote palpation with selectively placed ultrasonic radiation force. Experiments on ring resonance were carried out on an excised artery as well as a rubber tube. This family of resonances occurs at relatively low frequency, around 356 Hz for the artery. Estimation of the Young's modulus is 135 kPa by the ring frequency 356 Hz without the requirement of the arterial thickness. The

estimated modulus is very consistent with the other ring resonant frequency measurements. The theory of ring resonance is developed and verified with the rubber tube experiments.

11:40

2aBB12. Imaging with high-frequency annular arrays. Jeffrey A. Ketterling, Sarayu Ramachandran, Ronald H. Silverman (Frederic L. Lizzi Ctr. for Biomed. Eng., Riverside Res. Inst., 156 William St., New York, NY 10038), Orlando Aristizábal, and Daniel H. Turnbull (New York Univ. School of Medicine, New York, NY 10016)

Frederic Lizzi helped pioneer the use of high-frequency ultrasound (HFU) for ophthalmic imaging and disease diagnosis. A limitation of HFU single-element transducers is their limited depth-of-field (≈ 1 mm). One approach to improve the DOF for HFU applications is to use annular arrays. A five-element, 40-MHz annular array was developed for ophthalmic and small-animal imaging. The array consisted of a 9-μm PVDF film bonded to a patterned, copper-clad polyimide film. An experimental system was assembled to permit the digitization of radio-frequency (rf) data from the array. The rf data were collected from all transmit/receive ring pairs and the data were later postprocessed. The performance of the array was first validated with a wire phantom and then *in vitro* data were acquired from excised bovine eyes and a cadaver eye. *In vivo* data were acquired from an anesthetized rabbit and from mouse embryos. Image comparisons were made between rf data that had been summed with no delay corrections and rf data that had been processed with a synthetic-focusing algorithm. Synthetic-focus images showed a clear improvement in DOF in terms of image brightness and enhanced membrane and cavity-boundary definition. [This research was supported in part by grants from the NIH (EY014371 and NS038461).]

11:55

2aBB13. Ultrasonic and magnetic-resonance spectra in tissue-type imaging for planning and monitoring prostate cancer treatment. Shreedevi Dasgupta, Ernest Feleppa, Jeffrey Ketterling, Sarayu Ramachandran (Riverside Res. Inst., 156 William St., New York, NY 10038), Christopher Porter (Virginia Mason Medical Ctr., Seattle, WA), and Fernando Arias-Mendoza (Columbia Univ., New York, NY)

No reliable method of imaging prostate cancer currently exists. These prostate tissue-typing studies aim to improve the effectiveness of biopsy guidance and treatment monitoring by developing better imaging methods for identifying cancerous prostate tissue. Success will reduce the false-negative rate of biopsies and treatment side-effects. Ultrasonic (US) radio-frequency (rf) echo-signal data acquired during biopsy examinations were used to compute spectral parameters. These spectral parameters along with clinical parameters, e.g., PSA, were used to train a neural network classifier using biopsy results as the gold standard. Cancer-likelihood scores from a look-up table were used to generate tissue-type images (TTIs). The ROC-curve area for US neural-network-based classification was 0.844 ± 0.018 vs 0.638 ± 0.031 for B-mode-based classification, and the sensitivity of neural-network based classification was superior to that of B-mode-guided biopsies. These classification methods are being extended to include magnetic-resonance spectral (MRS) techniques that use the choline to citrate ratio-to-distinguish cancerous from noncancerous prostate tissue. 3D renderings of prostatectomy histology, US, and MR images show encouraging correlations, and combining MRS parameters with US spectral parameters appears to have potential to further improve prostate-cancer imaging. [Work supported in part by NIH Grant CA053561.]

Session 2aED

Education in Acoustics: Hands-On Experiments for High School Students

Uwe J. Hansen, Chair

Indiana State Univ., Dept. of Physics, Terre Haute, IN 47809

Students and senior scientists will staff experiment stations for local high school students who will perform the experiments. These students will thus be exposed to a variety of acoustics principles and get some practical hands-on experience with a number of research tools. Regular ASA meeting participants are welcome to the session as long as their participation does not interfere with student hands-on activities.

Session 2aMU

Musical Acoustics: Scaling of Musical Intstrument Families

George A. Bissinger, Chair

East Carolina Univ., Dept. of Physics, Greenville, NC 27858

Chair's Introduction—8:25

Invited Papers

8:30

2aMU1. Violin octet scaling revisited. George Bissinger (Phys. Dept., East Carolina Univ., Greenville, NC 27858, bissingerg@mail.ecu.edu)

Schelleng employed only two major resonances to scale the violin to other pitch regions, creating the violin octet: the main air (now labeled A0) to be placed at $1.5x$ and the main wood (B1, actually two modes) to be placed at $2.25x$, the lowest string frequency. B1 employed flat-plate scaling and an empirical relationship between plate mode frequencies and B1. Experimental modal analysis of a complete octet showed flat-plate scaling for the corpus generally worked well, but A0 scaling was unreliable, even with postassembly modifications. Failure of the rigid-cavity Rayleigh relationship for A0 originated from an obvious major omission, cavity wall compliance, and one unobvious source, A0 coupling with the next higher cavity mode A1, which emerged much later via Shaw's network model. Incorporating wall compliance into Shaw's model greatly improved A0 and A1 predictions to within 10% over the octet. Acoustical analysis combined with patch near-field acoustical holography showed that A1, via induced corpus motion, was the most important radiator for the largest instrument in the main wood region and that the B1 modes, via net volume changes, can radiate significantly through the f-holes, two radiation mechanisms never considered in Schelleng's original scaling.

9:00

2aMU2. Violin octet scaling: The practical side. Robert J. Spear (The New Violin Family Assoc., 42 Taylor Dr., Wolfeboro, NH 03894)

Schelleng' scaling procedure attempts to place A0 at $\sim 1.5x$ and the first corpus bending modes at $\sim 2.25x$, the lowest string pitch. An experiment to move A0 into this position (also influencing A1), using the Stradivari model G as reference, helps produce the characteristic sound of the violin in cases where B1 placement was not optimum. The experiments suggest that the model G worked well for the signature modes of a classic violin, successfully scaling at least four instruments (mezzo, alto, tenor, and a small baritone), but that instruments at the extremes—treble and soprano at one end of the scale and both basses at the other—will likely require a refinement of the model G and placement of modes different from those of the reference violin and from each other. Scaling for acoustics and scaling for size are not the entire answer since ergonomic demands are often at odds with both. Geometrical scaling is essential for aesthetics and the uniform shape of the instruments, but it does not hold up well for rib heights, rib thicknesses, and cavity air volumes. This experiment suggests that significant latitude of air and wood mode placement exists within which good results can occur.

9:30

2aMU3. Practical issues in designing strings for different-sized bowed instruments. Fan-Chia Tao (J. D'Addario & Co., P.O. Box 290, Farmingdale, NY 11735, fan.tao@daddario.com)

The playing length and frequency of musical strings are fixed by the design of the instrument. The playing tensions of commercially available strings for standard bowed instruments follow an approximately linear scaling law in relation to string length. However, the string designer can adjust the playing tension to yield different results, and playing tensions have changed throughout history as string technology and playing techniques have evolved. Another string property that can be varied by the string designer is damping. Empirical data suggest that over a wide variety of bowed instruments and frequency ranges, strings with similar damping give similar playing characteristics.

10:00−10:15   Break

10:15

2aMU4. Scaling of reed instruments: The case of the saxophone family. Jean-Pierre Dalmont, Bruno Gazengel (Laboratoire d'Acoustique de l'Universit'e du Maine, 72085, Le Mans cedex 9, France, jean-pierre.dalmont@univlemans.fr), and Jean Kergomard (CNRS, Marseille Cedex 20, France)

The saxophone family is one of the most extended musical instruments family from the contrabass to the sopranino. The timbre of these instruments is unmistakable: there is a small probability to mix up a saxophone with an oboe or a bassoon. Now a question is: What is so specific in the timbre of the saxophones? Notes of various saxophones are recorded, analyzed, and compared with those of other instruments such as the oboe. More specifically, the transient duration and the formant frequency are determined. Then, the question which physical parameters influence the transient duration and the formants is studied. The influence of geometrical parameters such as the cone angle and the length of the cone truncation will be discussed.

10:45

2aMU5. Scaling of flutelike instruments : An analysis from the point of view of the hydrodynamic instability of the jet. Benoit Fabre, Nicolas Montgermont, and Johan Gabriels (Lab. for Musical Acoust., LAM, Paris 6 Univ., Paris, France)

The sound produced by musical instruments of the flute family is the result of the efforts of the instrument maker as well as those of the player. For instance, different instruments of a recorder consort show, on one hand, dimensions optimized by the maker for their individual compass as well as for a global identity and coherence of the whole consort. On the other hand, the players have to adapt their blowing to each instrument of the consort. Therefore, the study of the different instruments has to include the geometry given by the maker as well as the required blowing conditions. The study reported here presents data gathered on different flute families: the geometries and the instrument passive resonances are studied together with the behavior of the instruments under artificial blowing and, finally, under the control of a musician. The analysis of the data in terms of the jet hydrodynamic instability offers an efficient framework to describe the choices of the maker and the adaptation of the player.

Contributed Papers

11:15

2aMU6. A new family of stringed musical instruments. Sophie Léger and Samuel Gaudet (Dept. of Mathematics, Univ. de Moncton, Moncton, NB, E1A 3E9 Canada)

A new family of stringed musical instruments will be introduced. The Tritare[TM] is the first member of this new family and has evolved from the mathematical analysis of the vibrations of networks of strings [Gaudet *et al.*, J. Sound & Vib. **281**, 219–234 (2005)]. It is a guitarlike instrument that uses six networks of three string sections (Y-shaped networks) instead of the classical single strings. The sounds that it produces can be more or less harmonic depending on the string attack conditions and the physical configurations of the string networks. A method of customizing the frequency spectrum to contain desired harmonic components will be demonstrated. A numerical model, which uses finite differences, used to simulate the 3D nonlinear vibrations of more complex string networks involving several string sections and junction points will also be described. The harmonic analysis of the frequency spectra resulting from the numerical simulations and experimental measurements will be presented, as well as network configurations which produce very unique and interesting tone colors. Some of these configurations form the basis for a new family of stringed musical instruments. The presentation will include a demonstration of the Tritare[TM].

11:30

2aMU7. Scaling of brasswind instruments. Robert W. Pyle, Jr. (11 Holworthy Pl., Cambridge, MA 02138, rpyle@post.harvard.edu) and Arnold Myers (Univ. of Edinburgh, Edinburgh EH8 9AG, UK)

Consider two musically similar brasswind instruments pitched an octave apart, like trumpet and trombone. The steady-state tones of the trumpet, shifted downwards in pitch an octave by playing a recording at half speed, sound very similar to those of the trombone. The bore shape of the trumpet, however, is not a half-size dimensional replica of that of the trombone. The trumpet bell is about half the size of the trombone bell, but the narrower parts of the trumpet are considerably more than half the diameter of the corresponding parts of the trombone. Acoustic impedance measurements show the Q of air-column resonances to be similar in both trumpet and trombone. This is consistent with the dimensions of the instruments, combined with the dependence of viscous and thermal damping on tubing diameter and frequency. (Viscous and thermal damping dominate radiation damping throughout the normal playing range.) These observations suggest that a plausible scaling law for a family of brasswinds would preserve the pitch-shifted spectrum and the Q of the resonances across members of the family. How well (or poorly) brass makers follow this in practice will be shown from measurements of dimensions and acoustic input impedance on a variety of instruments.

Session 2aNSa

Noise and Architectural Acoustics: Audio-Visual Design in Soundscapes I

Brigitte Schulte-Fortkamp, Cochair

Technical Univ. Berlin, Inst. of Technical Acoustics, Einsteinufer 25, 10587 Berlin, Germany

Bennett M. Brooks, Cochair

Brooks Acoustics Corporation, 27 Hartford Turnpike, Vernon, CT 06066

Chair's Introduction—9:30

Invited Papers

9:35

2aNSa1. Soundscape measurement and analysis.　Klaus Genuit　(HEAD Acoustics GmbH, Herzogenrath, Germany) and Wade Bray　(HEAD Acoustics, Inc., Brighton, MI 48116)

　　Soundscapes typically include a variety of sources, with individual and collective attributes (level, spectral structure, time structure, and perceived characteristics such as loudness, sharpness, fluctuation, and roughness). Relevant sound sources impinge on listeners from multiple directions and distances. Examples will be given of large differences between conventional and psychoacoustic values as a function of distance, and of multi-source summations. As an example, the subjective loudness difference for sound incidence over a small versus large solid angle for a constant sound pressure level and spectrum is known. Individual-source-relevant information, important in context, may be unrecognized unless spatial hearing is included in soundscape assessment. A demonstration of binaural versus monaural soundscape data acquisition, and subsequent analysis by both conventional and psychoacoustic measures considering human perception including time dependency, will be given. Evidence will be presented for considering human hearing not only in the analysis, but also in the acquisition.

9:55

2aNSa2. Soundscape measurement in an urban area.　Brigitte Schulte-Fortkamp　(TU-Berlin, Einsteinufer 25, 10587 Berlin, Germany)

　　This presentation is about a demonstration of soundscape evaluation of an urban area with respect to measurements and interviews. In a recent field study in Berlin, indoor and outdoor measurements were carried out in a residential area in which inhabitants were complaining about noise and vibration due to heavy traffic over day and until late at night. In this context, the question of the appropriate positioning and calibration of the measuring devices in indoor and outdoor measurements was very important. The measurement procedure as well as procedures which determine how to conduct interviews will be described and discussed by taking into account how to make progress in measuring soundscapes. Moreover, the focus will be on necessary contributions to a measurement which usually deals with dB(A) and lacks psychoacoustical measuring factors.

10:15

2aNSa3. Traditional measurement methods for characterizing soundscapes.　Bennett M. Brooks　(Brooks Acoustics Corp., 27 Hartford Turnpike, Vernon, CT 06066, bbrooks@brooksacoustics.com)

　　A soundscape may be defined as the acoustical environment at an outdoor location. The physical sound occurring at the location may be from natural or man-made sources, or both. Perception of the soundscape can provide comfort and needed information to the listener, or may be annoying. The existing soundscape may be subject to design modifications for a variety of purposes. The goal could be to protect or enhance some sounds, such as the laughter of children and chirps of wildlife in a park, or the speech in an amphitheater. Conversely, the goal may be to reduce or eliminate unwanted sounds, such as transportation or industrial noise. Engineering and aesthetic soundscape design may proceed based on the characterization of the acoustical environment. The soundscape can be characterized by a number of measured acoustical parameters. Traditionally, these parameters have included A-weighted sound level, spectral content, sound-level statistics and averages, alone or in combination, and sound-level time history. Listener perceptions and nonacoustical parameters, such as time of day or year, community relevance, and listener attitudes may also be applied to soundscape characterization and design. Examples of several different soundscapes, their characterizations, and proposed design modifications will be discussed.

10:35

2aNSa4. Documenting visual context for soundscapes.　Tim Lavallee　(LPES, Inc., 14053 Lawnes Creek Rd., Smithfield, VA 23430), Vincent Catania　(38 Temple St., Boston, MA 02114), and Robert Kull　(Parsons, Norfolk, VA 23502)

　　There are several considerations to make when documenting a given soundscape. Not only is the technical sound information important, but the space, receptor, and visual context are important as well. This is primarily important in furthering the ability of sound and soundscape professionals to communicate with architects, planners, and other engineers—and vice versa. A portable digital

recording system was assembled and used along a high-end digital camera to document a range of soundscapes. These tests were preformed in urban, suburban, and rural areas at different times and in different contextual settings. Both visual and audio information of a locale were documented. Although this method can be simple or complex, the combination of these two components highlights the human factors that are present for a given soundscape. This will be a discussion of the portable equipment and a presentation of sound recordings, spectra, and photographs of a range of soundscapes. The variety and complexity of soundscapes and the benefits of visual clues to add depth to a given documentation effort will be highlighted.

Contributed Papers

10:55

2aNSa5. Soundscape analysis and acoustical design strategies for an urban village development. Youngmin Kwon, Pattra Smitthakorn, and Gary W. Siebein (Architecture Technol. Res. Ctr., School of Architecture, Univ. of Florida, 134 ARCH, P.O. Box 115702, Gainesville, FL 32611, ymkwon@ufl.edu)

Overall establishment of an acoustical landscape (soundscape) would be the fundamental source of sonic information that is useful in initiating urban design and planning strategies. A soundscape analysis was conducted in conjunction with an urban transportation and community design project targeting the western section of Gainesville, FL. The community was acoustically characterized and analyzed by means of quantitative measurements and qualitative assessments by observation. The measurements included instantaneous, 1-min short-term and 24-h long-term measurements. The parameters involved in these measurements are A-weighted average Lp, Leq, Lmax, Lmin, and LDN. The overall sonic environment of the community as well as the types of sounds that are desirable and noises that are not desireble in the existing and future contexts were identified. The results showed that the community was dominated mostly by traffic noise from several major thoroughfares and by occasional fire alarm noise. Finally, methods or strategies to reduce, buffer, and mitigate the undesirable sounds and to preserve the desirable sounds were suggested. The addition of new characteristics to the soundscape, such as people activity sounds and ''natural'' sounds, were also proposed. The application of acoustical urban design strategies to the specifically proposed urban transportation design plans was further analyzed.

11:10

2aNSa6. Territoriality and sound delimitation of urban space. Cobis Rabah Derbal and Hamza Zeghlache (BP 12 SMK, Constantine 25003, Algeria, cobis.derbal@free.fr)

Whether the urban space is physically materialized or abstracted it is often delimited by perceptive and sociocultural relations. In the present methodological approach, urban space is supposed to be modulated in its delimitation by qualitative and quantitative criteria of its noise environments. This introduces clearly the concept of sound delimitation and of noise environment territoriality. One can notice that sound phenomena are very present and are marking the social practice and behavior in urban spaces and give them a very specific sonority. Just like the visual or sociocultural aspects, the sonority of the public place or the urban noise environment marks their territoriality and spatiality by creating a whole sound environment characteristic of dimensions and forms of the space. The urban noise can thus be an element of marking, connecting, breaking, or structuring spaces which are morphologically and visually irreconcilable. One can speak about sound delimitation of urban space beyond its morphological limits. That delimitation can well articulate the social practice of urban space to its dimensional aspect. The study put out the concept of acoustic space in opposition to morphological space which it is inevitably necessary to reconcile in order to better practice the urban space.

11:25

2aNSa7. Recomposition of the urban sonic environment. Cobis Rabah Derbal and Hamza Zeghlache (BP 12 SMK, Constantine 25003, Algeria, cobis.derbal@free.fr)

Acoustics-related considerations are taken to be secondary in most of the design processes of public places. This fact always leads to an unpredictable noise and urban sonic environment around such spaces. Studies have shown the importance of the noise and the quality of the urban sonic environment in the social practice and in the daily use of public places. These places strongly mark the degree of dynamism of the urban structure of our cities and are often subjected to a very complex situation of noise nuisance. The main aim of the study is to introduce a source of noise that might have an effect of recomposing positively a deficient sonic environment of such places. This approach is also to try to restore the sound in its utilitarian and practical function and to articulate the perception of the spatial and temporal aspects of the public place. It could be an element of transforming urban noise into a peaceful soundscape. A fountain turned out to be that sonic source and it can be easily introduced in the urban environment around the public places. It is a sort of urban instrumentation used in order to recompose or requalify an urban noise environment.

11:40−12:00
Panel Discussion

12:30
Session participants meet in room 554AB for experimental field measurements

1:00−3:00
Experimental field measurements conducted

Note: Panel discussion will be held in Session 2pNSb at 3:00 p.m.

Session 2aNSb

Noise, Education in Acoustics, Engineering Acoustics, and Architectural Acoustics: Essential Acoustical Curriculum for Noise Control Techniques for Engineers, Industrial Hygienists, and Architects

William J. Murphy, Cochair

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

Ralph T. Muehleisen, Cochair

Illinois Inst. of Technology, Civil and Architectural Engineering, 3201 S. Dearborn, Chicago, IL 60616

Chair's Introduction—8:50

Invited Papers

8:55

2aNSb1. Essential elements of acoustical curricula for noise control engineers. Robert J. Bernhard (Ray W. Herrick Labs., Purdue Univ., 140 S. Intramural Dr., West Lafayette, IN 47907, bernhard@purdue.edu)

Because the supply of noise control and acoustical engineers is small, many of the noise control practitioners practicing in general mechanical engineering fields are retrained on the job. These practitioners are often capable in the area in which they are mentored but lack sufficient background to move to new areas. In order to address a broad range of noise control issues, an engineer must be comfortable to work in waves, or modes, or dBese (engineering controls). The essential curriculum for engineering practitioners would include a foundation in each of these approaches as well as the analytical, computational, and experimental background upon which to build engineering solutions. This presentation will include discussion of how these objectives can be met at the undergraduate level, at the graduate level, and through continuing education.

9:25

2aNSb2. The need for an essential acoustics curriculum and its delivery via distance education. Victor W. Sparrow, Anthony A. Atchley, Karen P. Brooks, and Gerald C. Lauchle (Grad. Program in Acoust., Penn State, 214 Appl. Sci. Bldg., University Park, PA 16802, vws1@psu.edu)

Many individuals have identified a need to better educate engineers, industrial hygienists, and architects with an essential acoustical curriculum including noise control techniques. There are less than a dozen major programs training these individuals, and demand far exceeds the supply of education suppliers. One effective method of providing education to busy professionals who otherwise would not be exposed to an essential curriculum is distance education. Approximately 10 years ago Penn State developed a series of noise control engineering courses delivered through CD-ROM. Now with DSL and cable modems more common, full motion video streaming is a preferred delivery method. Students currently take graduate level courses in acoustics at Penn State via resident education, continuing education, and distance education. There is no reason why an essential acoustical curriculum could not be delivered similarly. One benefit of video streaming is an opportunity for both live and archived lecture material. Students can attend live recitation sections while retaining the ability to see archived material on demand. Technology such as high-bandwidth connections, inexpensive webcams, and microphones also allows two-way audio and video for direct student-instructor interaction.

9:55

2aNSb3. The acoustical education of an architect. Gary Siebein (School of Architecture, Univ. of Florida, 625 NW 60th St., Ste. C, Gainesville, FL 32607)

Sound can provide valuable contributions to the education and practice of architecture. First, architecture students should develop a theory of how sound and other environmental systems can become part of a larger theory of the making of architecture. Second, architects are trained to have highly developed sensibilities in the aesthetic design of space. Expanding the aesthetic component of buildings and environments to include the aural environment has the potential to enrich the fabric of communities and buildings. Third, architecture students must become acquainted with the materials, analytic techniques, and design approaches that acoustical consultants use. Furthermore, for acoustical information to become an integral part of architectural design and education, it must be organized in a way that allows expression or poesis to occur because these aesthetic principles are the building blocks of architectural design. Therefore, the science and engineering principles of noise control and architectural acoustics must be transformed into material that can become spatial, visual, and manipulable in creative ways by a design team. This requires that the basics of acoustics be reformulated in an architectural manner. This paper will address several of the transformations that have been attempted in the architectural curriculum at the University of Florida.

10:40

2aNSb4. Acoustics curriculum for architectural engineers. Ralph T. Muehleisen (Civil and Architectural Eng., Illinois Inst. of Technol., 3201 S. Dearborn, Chicago, IL, 60616, muehleisen@iit.edu)

As building design and technology increase in complexity, the need to consider acoustics in the design also increases. For decades, architects have relied on engineers for the design of structural, mechanical, electrical, and lighting systems of buildings, and are increasingly relying on engineers for the design of high-efficiency building enclosures and natural lighting systems. At the same time, building systems have become increasingly integrated, and good design of the building systems requires a very interdisciplinary education. These needs have given rise to the formal discipline known as architectural engineering (AE). With education in architecture, structures, building systems, and construction management, the architectural engineer is uniquely qualified to begin a career as an acoustical consultant. There are now over a dozen ABET-accredited AE programs throughout the country and most offer at least some education in architectural acoustics and noise control. In this talk, the requirements of an acoustics education for architectural engineers are outlined and the curriculum of several AE programs is presented.

11:10

2aNSb5. Industrial noise control engineering principles and standard practices applicable to engineers and industrial hygienists. Robert R. Anderson (James, Anderson & Associates, Inc., 2123 University Park Dr., Ste. #130, Okemos, MI 48864)

Industrial organizations have accepted the notion that noise control is no longer 100% of one person's responsibility, but a collaborative effort among engineers and industrial hygiene professionals. Proven noise control measures exist and are commercially available for many industrial noise sources. There are also noise control concepts and techniques that can be adopted by maintenance and other personnel as standard practices. Awareness of the elements of noise control engineering can offer benefit in areas such as equipment procurement, workplace risk assessment, and manufacturing program planning. The focus of this presentation is to discuss the elements of noise control engineering that manufacturing engineers and industrial hygiene professionals would need to apply out of their university education into practice, and the benefits that would be derived from this information.

11:40–12:00
Panel Discussion

TUESDAY MORNING, 6 JUNE 2006

ROOM 550AB, 9:00 TO 11:45 A.M.

Session 2aPA

Physical Acoustics: Outdoor Sound Propagation

Richard Raspet, Cochair
Univ. of Mississippi, National Ctr. for Physical Acoustics, Coliseum Dr., University, MS 38677

James P. Chambers, Cochair
Univ. of Mississippi, National Ctr. for Physical Acoustics, Coliseum Dr., University, MS 38677

Contributed Papers

9:00

2aPA1. Ground-to-ground sound propagation in cloud-topped boundary layers over land. Roger Waxler, Carrick L. Talmadge, and Kenneth E. Gilbert (NCPA, University, MS 38677)

Sound propagation under cloud-covered skies is discussed. Observations are presented which suggest that under dense cloud cover ground-to-ground sound propagation is dominated by a single mode similar to the classic surface mode for outdoor sound propagation. A theory of sound propagation under cloud covered skies has been developed. It is shown that there is a sound duct centered in the cloud layer. Of the modes in this duct a small number (typically one or two for frequencies up to a few hundred Hz) of ground interacting modes, modes whose amplitudes on the ground are significant, are identified. Over land, one of the ground-interacting modes can be identified with the surface mode.

9:15

2aPA2. A continuously cascaded conformal mapping for sound propagation over irregular terrain. Santosh Parakkal, Kenneth E. Gilbert, and Di Xiao (Natl. Ctr. for Physical Acoust., Univ. of Mississippi, Coliseum Dr., Universtiy, MS 38677)

The so-called earth flattening conformal mapping is a common approach for radio wave propagation over a spherical earth. Applied sequentially (cascaded), it can be used to represent a series of hills, each of which has a constant radius of curvature. We show that the mapping can also be applied to general terrain that has a continuously varying radius of curvature. In the mapped coordinates, the terrain is flat, but the mapped sound speed is $c/\sqrt{J}$, where c is the actual sound speed and J is the Jacobian of the transformation. The continuously cascaded mapping method is outlined, and numerical comparisons are made with a well-tested nonconformal mapping method.

2aPA3. The absorption of sound on Mars using the direct simulation Monte Carlo. Amanda D. Hanford and Lyle N. Long (Grad. Program in Acoust., The Penn State Univ., University Park, PA 16802, ald227@psu.edu)

The physical properties that govern the absorption of sound on Mars are very similar to those on Earth: classical losses associated with the transfer of acoustic energy into heat, and relaxation losses associated with the redistribution of internal energy of molecules. The difference in molecular composition between Earth and Mars as well as the lower atmospheric pressure on Mars results in larger values for the absorption coefficient on Mars. The direct simulation Monte Carlo (DSMC) method is the simulation tool used for modeling sound propagation in the Martian atmosphere. DSMC describes gas dynamics through direct physical modeling of particle motions and collisions. DSMC is based on the kinetic theory of gas dynamics, where representative particles are followed as they move and collide with other particles. The validity of DSMC for the entire range of Knudsen numbers (Kn), where Kn is defined as the mean free path divided by the wavelength, allows for the exploration of sound propagation in the Martian atmosphere for all values of Kn. Successful application of the DSMC method to acoustic waves in the Martian atmosphere, including the details of molecular relaxation in gas mixtures, will be shown. [Work supported by NASA.]

2aPA4. Correlation functions of temperature and velocity fluctuations in a turbulent atmosphere. Vladimir E. Ostashev (NOAA/Earth System Res. Lab., Boulder, CO 80305 and Dept. of Phys., New Mexico State Univ., Las Cruces, NM 88003), Sergey N. Vecherin (New Mexico State Univ., Las Cruces, NM 88003), D. Keith Wilson (U.S. Army Engineer. Res. and Development Ctr., Hanover, NH 03755), and Sandra L. Collier (U.S. Army Res. Lab., Adelphi, MD 20783-1197)

von Kármán spectra of temperature and velocity fluctuations are widely used in theories of turbulence and wave propagation in random media, including sound propagation through a turbulent atmosphere. They are probably the simplest generalization of the Kolmogorov spectrum of homogeneous and isotropic turbulence which accounts for the outer scale of turbulence. However, the correlation functions of temperature and velocity fluctuations corresponding to the von Kármán spectra are proportional to the modified Bessel functions and are rather involved. In this paper, we propose to use much simpler correlation functions of temperature and longitudinal velocity fluctuations whose dependence on the distance R between observation points is given by $\exp(-R^{2/3}/L^{2/3})$, where L is the outer length scale. It is shown that the spectra of the proposed correlation functions coincide with the Kolmogorov spectrum in the inertial subrange and are bounded in the energy subrange. For some problems, the proposed correlation functions can simplify analysis of sound propagation through a turbulent atmosphere. Examples of the use of these correlation functions in acoustic tomography of the atmosphere and calculations of the coherence function of a sound field are presented. [Work supported by ARO, Grant DAAD19-03-1-0104.]

2aPA5. Upward propagation of nonlinear sound and magnetic sound in the atmosphere. Lev Ostrovsky (Zel Tech/NOAA ESRL, R/PSD4, 325 Broadway, Boulder, CO 80305)

As is known, even relatively weak ground-level perturbations (earthquakes, tsunami, explosions) may cause very significant displacements in upper atmospheric layers where the gas density is small. It results in shock formation and subsequent rapid dissipation of wave energy. This presentation deals with the upward propagation of a short, finite-amplitude wave in atmosphere with exponentially decreasing density, including areas where the magnetic pressure prevails, and the acoustic energy is transformed to that of fast and/or slow magnetic sound. The height of shock formation is determined, and the wave attenuation due to shocks is described. Propagation from a localized source is also discussed. Some estimates are given.

2aPA6. Effect of longitudinal coherence on angle-of-arrival estimation in atmospheric turbulence. Sandra Collier (U.S. Army Res. Lab., AMSRD-ARL-CI-ES, 2800 Powder Mill Rd., Adelphi, MD 20783-1197, scollier@arl.army.mil), Vladimir Ostashev (NOAA/Earth System Res. Labs., Boulder, CO 80305), and Keith Wilson (U.S. Army Engineer Res. and Development Ctr., Hanover, NH 03755)

A statistical model that accounts for the effects of atmospheric turbulence on a received acoustic signal has been previously developed [Collier and Wilson, J. Acoust. Soc. Am. **113**, 2704–2718 (2003)]. It was used in calculations of the performance bounds, and of a maximum likelihood estimator (MLE), for the angle of arrival (AOA) of an acoustic signal received at an array operating in atmospheric turbulence with fluctuations described by a von Karman spectrum. This statistical model is based on the moments of a sound field as derived in the theory of wave propagation in a random medium (WPRM) [Ostashev, *Acoustics in Moving Inhomogeneous Media* (E&FN Spon 1997)]. In this model, an approximation to the second moment was used based on the transverse coherence from WPRM. Recently, the longitudinal coherence has been derived for WPRM [Ostashev *et. al*, 151st, Meeting of the Acoustical Society of America (2006)]. We examine limiting cases to determine the effects of the longitudinal coherence on the MLE and Cramer-Rao lower bound of the AOA.

2aPA7. The coherence function of a sound field propagating in a turbulent atmosphere. Vladimir E. Ostashev (NOAA/Earth System Res. Lab., Boulder, CO 80305 and Dept. of Phys., New Mexico State Univ., Las Cruces, NM 88003), Sandra L. Collier (U.S. Army Res. Lab., Adelphi, MD 20783), and D. Keith Wilson (U.S. Army Engineer. Res. and Development Ctr., Hanover, NH 03755)

The coherence function of a sound field propagating through an atmosphere with temperature and wind velocity fluctuations is important for many practical applications involving microphone arrays. In the literature, the transverse coherence function has been studied in detail, when acoustic sensors are located in a plane perpendicular to the sound propagation path. In this paper, for the case of line-of-sight sound propagation, a closed-form expression for the coherence function of a plane wave is derived for arbitrary location of acoustic sensors. Using this expression, for some limiting cases simple formulas for the coherence function are obtained which are valid for arbitrary spectra of turbulence. Furthermore, the coherence function is calculated and analyzed in detail for the Gaussian and von Kármán spectra of temperature and wind velocity fluctuations. In particular, the longitudinal coherence radius, when acoustic sensors are located along the sound propagation path, is calculated. It is also shown that contours of the coherence radius are elongated perpendicular to this path for relatively small propagation distances, and along it for large distances.

2aPA8. Measurements of acoustic and seismic pulses from outdoor explosions. Donald G. Albert (ERDC-CRREL, 72 Lyme Rd., Hanover, NH 03755), Keith Attenborough, and Patrice Boulanger (The Univ. of Hull, Hull HU6 7RX, UK)

Measurements of the ground vibrations produced by airborne detonations of C4 were conducted at locations with a variety of ground types, including concrete, soil, forest, tropical vegetation, and snow cover. The measurements show that, although two separate seismic (ground vibrational) arrivals can be detected in all cases, the early seismic arrival from an underground path is always much smaller than the vibration induced by the air blast arrival. The acoustic-to-seismic coupling ratio for the atmospheric wave is a constant with respect to distance and peak pressure at a given location, but varies from site to site, usually between 1 and 14 μm/s/Pa. A conservative empirical equation to predict ground vibration from explosions is derived. This equation predicts that the commonly used

vibrational damage criteria of 12 and 25 mm/s will be exceeded when the peak positive pressure exceeds 480 Pa (147.6 dB) or 1 kPa (154.0 dB), respectively. Either of these levels is much higher than the Army overpressure damage criterion of 159 Pa (138 dB). Thus in most situations damage from blast overpressure will occur long before damaging levels of ground vibration are reached. [This research supported by the U.S. Army Corps of Engineers and SERDP Seed Project SI-1410.]

11:15

2aPA9. Comparison of high-frequency measurements of blast wave propagation to computational model predictions. Alexandra Loubeau and Victor W. Sparrow (Grad. Prog. Acoust., Penn State, 202 Appl. Sci. Bldg., University Park, PA 16802, aloubeau@psu.edu)

In November 2005, the latest in a series of blast wave propagation experiments was conducted with the U.S. Army. Blast waveforms measured with a wideband microphone [W. M. Wright, ONR Report NR-384-321 (1971)] had shorter rise times than those captured with conventional $\frac{1}{8}$-in. microphones. This indicates that the $\frac{1}{8}$-in. microphone setup did not have the high-frequency bandwidth needed to capture such short rise times. The trend observed is that nonlinear effects steepen the waveform, thereby decreasing the shock rise time, up to a range of 50 m. At 100 m, the rise times had increased slightly, but they were still shorter than they were at 25 m. The experimental results are compared to predictions from a computational model. The model, based on the Pestorius/Anderson hybrid time-frequency domain algorithm, accounts for nonlinearity, geometrical spreading, thermoviscous absorption, and absorption and dispersion due to molecular relaxation. It is found that the model predicts shorter rise times than what was measured with the wideband microphone, but the

rise-time variation with range is similar. Atmospheric turbulence present in the experiments, but not accounted for in the model, is a likely cause for the difference in rise times. [Work supported by U.S. Army ERDC CERL.]

11:30

2aPA10. The use of Lorentz transformations for simplifying aeroacoustic boundary value problems. M. H. Dunn (Natl. Inst. of Aerosp., 100 Exploration Way, Hampton, VA 23666, mhd314@aol.com)

The prediction of community noise from aircraft flyovers is computationally complex when arbitrary aircraft configurations are subjected to realistic flows. Mathematically, the problem is described by a set of scattering surfaces (wings, nacelles, and fuselage) moving uniformly with known noise sources in the presence of an inviscid, irrotational, background flow. A four-dimensional (space-time) boundary value problem (BVP) for noise propagation and scattering is defined by the mass and momentum conservation equations along with appropriate hard- or soft-wall boundary conditions on the scattering surfaces and the Sommerfeld radiation condition. This system is augmented by the appropriate Kutta conditions for geometry with edges to produce a uniquely solvable BVP. The governing differential equations are complicated by factors strongly dependent upon the flow field. However, Lorentz transformations can be used to define new independent and dependent variables such that the differential equations depend on the local sound speed plus terms that involve flow gradients. Neglecting flow gradient terms produces the classical wave equation BVP, which can be solved using contemporary numerical methods. The original acoustic variables are easily recovered via the inverse Lorentz transformation. In addition, the formulation can be used to analyze physical conditions under which the uniform flow approximation is valid.

TUESDAY MORNING, 6 JUNE 2006   ROOM 551AB, 8:15 TO 9:45 A.M.

Session 2aPPa

Psychological and Physiological Acoustics: Loudness, Outer Hair Cells, and Otoacoustic Emissions

Walt Jesteadt, Chair

Boys Town National Research Hospital, 555 N. 30th St., Omaha, NE 68131

Contributed Papers

8:15

2aPPa1. A comparison of intensity discrimination and increment detection as measures of intensity resolution. Walt Jesteadt, Jessica Messersmith, Lori Leibold, Samar Khaddam, and Hongyang Tan (Boys Town Natl. Res. Hospital, 555 N. 30th St., Omaha, NE 68131)

Intensity resolution can be measured by determining the threshold for discrimination of two stimuli differing only in intensity or by determining the threshold for detection of an increment in a longer duration stimulus. Better resolution has been reported for the latter task, but the results are inconsistent across conditions and subjects. In the present study, data were obtained for the two tasks in a range of conditions for 12 adult subjects with normal hearing. Thresholds for intensity discrimination and increment detection were measured for a 4000-Hz tone presented at 20, 40, 60, or 80 dB SPL, using a two-track adaptive procedure that estimated levels required for 71% and 89% correct. The increment was 20, 40, 80, or 160 ms in duration. The pedestal was either the same duration as the increment or was 200 ms longer, centered on the increment. Thresholds expressed in units of delta L were more uniform across subjects and conditions for increment detection than for intensity discrimination. Thresholds were lower on average for increment detection than for intensity discrimination at pedestal levels above 20 dB, but only at the shorter increment durations, and thresholds were not consistently lower for all subjects. [Work supported by R01 DC006648.]

8:30

2aPPa2. Primary speech frequencies: Adaptation inside versus outside—Inhibition or excitation? Ernest M. Weiler, David E. Sandman (Dept. Of Commun. Sci., Univ. Cincinnati, Cincinnati OH 45267), and Joel S. Warm (Univ. Cincinnati, Cincinnati, OH 45267)

Previously, loudness adaptation from the ipsilateral comparison paradigm (ICP) and the simple adaptation (SA) procedure were compared at primary speech- related frequencies (250 to 4000 Hz), with only the ICP showing significant loudness decline/adaptation [Tannen *et al.*, J. Gen Psychol. **128**, 385–399 (2001)]. Eighty naive undergraduates participated in any one of three subsequent studies of ICP versus SA adaptation. In agreement with Tannen *et al.*, only the ICP showed significant loudness decline/adaptation from 250 to 6000 Hz. However, at 8000 Hz both the ICP and SA show significant adaptation, and correlate significantly. However, despite the lack of correlation between individual scores under 8000 Hz, a further analysis of *group mean* values from 250 to 8000 Hz showed a significant correlation ($r=0.97$, $p<0.001$). Perhaps the SA procedure shows a lack of adaptation at speech frequencies because the method is insensitive. Alternatively, it may be considered that a ''stable platform'' [i.e., no adaptation for *constant* stimuli] favors the perception of the constantly varying speech signals. Support for the ''stable platform'' versus the ''insensitivity'' hypothesis will be discussed.

2aPPa3. Induced loudness reduction as a function of frequency separation. Jeremy P. Marozeau, Michael Epstein, and Mary Florentine (Commun. Res. Lab., Dept. of Speech-Lang. Pathol. and Audiol., Northeastern Univ., Boston, MA 02155, marozeau@neu.edu)

When a loud tone (inducer tone) precedes a softer tone (test tone), the loud tone can reduce the loudness of the softer tone. This phenomenon is called induced loudness reduction (ILR). The amount of ILR is greatest when the inducer tone and the test tone are presented at the same frequency. As the difference in frequency between the two increases, the amount of ILR decreases [Marks L. E., J. Exp. Psychol. Hum. Percept. Perform. **20**, 382–396 (1994)]. However, the exact course of this decrease with increasing frequency separation has not been thoroughly tested. In this experiment, the amount of ILR produced by a 2.5-kHz 80-dB-SPL inducer was measured from 800 Hz to 6 kHz using a frequency-sweeping test tone. Nine normal-hearing listeners matched the loudness of the test tone to a 70-dB-SPL 500-Hz tone using a tracking procedure. Average results show a maximum amount of ILR between 2.5 and 3 kHz, with a loudness reduction equivalent to 12 dB. This amount decreases with frequency separation, but is still significant when the frequency of the test tone is between 1.5 and 4.5 kHz, corresponding to more than nine critical bands. [Work supported by NIH/NIDCD Grant R01DC02241.]

2aPPa4. A multicompartmental cochlear model with piezoelectric outer hair cells. Shan Lu, David C. Mountain, and Allyn E. Hubbard (College of Eng., Boston Univ., 8 Saint Mary's St., Boston, MA 01720, slu@bu.edu)

A hydromechanical, multicompartment model of the cochlea, which employed a phenomenological outer hair cell (OHC) cell-body electromotility, was able to mimic the physiologically measured response of the basilar membrane (BM) [A. E. Hubbard *et al.*, "Time-domain responses from a nonlinear sandwich model of the cochlea," in *Biophysics of the Cochlea: From Molecule to Model*, edited by A. W. Gummer (World Scientific, Singapore)]. An improved model that included OHC electroanatomic parameters and scalar electrical parameters was able to mimic cochlear microphonic (CM) data at low frequencies, but not BM responses at high-frequencies, because the OHC membrane capacitance severely reduces OHC transmembrane potential. To improve the high-frequency performance, we used a piezoelectrical model of the OHC. Consequently, the mechanical loading of the OHC translates bidirectionally with the electrical impedance of the OHC. The new model compares well with physiological data from Gerbil at 40 dB SPL [T. Ren and A. Nuttall, "Basilar membrane vibration in the basal turn of the sensitive gerbil cochlea," Hearing Res. **151**, 48–60 (2001)]. [This work was supported by NIH.]

2aPPa5. Application of force to the cochlear wall: Effect on auditory thresholds, outer hair cell transduction, and distortion-product otoacoustic emissions. Greg A. O'Beirne[a] and Robert B. Patuzzi (The Auditory Lab., Physiol., School of Biomed. and Chemical Sci., Univ. of Western Australia, Crawley WA 6009, Australia)

Described are the changes in cochlear sensitivity and mechanoelectrical transduction during a novel cochlear perturbation: the application of force to the cochlear wall. While numerous methods exist to create *transient* shifts in the operating point of the outer hair cell (OHC) transducer, including low-frequency acoustic bias [G. Frank and M. Kössl, Hear. Res. **113**, 57–68 (1997)] and hydrostatic bias [A. N. Salt and J. E. DeMott, Hear. Res. **123**, 137–147 (1998)], attempts to create *prolonged* operating point shifts are largely thwarted by the numerous sources of ac coupling in the auditory system which prevent transmission of dc stimuli to the hair cells. The application of force sufficient to deform the otic capsule produced a consistent drop in neural thresholds and a sustained bias of the OHC operating point that did not rapidly adapt back to normal. Near-simultaneous measurements of compound action potential thresholds, distortion-product otoacoustic emissions, the OHC transfer curves derived from low-frequency cochlear microphonic waveforms, and the endocochlear potential were performed. The data provide ample evidence of the resistance of the cochlea to dc mechanical stimuli, particularly those which do not cause a large pressure differential across the basilar membrane. [The authors gratefully acknowledge the surgical assistance of Dr. Peter Sellick.] [a] Currently at Department of Communication Disorders, University of Canterbury, Christchurch, New Zealand.

2aPPa6. Influence of the sweeping direction of the primaries f_1 and f_2 in the assessment of distortion product otoacoustic emissions (DPOAEs) fine structure. Miguel Angel Aranda de Toro, Rodrigo Ordoñez, and Dorte Hammershøi (Dept. of Acoust., Aalborg Univ., Fredrik Bajers Vej 7 B5, DK-9220, Aalborg, Denmark, maat@acoustics.aau.dk)

It was studied if the sweeping direction of the primaries f_1 and f_2 influences fine structure measurements of the $2f_1-f_2$ distortion product otoacoustic emission (DPOAE). Two different methodologies named DPOAE5$_{ASC}$ (ascending sweep) and DPOAE5$_{DES}$ (descending sweep) were implemented with the commercial system ILO96. Both methods perform fine structure measurements in the same frequency range (1.4–6 kHz) with $L1/L2=65/45$ dB, $f_2/f_1=1.22$ and 0.7 s averaging time per primary presented. DPOAE fine structures were measured in the right ear of 14 normal-hearing subjects with both methods in a balanced experiment and without refitting the sound probe. Results showed that the two methods are highly repeatable and able to detect fine structures even in subjects with low S/N. However, when DPOAE5$_{ASC}$ measurements are compared with DPOAE5$_{DES}$, fine structures appear shifted and the level contour is altered. A minor level difference was inadvertently induced due to differences in the level calibration procedure of the two methods employed. The significance of this is currently being studied. [Work supported by the Danish Research Council for Technology and Production.]

Session 2aPPb

Psychological and Physiological Acoustics, ASA Committee on Standards, and Noise: Individual Susceptibility to Noise-Induced Hearing Loss

Sharon G. Kujawa, Cochair

Massachusetts Eye and Ear Infirmary, Audiology Dept., 243 Charles St., Boston, MA 02114

Lynne Marshall, Cochair

Naval Submarine Medical Research Lab., Subase New London, Box 900, Groton, CT 06349

Invited Papers

10:15

2aPPb1. Individual susceptibility to noise-induced hearing loss: A review. Donald Henderson (Ctr. for Hearing and Deafness, State Univ. of New York at Buffalo, 137 Cary Hall, Buffalo, NY 14214)

After 40+ years of study, we can conclude that there is a great range of individual differences in susceptibility to noise-induced hearing loss (NIHL). The presentation will review and evaluate the major sources of variances: *acoustic*, where all exposures are evaluated with dBA and some energy-related metric; *exogenous*, where the effects of noise can be mitigated by external factors such as temperature, coexposure to chemicals, smoking, etc.; *endogenous*, where the effects of noise are partially related to an individual's sex, skin color, acoustic reflex, distortion product suppression, degree of fatigue, and genetic factors.

10:45

2aPPb2. Searching the whole genome to identify genes contributing to noise resistance in inbred mouse strains. Bruce L. Tempel, Valerie A. Street (Dept. of Otolaryngol.-HNS and the V.M. Bloedel Hearing Res. Ctr., Univ. of Washington, Seattle, WA 98195-7923, bltempel@u.washington.edu), and Sharon G. Kujawa (Harvard Med. School, Boston MA)

Toward understanding the biological basis of noise resistance and eventual formulation of therapies to prevent noise-induced hearing loss (NIHL), the identification of genes contributing to noise resistance is a critic step. To simplify our whole genome search, we chose to study isogenic, inbred strains of mice. When exposed to noise (octave band 8–16 kHz, 103 dB SPL, 2 h) the good-hearing CBA/CaJ strain showed ~50-dB thresho shift measured 2 weeks postexposure; in contrast 129S6 mice showed ~10-dB shift. Through directed breeding schemes we showed that the noise resistance trait in 129S6 is recessive to CBA. Screening N2 backcross animal for noise resistance suggested a multigenic pattern of inheritance with 4 to 6 genes contributing. Quantitative trait locus (QTL) mapping of 234 N2 animals reveals 4 different chromosomal regions that contribute significantly to the resistance. Congenic strains carrying only one or two of these QTL regions are currently being bred, with the expectation that these strains will show partial noise resistance. Further studies including ancestral haplotype analysis and differential gene expression array analysis will allow us to narrow the contributing genomic regions and identify specific genes contributing to noise resistance. [Work supported by NIH.]

11:05

2aPPb3. Can otoacoustic emissions indicate susceptibility to noise-induced hearing loss in individual ears? Lynne Marshall, Judi A. Lapsley Miller (Naval Submarine Medical Res. Lab., Groton, CT 06349-5900, marshall@nsmrl.navy.mil), and Laurie M. Heller (Brown Univ., RI 02912)

Otoacoustic emissions (OAEs) can potentially be used to identify normal-hearing individuals who are susceptible to imminent noise-induced hearing loss. Until now, there has been no way to monitor an individual's susceptibility dynamically as it varies due to physiological and environmental factors. Although it is known that groups of normal-hearing noise-exposed people have lower than average levels of OAEs, such data have never been used to predict future hearing loss in the same individuals. Here, we discuss two studies where the OAEs and hearing thresholds of individuals were measured before and after hazardous noise exposure. Individuals with normal hearing and low-level or absent OAEs at baseline were at increased risk for acquiring noise-induced hearing loss after the noise exposure. This supports the theory that OAEs can reflect incipient inner-ear damage undetected by standard behavioral hearing tests. Furthermore, OAE efferent strength measures may be predictors of NIHL in humans. The animal data are promising, but developing an efficient OAE test that reliably distinguishes a strong efferent reflex from a weaker one in humans is challenging. [Work supported by ONR & USAMRMC.]

2aPPb4. Noise-induced and age-related hearing loss interactions. Sharon G. Kujawa (Dept. of Otology and Laryngology, Harvard Med. School, and Dept. of Audiol., Massachusetts Eye and Ear Infirmary, 243 Charles St., Boston, MA 02114) and M. Charles Liberman (Massachusetts Eye and Ear Infirmary, Boston, MA 02114)

Noise-induced hearing loss (NIHL) and age-related hearing loss (AHL, or presbycusis) are widespread health problems that will continue to increase as our society ages. NIHL and AHL often coexist in the same ear; however, the conditions under which they interact and the mechanisms by which they do so remain poorly understood. Inspired by epidemiological studies suggesting that noise-exposed ears age *differently* from nonexposed ears [Gates *et al.*, "Logitudinal threshhold changes in people with audiometric notches," Hear. Res. **141**, 220–228 (2000)], we studied interactions between NIHL and AHL in mouse; an animal with a short life span, with intrastrain genetic homogeneity to minimize variability and with interstrain differences in vulnerability which can be exploited to probe mechanisms. Using such models, we have uncovered evidence that early noise exposure can have an ongoing influence on the nature and progression of an age-related hearing loss. The nature of this age-related hearing loss exacerbation is special; it can occur even in ears without permanent threshold shifts from the noise and leads to massive loss of spiral ganglion neurons despite intact hair cell populations. Findings have practical importance for investigations of NIHL in animal models, and may have implications for clinical practices allocating noise-induced and age-related components of hearing loss as strictly additive. [Work supported by NIH.]

Contributed Paper

2aPPb5. Hearing levels in US adults aged 20–69 Years—National Health and Nutrition Examination Survey (NHANES) 1999–2004. William J. Murphy, Christa L. Themann, and Mark R. Stephenson (Hearing Loss Prevention Team, Natl. Inst. for Occupational Safety and Health, 4676 Columbia Pkwy., MS C-27, Cincinnati, OH 45226-1998)

The National Health and Nutrition Examination Survey (NHANES) is a nationally representative, population-based survey designed to assess the health and nutritional status of the civilian, noninstitutionalized U.S. population. Data were collected through a personal interview regarding health history and through physical examination. Earlier NHANES surveys were conducted on a periodic basis; however, in 1999, NHANES began collecting data on a continuing, annual basis. During NHANES I, which ran from 1971 to 1975, audiometric testing was conducted on adults aged 25–74 years. No subsequent testing of adults was conducted in the NHANES program until 1999, when NHANES began audiometric testing of adults aged 20–69 years. In 2004, the sampling from this age group was completed. This report examines the hearing levels for adults in the United States and compares them with the hearing data from NHANES I. Hearing levels are grouped by age and by ethnicity and gender.

TUESDAY MORNING, 6 JUNE 2006   BALLROOM D, 8:15 A.M. TO 12:00 NOON

Session 2aSC

Speech Communication: Enhancement and Multiplicity of Cues in Speech (Lecture/Poster Session)

Yi Xu, Cochair
Wolfson House, 4 Stephenson Way, London, NW1 2HE, UK

Carlos Gussenhoven, Cochair
Radboud Univ. Nijmegen, Ctr. for Language Studies, Postbus 9103, 6500 HT Nijmegen, The Netherlands

Chair's Introduction—8:15

Invited Papers

2aSC1. Defining and enhancing attributes for features. Kenneth N. Stevens (Res. Lab of Elec., Dept. EECS, & Div. Health Sci. & Tech, MIT, Cambridge, MA 02139, stevens@speech.mit.edu)

We take the view that words are represented in memory as sequences of bundles of binary distinctive features. It is further assumed that each feature is defined by a quantal relation between an articulatory attribute and a corresponding acoustic property that provides acoustic cues to the feature. In the speaking process, other acoustic properties are introduced in addition to the defining acoustic attributes, and these contribute additional cues that enhance the perceptual saliency of the feature. For example, the defining attributes for the voicing feature is the presence or absence of glottal vibration during an obstruent consonant, but there are several possible enhancing properties, such as the presence of aspiration noise and a modified fundamental frequency in the following vowel. In running speech, the saliency of a feature may be in jeopardy, especially when there is interfering noise or when gestural overlap weakens some of the cues, particularly those derived from the defining gesture for the feature. An example is the casual production of batman where the closure for /t/ may be eliminated by gestural overlap, but two enhancing gestures ($F2$ transition for the alveolar and the glottalization for /t/) are preserved. [Work supported in part by NIH Grant DC00075.]

2aSC2. Is some information perceptually primary? John Kingston (Ling. Dept., Univ. of Massachusetts, Amherst, MA 01003-9274, jkingston@linguist.umass.edu) and Randy Diehl (Univ. of Texas, Austin, TX 78712)

The existence of multiple correlates for phonological contrasts invites the hypothesis that one is perceptually primary and that the others enhance the primary correlate. (Stevens Acoustics Phonetics, MIT Press, 1998) particularly emphasizes the perceptual primacy of acoustic landmarks. Our experiments manipulate the quality or variability of the acoustic information in a landmark versus a neighboring nonlandmark to determine whether listeners rely on the landmark or the most reliable information source. Stimuli are synthetic stop-vowel sequences where the stop's place of articulation is conveyed by its burst spectrum—landmark—and the formant transitions to the following vowel—nonlandmark. Parameter values in the two intervals fall on opposite sides of, but close to, a category boundary and convey competing place information. Information is degraded in one interval by broadening the bandwidths of the spectral peaks or formants, or it is made more variable by varying the peaks frequencies or the following vowel quality. If landmark information is perceptually primary, then degrading or varying that information should shift place judgments more toward the category conveyed by the nonlandmark interval than vice versa. However, if the quality or consistency of the information matters, then manipulating either interval should produce comparable shifts in place judgments. [Work supported by NIH.]

2aSC3. Weighted multiple cues and phonological features. Patrice S. Beddor (Dept. of Linguist., Univ. of Michigan, 1190 Undergraduate Sci. Bldg., Ann Arbor, MI 48109, beddor@umich.edu)

Do speaker-listeners determine a primary phonological feature from among multiple cues for a phonological contrast? Linguistic analyses traditionally postulate a primary feature, yet experimental evidence indicates that speaker-listeners assign relative weights to multiple cues, in context-dependent ways, and that these weighted cues are part of the phonological grammar. In this paper, evidence for a noncategorical, cue-weighted perspective is drawn in part from perception of nasal consonants (Ns). English Ns are shorter before voiceless than before voiced obstruents (in VNC contexts), and anticipatory vowel nasalization is temporally more extensive before the shorter, prevoiceless Ns. Unsurprisingly, listeners' judgments of stimuli in which N duration and temporal extent of vowel nasalization were independently manipulated in CV(N)C sequences (coda C = [t] or [d]) broadly parallel the production data. But individual listeners differed in their cue weightings. Some listeners heard vowel nasalization and the nasal murmur as perceptually equivalent information about the presence of N, other listeners were highly sensitive to vowel nasalization, and still others' cue sensitivity depended on coda voicing. The patterns of responses are difficult to interpret in terms of a primary feature, and the phonological implications of these patterns will be discussed. [Work supported by NSF.]

2aSC4. Enhancing the durational enhancement of a tone contrast. Carlos Gussenhoven (Radboud Univ. Nijmegen, Letteren, Taalwetenschap, Postbus 9103, 6500 HT Nijmegen, The Netherlands and Queen Mary, Univ. of London, Mile End Rd., London E14NS)

In central and low Franconian dialects of German and Dutch, a lexical tone contrast (accent 1 versus accent 2) exists that is enhanced by a durational difference. While pitch differences vary according to intonation and sentence context, syllables are generally shorter when they have accent 1. In addition, vowels in those syllables are more open, in comparison with vowels in syllables with accent 2. Perception experiments with natural vowels whose acoustic durations had been manipulated revealed that high vowels are perceived as longer by Dutch listeners than midvowels when acoustic durations are equal. It is argued that vowel raising of longer vowels, or vowel lowering of shorter vowels, serves as an enhancement of duration differences. The tendency is widely attested to in the vowel systems of Germanic languages, where short ("lax") vowels are opener than long ("tense") vowels, and vowel height is used to enhance a *phonological* duration contrast. In the dialects investigated, vowel height is manipulated to enhance a durational difference, which in turn enhances a tonal contrast. The explanation of the durational illusion of vowel height is the listener's compensation of the universally shorter duration of higher vowels.

2aSC5. Enhancement between voice onset time (VOT) and $F0$: Evidence from Kera voicing perception. Mary D. Pearce (Univ. College London & SIL. Phonet. and Linguist., UCL, Gower St., London, WC1E 6BT, UK, mary_pearce@sil.org)

Previous research on production data from Kera (a Chadic tone language) has shown that tone plays a major contrastive role, whereas VOT has a lesser role which could indicate an "enhancing" relationship. This paper investigates the roles of VOT and $F0$ in Kera voicing perception, and compares this to the nontonal language English. Fifteen Kera and 15 English subjects were given voicing judgment tasks over 200 tokens that had been manipulated in PRAAT to cover a range of VOT and $F0$ values. The results show a marked difference in English and Kera perception reflecting the phonological differences between the two languages. For Kera, $F0$ is of major importance, while the VOT plays a lesser role in the judgment. The present findings are consistent with a view that whereas in English, the VOT is contrastive and $F0$ arguably acts as an enhancing cue, in Kera, the tone is the main contrastive cue to voicing perception while the VOT values enhance the tone cues. A bonus finding from the perception tests is that the relationship between VOT and $F0$ in Kera voicing perception tasks depends on their exposure to French during childhood.

2a TUE. AM

10:15

2aSC6. Commentary on invited papers for "Enhancement and multiplicity of cues in speech." D. H. Whalen (Haskins Labs., 300 George St., New Haven, CT 06511)

Enhancement has been shown to occur in a variety of contexts in the papers presented here. This commentary will examine the similarities of these phenomena and look for their proper level of description within linguistic theory.

10:35

2aSC7. Possible mechanisms of enhancement and multiplicity of cues. Yi Xu (Dept. of Phonet. and Linguist., Univ. College London, Wolfson House, 4 Stephenson Way, London NW1 2HE, UK, yi@phon.ucl.ac.uk)

It has been proposed that phonological contrasts are phonetically realized not only by their primary articulations, but often also by enhancing gestures [Kingston & Diehl "Phonetic knowledge," Language, **70**, 419–454 (1994); Stevens & Keyser, "Primary features and their enhancement in consonants," Language, **65**, 81–106 (1989)]. Meanwhile, enhancement appears to parallel the well-known phenomenon of multiplicity of cues in speech perception [Lisker "Voicing in English: A catalogue of acoustic features signaling /b/ versus /p/ in trochees." Lang. Speech, **29**, 3–11 (1986)]. An interesting question is therefore whether the two are one and the same. By a narrow definition, only articulatory maneuvers not obligated by the articulatory gestures of the phonetic units in question play an enhancing role. Thus not all cues that benefit perception are generated for the sake of enhancement. On the other hand, there may indeed be cases where genuine enhancement occurs, for which a further question is why is it needed in the first place? One possibility is that enhancement is not only for resisting noise in the speech environment, but also for handling the competition among communicative functions that are transmitted in parallel. That is, multiple communicative functions often share the same acoustic/articulatory dimension and such functional crowding may create the need to exploit additional means of encoding. If so, both enhancement and multiplicity of cues could be part of a live process that maintains the effectiveness of speech communication.

10:55–11:00   Break

Contributed Papers

All posters will be on display and all contributors will be at their posters from 11:00 a.m. to 12:00 noon.

2aSC8. Enhancement of the vowel quantity contrast in Japanese and German by F0 and vowel quality. Heike Lehnert-LeHouillier (Dept. of Linguist., Univ. at Buffalo, 602 Baldy Hall, Buffalo, NY 14260, hlehnert@buffalo.edu) and D.H. Whalen (Haskins Labs., New Haven, CT 06511)

Previous studies on the perception of vowel quantity in Japanese and German have shown an influence of F0 for Japanese and of vowel quality for German. The current study investigates F0 and vowel quality enhancement of vowel length perception in both languages. Native speakers of Japanese and German were presented with three series of stimuli: (*i*) vowel continua for the vowels /a/, /e/, and /i/ which differed in duration only; (*ii*) vowel continua for the same vowels which differed in duration and the steepness of the F0 fall over the vowel; and (*iii*) vowel continua for the same vowels containing both durational and quality cues. While vowel quantity in the first two series was perceived categorically by Japanese listeners, vowel quantity in the stimuli testing the influence of vowel quality was perceived as gradual rather than categorical by the same listeners. German listeners perceived all series as categorical. Thus F0 cues generally enhance the distinction of the vowel length contrast whereas the presence of quality cues facilitates the distinction in German but impedes the perception of the same distinction in Japanese. This suggests that enhancement of vowel quantity is governed by perceptual cues that are in part language specific.

2aSC9. The role of acoustic familiarity in the perception of L2 contrasts. Gautam K. Vallabha (Ctr. for the Neural Basis of Cognition, Carnegie Mellon Univ., 4400 Fifth Ave., Mellon Inst. 110, Pittsburgh, PA 15213)

Difficulties in L2 perception are often attributed to assimilation of the L2 contrast by an L1 category or to interference by L1-relevant acoustic cues [Iverson *et al.* "Phonetic training with acoustic cue manipulations: A comparison of methods for teaching English /r/-/p/ to Japanese adults," J. Acoust. Soc. Am. **118**, 3267–3278 (2005)]. These approaches assume that cue weights change slowly over developmental time and are relatively constant for short time scales. An alternative view is that cue extraction and weighting are dependent on the overall familiarity of the current sound, with more familiar sounds facilitating more robust cue extraction and better discrimination [e.g., McFadden and Callaway, JEP:HPP, **25**, 543 (1999)]. This hypothesis was tested with the perception of English sounds by Japanese adults. The Japanese listeners are insensitive to a cue (F3 onset) when it is part of (putatively unfamiliar) English /ra/ and /la/ sounds. The effect of familiarity was evaluated by fixing the range of F3 onsets and manipulating the rest of each sound in two ways: (1) by changing the transition durations to be more suitable for a moraic rhythm and (2) by changing the F1, F2, and amplitude trajectory to be more like the Japanese /r/. Discrimination improved markedly with the latter manipulation, suggesting that cue sensitivity is conditioned upon overall acoustic familiarity. [Work supported by NIH MH64445.]

2aSC10. Learning to ignore a perceptual cue: Nonnative listeners outperform native listeners. Mirjam Broersma (Max Planck Inst. for Psycholinguisti., P.O. Box 310, 6500 AH Nijmegen, The Netherlands, mirjam.broersma@mpi.nl)

Native and nonnative listeners categorized /v/ and /f/ at the end of English nonwords. For each participant, the duration of the previous vowel was kept constant, so that it was not informative and sometimes mismatched other information in the signal. Vowel duration was varied between participants. Previously presented results [M. Broersma, J. Acoust. Soc. Am. **117**, 3809–3901 (2005)] showed that native English listeners relied strongly on the misleading vowel duration cue. For Dutch listeners, no effect of vowel duration was found. Due to the redundancy of information in the signal, Dutch listeners categorized the contrast more categorically than English listeners. New analyses investigated whether Dutch listeners did not attempt to use vowel duration at all, or whether they learned to ignore the misleading cue more easily than the English listeners did. The results showed that Dutch listeners did use vowel duration initially, but stopped using this cue after very few trials. By the end of the practice part (33 trials) the effect of vowel duration had fully disap-

peared. The English listeners used vowel duration as a voicing cue throughout the experiment. This suggests that it may be easier to learn to ignore uninformative perceptual cues in a nonnative language than in one's native language.

2aSC11. Effects of task complexity on perceptual organization of speech cues by older listeners. Jessica A. Rossi-Katz and Kathryn H. Arehart (Dept. Speech Lang. and Hearing Sci., Univ. of Colorado, 409 UCB, Boulder, CO 80309)

Older adults with and without clinically significant hearing loss often report difficulty communicating in multitalker environments. Past work in our laboratory has shown that older listeners with hearing loss benefit less from differences in talkers' vocal characteristics (e.g., fundamental-frequency differences) relative to younger listeners with comparable hear-

ing losses. The present study examined the extent to which age-related changes in the perceptual organization of speech cues are due to task demand. Two experiments used the Coordinate Response Measure to examine how the benefit listeners derive from talker characteristics interacts with the complexity of the response task as a function of both age and hearing status. Stimuli were amplified to assure audibility for listeners with hearing loss. Task complexity depended on whether the distracting signal was linguistically meaningful as well as the level of auditory processing required (discrimination versus identification). Older listeners generally benefited more from talker characteristics when task complexity was reduced (e.g., nonmeaningful distracter). The results are interpreted in the context of an information-processing model in which the increased difficulty of older listeners in multitalker environments is a consequence of reallocating limited resources towards the initial stages of processing at the expense of higher-level stages. [Work supported by Centers for Disease Control.]

TUESDAY MORNING, 6 JUNE 2006

ROOM 552B, 8:00 A.M. TO 12:00 NOON

Session 2aSP

Signal Processing in Acoustics: General Topics in Signal Processing in Acoustics

David J. Moretti, Chair

Naval Undersea Warfare Ctr., Code 71, Newport, RI 02841

Contributed Papers

8:00

2aSP1. A biomimetic robotic system for localizing gunfire. Socrates Deligeorges, Aleksandrs Zosuls, David Mountain, and Allyn Hubbard (Dept. Biomed. Eng, Boston Univ., 44 Cummington St., Boston, MA 02215, sgd@bu.edu)

Using a biomimetic approach, a new method of acoustic signal processing was created which has tremendous advantages in complex acoustic environments. The biomimetic approach was used as the basis for a system to localize and identify sound sources in noisy and reverberant conditions. The algorithms are based on mammalian hearing and mimic the acoustic processing of the auditory periphery and midbrain. The system uses spectro-temporal cues exploited by the auditory system including such features as interaural time difference (ITD), interaural level difference (ILD), spectral profile, and periodicity content. The initial system of algorithms were designed and tested using the EARLAB [earlab.bu.edu] software modeling environment. The system of algorithms was then adapted to a mixed-signal real-time hardware solution and mounted on an iRobot PackBot robotic platform to perform simple behavioral tasks. The integrated system can detect and localize gunfire in a complex, reverberant acoustic environment and orient a camera towards the shooter. Accuracy in field tests with live fire was 1.5 deg in azimuth. The performance of the prototype platform demonstrates the potential of the biomimetic approach and its applications to practical problems for commercial, civilian, and military acoustic processing. [Work funded by ARL:DAAD19-00-2-0004.]

8:15

2aSP2. The multiple coherence function for sound field prediction. Torsten Leth Elmkjær (Terma AS, Hovmarken 4, DK-8520 Lystrup, Denmark, tln@terma.com)

From the *uniqueness theorem* it is well known that the sound field in a region exterior to the source region can be determined from knowledge of the pressure or the normal fluid velocity *everywhere* on the enclosing surface. In practice the acoustical field quantities are obtained from space-time discretization. The space-time samples on the enclosing surface and

arbitrary field points in the interior to the volume will be considered as input and output processes to a multiple-input–multiple-output (MIMO) system, respectively. The *multiple coherence function* $\gamma_{y:x}^2$ provides a measure of linearity between a set of inputs and an output *independent* of the correlation among the inputs. Various system imperfections, however, will lead to a nonunity $\gamma_{y:x}^2$ In particular, spatial sampling, similar to time-domain sampling, requires limited bandwidth to avoid aliasing. Therefore, a sufficient number of sensors must be allocated to ensure $\gamma_{y:x}^2 \approx 1$. In this paper a spherical enclosure is investigated. The objective is to express $\gamma_{y:x}^2$ as a function of internal field point position for different space-time sampling strategies. Simulation results for single plane wave, diffuse fields, and multiple-source fields is presented. Measurement results obtained in a reverberant room are discussed. The work is also useful in active control system design.

8:30

2aSP3. Adaptive filtering algoritm for time-variant symmetric α-stable signals. Torsten Leth Elmkjær (Terma AS, Hovmarken 4, DK-8520 Lystrup, Denmark, tln@terma.com)

The existence of non-Gaussian distributed impulsive signals has roots in the *generalized central theorem*. In particular, the symmetric α-stable (SαS) distribution has been used to model heavy-tailed phenomena encountered in communication, underwater acoustics, and radar. Most previous work in the field of active control uses some of the numerous variants of LMS algorithm extended with the *filtered-x* scheme. The stable processes, however, do not possess finite second-order moments. Hence, the filter should be based on l_p-norm optimization. The p-norm enters the expression for the tap-weight update $[p(t) < \hat{\alpha}(t)]$. In this work a new adaptive algorithm referred to as regularized normalized least−mean p−norm algorithm $[\epsilon \text{NLM}p(t)]$ is presented. A running estimate $\hat{\alpha}(t)$ of the time−variant characteristic exponent of the stable signal is obtained. Hence, by adaptively tuning the filter p−norm a more optimal filter performance results in time−variant situations. Simulation analysis provides insight into the adaptive filter performance for various time−invariant and time−variant noisy signals for $1 < \alpha(t) \leqslant 2$. Moreover, it is demonstrated

3271   J. Acoust. Soc. Am., Vol. 119, No. 5, Pt. 2, May 2006

151st Meeting: Acoustical Society of America 2006   **3271**

that the filter performance is only marginally degraded compared with LMS algorithm for ordinary time−invariant Gaussian noise signals. Performance of an active control system exposed to time−varying heavy−tailed noise signals is presented.

8:45

2aSP4. Loudspeaker defect analysis using ultrasonic harmonic characterization. W. Scott Galway, Matthew A. LaBruzzo, Robert D. Celmer (Acoust. Program & Lab., Univ. of Hartford, 200 Bloomfield Ave., W. Hartford, CT 06117), and Daniel Foley (Independent Consultant, Marlborough, MA 01752)

This study presents the results of an ongoing investigation of defect detection in transducers. Loudspeaker assembly faults, such as a rubbing voice coil, bent frame, loose spider, etc., have traditionally been detected using experienced human listeners at the end of a production line, or through the use of low-order harmonics for total harmonic distortion measurement. The findings of Thompson et al. ["Higher-order harmonic signature analysis for loudspeaker defect detection," J. Acoust. Soc. Am. **114**, 2400 (2003)] described a new method wherein the total energy of high-order harmonics groups (10th through the 20th, or 31st through the 40th), were measured and analyzed. This approach demonstrated the ability to delineate distinct signatures that correlated to the root cause of audible rub and buzz distortions. This paper presents the results of a follow-up study using equipment that collected harmonic data up to 100 kHz. Results are presented for ten 5-in. midrange loudspeakers, in which trends for specific acoustic signatures were correlated to known defects. The use of a computer-based electroacoustic measurement and analysis system using frequencies that are inaudible but beneficial for analysis of defects are discussed. [Work supported by Listen Inc., Boston, MA.]

9:00

2aSP5. Adaptive collaborative array trajectories for optimum passive detection. Granger W. Hickman and Jeffrey L. Krolik (Dept. of Elec. and Comput. Eng., Duke Univ., Hudson Hall, Durham, NC 27708)

Two major challenges faced by sensor arrays used for passive sonar detection in littoral ocean channels are the suppression of highly dynamic interferers and signal wave-front mismatch resulting from complex multi-path propagation. Large static arrays require accurate signal wave-front modeling and a highly stationary noise field to achieve their maximum potential array gain. In this paper, the detection performance of collaborative unmanned underwater vehicles (UUVs) towing short-line arrays is evaluated. In particular, vehicle trajectories which maximize overall probability of detection are computed as a function of the noise-field directionality and target location probability distribution. The UUV trajectories are constrained to share common endpoints connected by a series of line segments for each vehicle. The probability of at least one far-field detection by at least one UUV over the course of their transit is then maximized with respect to the sequence of the vehicles' headings. The results for a 2D horizontally isotropic noise field indicate that trajectories with a diversity of endfire (or equivalently broadside) bearings maximize detection performance. Scenarios with multiple interferers lead to solutions which minimize backlobe masking of the target. [Work supported by ONR.]

9:15

2aSP6. Data error covariance matrix for vertical array data in an ocean waveguide. Chen-Fen Huang, Peter Gerstoft, and William S. Hodgkiss (Marine Physical Lab., Scripps Inst. of Oceanogr., La Jolla, CA 92093-0238)

Information about the data errors is essential for solving any inverse problem. The likelihood function plays a critical role in describing the data uncertainties in geoacoustic inversion. The choice of likelihood function depends on the statistics of the errors (the difference between observed and estimated fields). In all work to date, the likelihood function has been derived based on an assumption of Gaussian errors. Typically, the errors are assumed to be independent, identically distributed with equal variance (referred to as the error variance), and the error variance is estimated as

part of the optimization. Recently, there has been interest in estimating a more full data error covariance matrix. To estimate a truly full data error covariance matrix, we adopt a maximum-likelihood approach based on ensemble averages using the observed errors over many inversions. The approach is illustrated using data obtained during the ASIAEX 2001 East China Sea experiment. The parameter uncertainties resulting from incorporating the full data error covariance matrix are compared with those obtained from the simplified data error covariance matrix characterized by the error variance alone.

9:30

2aSP7. Increasing cross-talk immunity in simultaneous multiple acoustic sources measurements using coded signals. Shu Li and Ning Xiang (Architectural Acoust., School of Architecture, Rensselaer Polytechnic Inst., 110 8th St., Troy, NY, 12180)

Maximal-length sequences (MLS) and related signals demonstrate excellent correlation properties including a pulselike autocorrelation function and a relatively smaller-valued cross-correlation function. They have been increasingly applied in acoustical measurements [Xiang et al., J. Acoust. Soc. Am. **117**, 1889–1894 (2005)]. Recent work employs specialized FFT-based algorithm [Daigle and Xiang, J. Acoust. Soc. Am. **119**, 330–335 (2006)] using MLS-related signals into simultaneous multiple acoustic sources measurements (SMASM). The resulting impulse responses of acoustic channels, however, suffer from cross-talk noise. To enhance cross-talk immunity, this paper proposes an algorithm using pseudoinverse algorithm to improve signal-to-noise ratio in the SMASM. This proposed algorithm applies specialized correlation method and pseudoinverse processing to implement acoustic measurements for the SMASM technique. This paper discusses both simulation and experimental results of the SMASM with the pseudoinverse algorithm to demonstrate considerable improvements in signal-to-noise ratio. This paper also evaluates the efficiency of pseudoinverse algorithm for the increased cross-talk immunity.

9:45

2aSP8. Multi-static target tracking using a bistatic invariance constraint. Jorge Quijano, Manish Velankar, and Lisa Zurk (Portland State Univ., 1900 SW Fourth Ave., Ste. 160-17, Portland OR 97207, zurkl@cecs.pdx.edu)

The ability to track moving targets in shallow water channels is critically effected by bottom reverberation which can generate false alarms and substantially increase the domain of possible tracks. Even with application of advanced tracking algorithms, false alarms degrade performance and increase computational burden. The invariance principle, which predicts structure in the frequency-time domain, has been applied to passive sonar systems but has not been considered for active systems. In this work, a mathematical expression (called the bistatic invariance principle, see companion paper) is proposed as a constraint to discriminate target detections from false alarms and thus reduce the search domain for multi-static tracking algorithms. The constraint does not require knowledge of the underwater environment and is thus robust to environmental uncertainty. The bistatic invariance principle provides a relationship between changes in the bistatic range of a moving target to changes in the source frequency, and this relationship can be used to track the target position over time in broadband active sonar systems. Simulations of the bistatic invariance principle for multi-static active systems are presented and discussed and multi-static tracking performance is characterized for several choices of target scattering matrices and different environmental conditions.

10:00−10:15   Break

10:15

2aSP9. A time-frequency approximation with applications in target recognition. Patrick Loughlin (Univ. of Pittsburgh, 348 Benedum Hall, Pittsburgh, PA 15261)

In active sonar, the received signal can often be modeled as a convolution of the transmitted signal with the channel impulse response and the target impulse response. Because the received signal may have a time-varying spectrum, due, for example, to target motion or to changes in the

channel impulse response, time-frequency methods have been used to characterize propagation effects and target effects, and to extract features for classification. In this presentation, we consider the Wigner time-frequency representation of a received signal modeled as the convolution of the transmitted signal with the channel and target responses. We derive a simple but very insightful approximation that shows the effects of the magnitude and phase of the frequency response of the target and of the channel on the Wigner representation of the transmitted signal. The results suggest possible features for classification, which we discuss. We also consider time-varying effects on the Wigner representation, such as changes in reflected energy due to changes in aspect angle, which we model by amplitude modulation. [Research supported by ONR Grant No. N000140610009.]

10:30

2aSP10. Model-based blind deconvolution for noisy dynamic channels. Michael Roan (Virginia Polytechnic Inst. and State Univ., Blacksburg, VA 24060) and Mark Gramann (Aerosp. Corp., Chantilly, VA)

It is common in acoustics to measure a signal that has been degraded by propagation through an unknown, noisy channel prior to measurement. While only the degraded measured signal is available, the data of interest are the original signal and the channel parameters. Often, it is desirable to reverse the filtering process by application of an inverse filter to recover the original signal. When neither the input signal properties nor the channel properties are known, this process is known as blind deconvolution (BDC). Typically, BDC algorithms assume noiseless, stationary propagation channels and input sources. These assumptions are usually violated in practical applications (e.g., noisy multipath propagation environments with moving source and receiver). To model these effects, predictive techniques are applied to incorporate *a priori* information about the system into the existing blind processing framework. The original contributions of this work follow. First, a novel formulation of the extended Kalman filter (EKF) is proposed. This allows incorporation of *a priori* information into gradient-based blind processing algorithms. This formulation is then applied to the natural gradient (NG) BDC algorithm. Finally, results are presented that suggest significant improvement in signal recovery performance over the NG BDC algorithm for dynamic noisy channels.

10:45

2aSP11. Detection of phase-coupled noises in the ocean. M. H. Supriya and P. R. Saseendran Pillai (Dept. of Electron., Cochin Univ. of Sci. and Technol., Kochi—682022, India)

Estimates of power spectral density are being used in a variety of signal processing applications. For the precise characterization of certain noise sources in the ocean, the phase blind power spectral estimation does not yield the required result. Estimates of the bispectrum and bicoherence have been found useful in detecting non-Gaussianity and nonlinearity, in system identification, as well as transient signals. The power spectrum, in general, fails to quantify any nonlinear interactions between the component frequencies. Such interactions induced by second-order nonlinearities give rise to certain phase relations called quadratic phase coupling (QPC). Bispectral analysis, which is based on the third-order cumulant sequence of the signal, can reveal the presence of as well as provide a measure to quantify such phase couplings. Bispectral analyses of various noise data records collected from different ocean noise sources have been carried out and it is seen that the bispectrum is not identically zero, implying the non Gaussian nature of the noise data. This paper presents the bispectral estimation approach for identifying the nonlinearities in the underwater noise-generating mechanisms as well as detecting the presence of quadratic phase couplings, thus leading to the identification of the noise sources.

11:00

2aSP12. Exploiting temporal coherence models to extend detection range. Ronald A. Wagstaff (Natl. Ctr. for Physical Acoust., University, MS 38677)

Turbulent flow over acoustic sensors and other sources of noise can severely corrupt signals and reduce detection range. The ranges can be much greater when based on the temporal coherences of signals and noise instead of their magnitudes. The signals need not be perfectly coherent to be detected at longer ranges, just more coherent than noise. This is illustrated using a fluctuation-based temporal coherence signal-processing model. This model uses measured data to produce temporal coherence patterns that are random for noise, deterministic for signal, and a blend of the two for small, e.g., negative, signal-to-noise ratio (SNR). Fluctuation-based processing techniques are presented for temporal coherence pattern generation, recognition, and automatic identification of signal presence or only noise in a frequency bin. It is shown for both underwater and atmosphere acoustic data that a signal's influence in a frequency bin alters the usual random "noiselike" temporal coherence patterns by causing them to cluster. Hence, signal-altered patterns can be recognized and the signals preserved. Corresponding "noise-only" patterns, being random, can also be recognized and attenuated, or eliminated. Using these models results in increased SNR and automatic signal detection at longer ranges with greater confidence. [Work supported by ARDEC and SMDC.]

11:15

2aSP13. Generating simulated reverberation using noiselets. Edmund J. Sullivan, Robert P. Goddard (Prometheus Inc., 103 Mansfield St., Sharon, MA 02067), Hyman A. Greenbaum (Anteon Corp., Middletown, RI 02842), and Kevin P. Bongiovanni (Naval Undersea Warfare Ctr., Newport, RI 02841)

Of the approaches to generating stochastic realizations of reverberation time series, the simple point scatterer model (PSM) is appealing because the statistics evolve naturally as the number of scatterers increases. Unfortunately, the number of eigenrays required is often overwhelming, especially for broadband signals. The more common "variance factorization model" (VFM) uses far fewer eigenrays to compute a time-dependent estimate of the variance of the reverberation, and applies the spectral factorization theorem to generate a sequence of short-time spectra, which is inverse transformed and concatenated to form the desired time series. However, the variance and factorization steps scale with the square and the cube, respectively, of the number of receiver channels, so the advantage is lost for element-level simulations of complex receivers. The "noiselet model" is conceptually similar to the PSM, but the transmit pulse is replaced by an ensemble of precomputed "noiselets," which are sums of randomly weighted and delayed copies of the original transmit pulse. The number of eigenrays is thus comparable to the VFM, but scaling is linear in the number of channels, rendering this method tractable for real-time broadband element-level simulation. Examples are shown for a linear FM pulse at 10 kHz and 400-Hz bandwidth.

11:30

2aSP14. A phenomenological model for sonic crystals based on artificial neural networks. Elies Fuster-Garcia, Vicent Romero-Garcia, Juan V. Sanchez-Perez (ACARMA and Dep. de Fisica Aplicada, ETSICCP, UPV, Cno. de Vera s/n, 46022 Valencia, Spain, jusanc@fis.upv.es), Luis M. Garcia-Raffi, and Enrique A. Sanchez-Perez (ETSICCP, UPV, 46020 Valencia, Spain)

In this paper we develop a phenomenological model that simulates the acoustic attenuation behavior of sonic crystals. The input of the model is a

set of parameters that characterizes each experimental setup and the output is the associated simulated attenuation spectrum. The model consists of a combination of multiresolution analysis based on wavelet functions and a set of artificial neural networks. An optimized coupling of this tool allows us to reduce drastically the experimental data needed and also to have a fast computational model that can be used for technological purposes. [This research has been partially supported by the Generalitat Valenciana (Spain), under Grant GV04B-371, by Comision Interministerial de Ciencia y Tecnologia of Spain, under Grant BFM2003-02302, and by Comision Interministerial de Ciencia y Tecnologia of Spain, Contract Ns. MAT 2003-04993-03.]

2aSP15. Nonlinear behavior of interacting harmonics in audio spectra. V. Vijayakumar and C. Eswaran (Multimedia Univ., FOSEE, Jalan AIr Keroh Lama, 75450, Melaka, Malaysia)

In this paper we will show how individual harmonics of a frequency-modulated dynamic audio spectrum undergo interactions. This results in nonlinear behavior of the spectrum. We have illustrated the nonlinear behavior with respect to the number of harmonics and the modulation parameter. It is shown that an audio spectrum containing higher-order harmonics is more susceptible to interation among itself. Musical instrument spectra are used to illustrate this effect and audio samples are provided.

TUESDAY MORNING, 6 JUNE 2006   ROOM 553AB, 8:00 A.M. TO 12:00 NOON

Session 2aUW

Underwater Acoustics, Acoustical Oceanography, and Animal Bioacoustics: Joe Blue Memorial Session I: Shallow Water and Marine Animal Acoustics

Edmund R. Gerstein, Cochair
Leviathan Legacy Inc., 1318 S.W. 14th St., Boca Raton, FL 33486

George V. Frisk, Cochair
Florida Atlantic Univ., Dept. of Ocean Engineering, 101 North Beach Rd., Dania Beach, FL 33004-3023

Chair's Introduction—8:00

Invited Papers

8:05

2aUW1. The direct path impulse response in bubbly shallow water. Ralph Goodman, Jerald Caruthers, and Timothy Ruppel (Stennis Space Ctr., Univ. of Southern Mississippi, Bay St. Louis, MS 39529)

It is well known that bubbles, even in small numbers, cause significant changes in the acoustic properties of seawater. In shallow water bubbles enter the sea, either by wind-driven breaking waves or by surf-induced rip currents. Both have been observed in acoustic transmission experiments [IEEE, JOE **25**, 507–515 (2000); J. Acoust. Soc. Am. **106**, 617–625 (1999)]. Wind-driven bubbles were observed, during a period of high winds, to reduce sound speed from the undisturbed sea condition of 1520 m/s to about 1494 m/s. From bubble theory, it is estimated that the average bubble void fraction was approximately 2.6×10^{-6}. For a distribution of bubble radius size, $n(a)$, there will be a frequency-dependent sound speed and absorption related by the Kramers-Kronig relation, from which the change in a pulsed signal can be determined. By using a bubble size distribution of $n(a) = Na \times 10^{-4}$, for radii between values of 200 and 10 microns and the void fraction estimated above, the dispersion and absorption are calculated. The impulse response can be calculated as a function of range. [Work supported by the Office of Naval Research.]

8:25

2aUW2. Modeling nonlinear acoustics in shallow water. B. Edward McDonald (Naval Res. Lab, Washington, DC 20375, mcdonald@sonar.nrl.navy.mil) and William A. Kuperman (Scripps Inst. of Oceanogr., La Jolla, CA 92093)

Lessons learned in modeling nonlinear acoustics and weak shock propagation in deep water are helpful in application to shallow water (mine countermeasures in particular), but the medium is much more complicated. Numerous improvements are needed; Joe Blue's work touched on several of them. First is the need to treat higher propagation angles. Second is the presence of bubbles near the surface and possibly in the bottom. Finally, there are problems introduced by nonlinear propagation in granular sediments. These include greatly increased nonlinearity parameter and time-domain formulation of frequency—linear attenuation. Recent progress in the application of the NPE model [McDonald and Kuperman, "Time domain formulation for pulse propagation including nonlinear behavior at acoustic," J. Acoust. Soc. Am. **81**, 1406–1417 (1987)] to mine countermeasures will be outlined. [Work supported by the Office of Naval Research.]

2aUW3. Sound waves and shear waves in sediments. Michael J. Buckingham (MPL, SIO, UCSD, 9500 Gilman Dr., La Jolla, CA 92093-0238, mjb@mpl.ucsd.edu)

In a long and illustrious career, Joe Blue made many significant contributions to the discipline of underwater acoustics, although he is probably best known for his work on the design and calibration of hydrophones and hydrophone arrays. Such instrumentation has many applications, including measurements of the geoacoustic properties of marine sediments. In this paper, recent developments on the properties of sound waves and shear waves in sediments will be summarized and certain difficulties that are encountered when attempts are made to match theory to data will be highlighted. Central to the discussion are the compressional-wave and shear-wave dispersion relations, that is, phase speed and attenuation as functions of frequency. In a saturated sediment, the phase speeds and attenuations also depend on various physical properties, for example, the porosity, density, grain size, and overburden pressure (depth). These dependencies will be illustrated using extensive data sets taken from the literature. Any satisfactory theory of waves in sediments must yield correctly not only the dispersion relations but also all the dependencies on the physical properties. A recently developed theory that satisfies these criteria will be discussed in the context of the available data. [Research supported by ONR.]

2aUW4. Joe Blue's influence on low-frequency, shallow-water acoustics. George V. Frisk (Florida Atlantic Univ., 101 N. Beach Rd., Dania Beach, FL 33004)

As Superintendent of the Naval Research Laboratory's Underwater Sound Reference Division, Joe Blue played a key role in the advancement of low-frequency, shallow-water acoustics. Specifically, his leadership and oversight of this unique facility provided an engineering science environment in which low-frequency transducers were developed, calibrated, and maintained. These assets were made available to the research community in ocean acoustics at reasonable cost and have offered dependable, reliable service for more than 25 years. This talk focuses on advances made using the J15-3 transducer, which emerged as a workhorse in numerous experiments in low-frequency (50–500 Hz), shallow-water acoustics. A series of experiments is described in which this transducer was used as the principal source of sound. These measurements led to an increased understanding of acoustic propagation, geoacoustic inversion, and physical oceanography in shallow water. [Work supported by ONR.]

2aUW5. A long-range, high-resolution Doppler sonar for fishery stock assessment. Harry A. DeFerrari (RSMAS—Univ. of Miami, 4900 Rickenbacker Cswy., Miami, FL 33149)

A new approach is developed for horizontal-looking active sonar that uses continuous bi-/multistatic transmission of very long period m-sequences. Compared to conventional monostatic ping rather than listen sonar, this approach improves the Doppler resolution by a factor of 20 and increases FOM by perhaps 13 dB by increasing the temporal coherent processing gain. The improved time/Doppler resolution helps to discriminate fish school from bottom reverberation, and improves the resolution of taxonomic parameters of fish assemblages. The improved FOM allows coverage of large areas of shallow oceans with source levels low enough to avoid environmental concerns for marine mammals. The principal obstacle to continuous transmission sonar operation is the masking of the target return by the time and frequency leakage of the direct path arrivals. Unique time-correlation property of m-sequences elimination time leakage and a process calls coordinate zeroing (CZ) eliminates all the Doppler leakage. A linear temporal Doppler search (LTDS) algorithm is used to shift the detection problem to a high-resolution pulse arrival-time/Doppler plane with a noise-limited background. Ultrafast Hadamard transforms lighten the computational burden. The combination of these ideas is presented including the theory, numerical experiments that demonstrate the concept, and verification with real ocean signals.

2aUW6. Development and application of high-frequency sonars for imaging naturally occurring plumes in the ocean. David R. Palmer (NOAA/Atlantic Oceanogr. and Meteorological Lab., 4301 Rickenbacker Cswy., Miami, FL 33149)

Hydrothermal plumes are formed when hydrothermal solutions are discharged from chimneylike vents on the ocean floor where adjacent tectonic platesare separating. Their study provides information on the transfer of heatand chemicals from the solid earth to the ocean. When high-frequency acousticswas proposed for imaging and studying hydrothermal plumes, Joe Blue contributedin a general way to the development of the effort. In this presentation the history of this effort is reviewed. After an introduction to hydrothermal venting, some of the problems addressed in the development and application of the sonar are discussed. These include sonar calibration, identification of the dominant scattering mechanism, estimating cross sections, and the intrinsic statistical variability associated with incoherent scattering. The results obtained from four sea trials are recalled. These trials involved either three-person, deep-diving submersibles or the JASON/MEDEA ROV system. The opportunities provided by the somewhat surprising progress that has occurredare discussed. Finally, an application of the developed technology is mentioned, acoustical imaging of the bubble plume emitted from the underwater volcano Kick em Jenny.

2aUW7. The art of merging physics and math with biology and psychology to study marine animal acoustics. Mardi C. Hastings (Office of Naval Res., Life Sci. Res. Div., 875 N. Randolph St., Arlington, VA 22203-1995, mardi_hastings@onr.navy.mil)

Joe Blue's most recent contributions to marine animal acoustics resulted from successful collaborations with biologists and psychologists, each bringing their individual expertise to the table and working together to understand how the characteristics of the acoustic field generated by a moving vessel could provide critical insight into the observed behavior of right whales and manatees.

Joe's hands-on approach of going out and actually recording and measuring the sound fields provided data that not only verified his analyses, but also enhanced practical understanding of acoustics in the animal bioacoustics community. His work demonstrated the importance of physics and math in studying complex multidisciplinary acoustic problems and resolving perplexing dichotomies. Examples will be presented showing that Joe's collaborative style is a model for everyone to follow.

10:45

2aUW8. Simulation models of click production in dolphins.   Nikolay A. Dubrovskiy and Igor A. Urusovskiy   (Andreyev Acoust. Inst., Shvernika Ul, #4, Moscow 117036, Russia)

The simulation model (SM) of dolphin click source originally presented by [Dubrovskiy and Giro, "Echolocation in Bats & Dolphins," Chap. 10, 59–63 (2003)] was further developed by Gladilin, Dubrovskiy, Mohl, and Walberg, Acoust. Phys. **50**, 463-468 (2004). The SM describes movements of a solid sphere frozen in a compliant ring. A jet of air in a duct connecting air cavities brings the sphere into self-oscillations. The SM allows calculating the waveform and sound pressure of the generated clicks. Here the mathematical and simulation model (MSM) of click production in dolphins is further developed. The MSM describes radiation of clicks by bodies comprising of waterlike tissue with different shapes (a ring, a blunted cone) pulsating and (or) oscillating under action of muscles tension. Calculations are made for the energy efficient case when acting force is consecutively applied not to the entire body but to its thin layer, which resulted in action of the progressing wave array. Directivity patterns of such a waterlike compliant body in main directions are similar to that of a rigid body with the same shape and dimensions. Comparisons are made of MSM predictions with properties of an actual radiation field of dolphins (in clicks wave forms, sound pressure levels, and directivity patterns) and dimensions of biological structures supposedly responsible for click production in dolphins.

11:05

2aUW9. A dolphin acoustic tapetum?   Sam Ridgway, Brad Blankenship, Dorian Houser, Don Carder   (SPAWAR Systems Ctr. San Diego, Div. D235, 53560 Hull St., San Diego, CA 92152-5001), and Carl Hoh   (Univ. of California, San Diego, CA 92093)

Many mammals including dogs, cats, deer, and dolphins, but not humans, have a reflective membrane at the back of the eye. This tapetum lucidum affords the light-sensitive retinal cells a second chance for photon-photoreceptor stimulation, enhancing low light visual sensitivity. Dissection of preserved specimens and computed tomography (CT) scans of live dolphins have shown prominent air cavities around the ear, especially medial and posterior. Histologic evaluation of peri-otic tissue revealed epithelial lined air sinuses and erectile tissue containing nerves, ganglia, smooth muscle bands, vascular spaces, and periarterial venous networks. Two dolphins were trained to periodically recognize and respond to a 1.5-s, 70-kHz target tone. In three experiments, 20 target tones were presented through a jaw phone after injection of 18 Fluro-deoxy-d-glucose (FDG). Subsequent positron emission tomography (PET) scans revealed that the tone, to which the dolphin had responded and been rewarded numerous times in the past, elicited a marked increase in metabolism in the erectile tissue medial to the stimulated ear but not the contralateral ear. Tissue around reflective air sinuses medial to the ear become active with acoustic stimulation, suggesting that sinus air may be shaped so as to provide an "acoustic tapetum" for dolphins.

11:25

2aUW10. Of manatees and men: A future: Thanks to you, Joe.   Edmund Gerstein, Laura Gerstein, Joseph Blue   (Leviathan Legacy Inc., 1318 SW 14th St., Boca Raton, FL 33486), and Steve Forsythe   (Naval Undersea Warfare Ctr., Newport, RI 02841)

A comprehensive series of underwater psychoacoustic tests was conducted to measure the hearing abilities of the West Indian manatee under varying acoustic conditions. Pure tones, complex noise and real sounds, boat noise and species-specific calls, were presented to captive manatees under various acoustical conditions. The results from more than 30 000 threshold trials define the first behavioral audiograms, directional hearing, and critical masking ratios for any sirenian species. Psychoacoustic tests demonstrate the noise produced by slow-moving boats is more difficult for manatees to hear and locate than the high-energy cavitation noise from faster moving vessels. The dominant frequency spectra of slow-moving boats fall near the fringe of their low-frequency hearing limits and below critical masking ratios. Complementing underwater ambient noise and vessel noise measurements revealed physical propagation factors that further render manatees vulnerable to watercraft collisions in their shallow-water habitats. Ironically, slow speed zones implemented to protect manatees may exacerbate the problem, making vessels difficult to hear, while increasing transect times and opportunities for collisions. By exploiting the manatee's best hearing capabilities, it may be possible to mitigate collision risks with a low-intensity, environmentally friendly acoustic warning device that alerts manatees of approaching vessels. [Research supported by DOD Navy Legacy Program, Inland Navigation District, FFWCC]

Contributed Paper

11:45

2aUW11. Listening to fish: Applications of passive acoustics to fisheries. Rodney Rountree, Francis Juanes (Dept. of Natural Resources Conservation, UMass Amherst, Holdsworth Hall, Amherst, MA 01003, rrountree@fishecology.org), and Cliff Goudey (MIT Sea Grant College Program, Cambridge, MA 02139)

Passive acoustics is a rapidly emerging field of marine science that until recently has received little attention from fishery scientists and managers. In its simplest form, it is the act of listening to the sounds made by fish and using that information as an aid in locating fish so that their habitat requirements and behaviors can be studied. We believe that, with the advent of new acoustic technologies, passive acoustics will become one of the most important and exciting areas of fisheries research in the next decade. However, widespread unfamiliarity with the technology, methodologies, and potential of passive acoustics has hampered the growth of the field and has limited funding opportunities. Herein, we provide an overview of important new developments in passive acoustics together with a summary of research, hardware, and software needs to advance the field.

2a TUE. AM

Meeting of the Standards Committee Plenary Group

to be held jointly with the meetings of the

ANSI-Accredited U.S. Technical Advisory Groups (TAGs) for:
ISO/TC 43, Acoustics,
ISO/TC 43/SC 1, Noise,
ISO/TC 108, Mechanical vibration and shock,
ISO/TC 108/SC 2, Measurement and evaluation of mechanical vibration and shock as applied
to machines, vehicles and structures,
ISO/TC 108/SC 3, Use and calibration of vibration and shock measuring instruments,
ISO/TC 108/SC 4, Human exposure to mechanical vibration and shock,
ISO/TC 108/SC 5, Condition monitoring and diagnostics of machines,
ISO/TC 108/SC 6, Vibration and shock generating systems,
and
IEC/TC 29, Electroacoustics

P. D. Schomer, Chair, U.S. Technical Advisory Group (TAG) for ISO/TC 43, Acoustics and ISO/TC 43/SC 1 Noise
2117 Robert Drive, Champaign, IL 61821

D. J. Evans, Chair of the U.S. Technical Advisory Group (TAG) for ISO/TC 108, Mechanical Vibration and Shock
and Chair of the U.S. Technical Advisory Group (TAG) for ISO/TC 108/SC 3 Use and calibration of vibration and shock
measuring devices
National Institute of Standards and Technology (NIST), 100 Bureau Drive, Stop 8220, Gaithersburg, MD 20899

A. F. Kilcullen, Co-Chair of the U.S. Technical Advisory Group (TAG) for ISO/TC 108/SC 2, Measurement and evaluation
of mechanical vibration and shock as applied to machines, vehicles and structures
734 Walden Road, Hedgesville, WV 25427

R. Taddeo, Co-Chair of the U.S. Technical Advisory Group (TAG) for ISO/TC 108/SC 2, Measurment and evaluation
of mechanical vibration and shock as applied to machines, vehicles and structures
1333 Isaac Hull Avenue, SE, Washington Navy Yard, Washington DC 20376

D. D. Reynolds, Chair, U.S. Technical Advisory Group (TAG) for ISO/TC 108/SC 4, Human exposure to mechanical
vibration and shock
3939 Briar Crest Court, Las Vegas, NV 89120

D. J. Vendittis, Chair of the U.S. Technical Advisory Group (TAG) for ISO/TC 108/SC 5, Condition monitoring
and diagnostics of machines
701 Northeast Harbour Terrace, Boca Raton, FL 33431

R. Taddeo, Vice Chair of the U.S. Technical Advisory Group (TAG) for ISO/TC 108/SC 5, Condition monitoring
and diagnostics of machines
1333 Isaac Hull Avenue, SE, Washington Navy Yard, Washington DC 20376

G. Booth, Chair of the U.S. Technical Advisory Group (TAG) for ISO/TC 108/SC 6, Vibration and shock generating systems
44 Bristol Street, Branford, CT 06405-4842

V. Nedzelnitsky, U.S. Technical Advisor (TA) for IEC/TC 29, Electroacoustics
National Institute of Standards and Technology (NIST), Sound Building, Room A147, 100 Bureau Drive, Stop 8221,
Gaithersburg, MD 20899-8221

The reports of the Chairs of these TAGs will not be presented at any other S Committee meeting.

The meeting of the Standards Committee Plenary Group will precede the meetings of the Accredited Standards Committees S1, S2, S3, and S12, which are scheduled to take place in the following sequence:

ASC S1 Acoustics	6 June 2006	1:45 p.m. to 3:15 p.m.
ASC S12 Noise	6 June 2006	3:30 p.m. to 5:00 p.m.
ASC S2 Mechanical Vibration and Shock	7 June 2006	8:30 a.m. to 10:00 a.m.
ASC S3 Bioacoustics	7 June 2006	10:30 a.m. to 12:00 noon

Discussion at the Standards Committee Plenary Group meeting will consist of national items relevant to all S Committees.

The U.S. Technical Advisory Group (TAG) Chairs for the various international Technical Committees and Subcommittees under ISO and IEC, which are parallel to S1, S2, S3, and S12 are as follows:

U.S. TAG Chair/Vice Chair	TC or SC	U.S. Parallel Committee
ISO		
P. D. Schomer, Chair	**ISO/TC 43** Acoustics	S1 and S3
P. D. Schomer, Chair	**ISO/TC 43/SC1** Noise	S12
D. J. Evans, Chair	**ISO/TC 108** Mechanical Vibration and Shock	S2
A. F. Kilcullen, Co-Chair R. Taddeo, Co-Chair	**ISO/TC 108/SC2** Measurement and Evaluation of Mechnical Vibration and Shock as Applied to Machines, Vehicles and Structures	S2
D. J. Evans, Chair	**ISO/TC 108/SC3** Use and Calibration of Vibration and Shock Measuring Instruments	S2
D. D. Reynolds, Chair	**ISO/TC 108/SC4** Human Exposure to Mechanical Vibration and Shock	S3
D. J. Vendittis, Chair R. F. Taddeo, Vice Chair	**ISO/TC 108/SC5** Condition Monitoring and Diagnostics of Machines	S2
G. Booth, Chair	**ISO/TC 108/SC6** Vibration and Shock Generating Systems	S2
IEC		
V. Nedzelnitsky, U.S. TA	**IEC/TC 29** Electroacoustics	S1 and S3

TUESDAY AFTERNOON, 6 JUNE 2006　　　　　　　　　　　　　　　ROOM 555AB, 1:00 TO 4:45 P.M.

Session 2pAAa

Architectural Acoustics, Musical Acoustics, and Engineering Acoustics: Surround Sound Essentials II

Alexander U. Case, Cochair
Fermata Audio + Acoustics, P.O. Box 1161, Portsmouth, NH 03802

K. Anthony Hoover, Cochair
Cavanaugh Tocci Associates, 327F Boston Post Rd., Sudbury, MA 01776

Invited Papers

1:00

2pAAa1. Mastering music in a 5.1 world. Adam Ayan (Gateway Mastering Studios, 428 Cumberland Ave., Portland, ME 04101, adam@adamayan.com)

The evolution of 5.1 surround sound in the world of popular music has changed the way artists, producers, and engineers conceptualize their recordings. Along with the artistic benefits of 5.1 surround production come the challenges of working in this relatively new format. Mastering engineers are responsible for making the final sonic enhancements to a recording, musically maximizing the recording, and making it compatible and competitive in the marketplace. 5.1 surround mastering shares many of the same techniques and philosophies as stereo mastering, but also presents a new set of concerns for the mastering engineer. This talk focuses on the specific production and engineering concerns that are unique to the world of 5.1 surround production, specifically as they relate to mixing and mastering techniques.

2pAAa2. The real reasons you should invest in a surround-sound system. Barbara G. Shinn-Cunningham (Boston Univ. Hearing Res. Ctr., 677 Beacon St., Boston, MA 02215)

Most consumers of surround-sound systems understand that increasing the number and locations of speakers provides better control of perceived source location. The first goal of this talk is to explore why surround-sound systems generally work well in a variety of common settings, even though the acoustic effects of the rooms in which they are used can differ dramatically. What acoustic and psychoacoustic principles come into play? A second goal is to discuss the different perceptual benefits provided by a typical surround-sound system. Specifically, a good system not only allows listeners to hear sources at different directions, but also improves a listener's ability to understand simultaneous sources and monitor a complex auditory scene. Finally, we will discuss why it is important to realize that surround-sound systems are entertainment systems, not simulators. If asked, most consumers would probably argue that surround sound is good because it makes it possible to recreate a realistic auditory experience. In fact, the aim of many recordings and games that use surround-sound technology is not realism, but vicarious thrill and vivid experience. In entertainment, bigger (and louder and more extreme) is better than everyday, ordinary sensations. [Work supported by NIDCD.]

2pAAa3. The influence of changes in diffusive acoustical treatment on spatial imagery associated with multichannel sound reproduction. William L. Martens (Schulich School of Music, McGill Univ., Montreal, QC, Canada)

Listening experience in the newly constructed, largely diffuse small room at Blackbird Studios has led to the design of a new research space for the Sound Recording program in McGill University's Schulich School of Music. The space was created to allow for experimental investigation of the influence of changes in diffusive acoustical treatment on variation in auditory imagery associated with multichannel sound selections while holding listener location and orientation constant relative to a fixed five-channel loudspeaker array. Preliminary tests suggest that a number of benefits result when low-amplitude indirect sound is provided by diffusive room acoustic treatment within the first 30 ms of the arrival of the loudspeaker signals. Both for conventional multitrack mixes, and for recordings made using multichannel microphone arrays, these benefits include greater Listener EnVelopment (LEV), improved segregation of simultaneously sounding musical sources, and more solid phantom images of virtual sources, especially those located to the listener's extreme sides between front and rear loudspeakers (between 30 and 110 deg of azimuth to the left and right). This paper will summarize those results, and describe the specially designed system of variable acoustic treatment that enables such direct experimental investigation. [Work supported by Canada Foundation for Innovation.]

2pAAa4. Low-frequency control options in surround-sound critical listening rooms. Peter D'Antonio (RPG Diffusor Systems, Inc., 651-C Commerce Dr., Upper Marlboro, MD 20774)

Acoustically small rooms, which are typically used for critical listening, be it mono or surround, can potentially overwhelm the spatial and spectral textures that contribute to an enjoyable listening experience. The options to control potential modal and speaker boundary interference in rooms include optimal dimensional room ratios, optimal speaker/listener placement, optimal multiple sub-placement and dedicated low-frequency absorbers, including quarter-wave traps, anechoic wedges, Helmholtz and membrane resonators, and a relatively new and potentially significant addition, the metal plate resonator, which absorbs efficiently down to 50 Hz with a thickness of only 4 in. The design theory, proof-of-performance evaluation testing methods, and optimal placement guidelines for all of these tools will be reviewed.

2pAAa5. From mono to surround: A review of critical listening room design and a new immersive surround design proposal. Peter D'Antonio (RPG Diffusor Systems, Inc., 651-C Commerce Dr., Upper Marlboro, MD 20774)

As our critical listening rooms have evolved from mono to stereo to an ill-fated attempt at quad to full surround, many approaches have been taken in the design of these spaces. This presentation will review significant milestones in this evolution including the early compression ceilings, reflective front-absorptive rear, absorptive front-absorptive rear nonenvironment, absorptive front-diffusive rear, spatio-temporal reflection-free zone front-diffusive rear, and reflection-rich zone in front-diffusive rear. Recently there has been a renaissance in the evaluation of the importance of specular and diffuse reflections in auditory perception. The presentation will conclude with a proposal for a new immersive surround environment, which can be described as an ambient anechoic or ambechoic environment. Fully enveloping diffuse room reflections are 30 dB below the direct sound, as in an anechoic environment, yet the room feels ambient, provides precise localization, and offers a creative work environment for recording and monitoring. An example of a proof-of-concept room will be illustrated and described.

2pAAa6. An innovative professional music recording/monitoring space. George Massenburg (2805 Bransford Ave., Nashville, TN 37204-3101)

A new kind of professional recording/monitoring space has been designed and constructed as part of the Blackbird Studio complex in Nashville, Tennessee. The room will be introduced and the early subjective impressions will be discussed. The objective of this new room was threefold: first, to provide a surround mixing space with much-improved virtual source localization over a broader footprint in the space, second, to provide a highly diffuse space for the recording of live music performances by one or more vocalists and/or

musicians, and third, to investigate a largely diffuse monitoring environment as it might benefit the quality and/or efficiency of professional music mixing in different formats. This presentation will discuss some of the results and some of the surprise outcomes, including practitioner-reported improvements in quickly attaining comprehensive, surgical equalization of sources, possibly attributable to an environment with somewhat more equal reverb times across the frequency spectrum.

3:20

2pAAa7. Surround-sound wish list. Bob Ludwig (Gateway Mastering Studios, Inc., 428 Cumberland Ave., Portland, ME 04101, staff@gatewaymastering.com)

This presentation describes this engineer's involvement with surround sound since the days of vinyl LP, 8-track, and four-channel Quadrophonic sound. It includes my experiences as one of the few mastering engineers hired by Digital Theater Systems in 1997 to master and help establish their new 5.1 surround compact disc. In 2005, the author mastered 2 of the 5 Grammy-nominated ''Best Surround Sound discs''—the SA-CD Dire Straits 20th Anniversary edition of ''Brothers in Arms'' and the Foo Fighters ''In Your Honor'' DualDisc. This presentation also includes a short wish list of issues with popular music surround-sound mixes with some suggestions for improvement.

Contributed Papers

3:40

2pAAa8. Simultaneous equalization of a loudspeaker and a room: Insights from a one-dimensional model. Xi Chen and Timothy Leishman (Acoust. Res. Group, Dept. of Phys. and Astron., Brigham Young Univ., Provo, UT 84602)

In recent years, many audio practitioners have concluded that a room excited by a loudspeaker cannot be effectively equalized. They argue that the free-field response of the loudspeaker may be equalized, but that the room response itself must be ignored. This paper presents a one-dimensional model to illustrate many of the difficulties encountered in past equalization methods that have led to this way of thinking. It explores the spatial-spectral variations of the reproduced sound field that make the methods problematic. It then uses the model to explain the benefits of using total energy density as a basis for equalization instead of acoustic pressure. Both the loudspeaker and the enclosed field can then be simultaneously equalized for nearly optimal results. The method is also implemented effectively using adaptive filtering techniques.

3:55

2pAAa9. A hybrid approach to rendering multichannel audio, using wave-field synthesis and stereophonic techniques, with virtual microphone control. Daniel Valente and Jonas Braasch (Rensselaer Polytechnic Inst., Troy NY 12188)

In the evolution of multichannel audio presentation, a number of formats have emerged including 5.1 surround and wave-field synthesis (WFS). Due to the complexity and number of components required to realize an immersive WFS system, the use in a home theater environment is currently impractical. This study aims to investigate the perceptual advantages of integrating both systems into a hybrid array capable of providing immersive sound for a number of listeners, at a scale that is attainable for the consumer in terms of budget and complexity. In this approach the three front speakers (left, center, and right) are replaced with a linear line array allowing for a better lateral and depth imaging, while increasing the sweet spot. The rendering will be created using Virtual Microphone Control (ViMiC) for both WFS, simulating a line of omnidirectional microphones with ViMiC, and surround sound, which will allows dynamic source movement, speaker positioning, and scalability. The advantage of using this hybrid approach is the application of WFS to home theater systems without facing the complexity of a complete two-dimensional WFS array, while still allowing for an immersive sound presentation. A psychoacoustic evaluation will be presented, as well as measurements of the composed sound field.

4:10–4:45
Panel Discussion

Session 2pAAb

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

Robert C. Coffeen, Cochair
Univ. of Kansas, School of Architecture and Urban Design, Lawrence, KS 66045

Norman H. Philipp, Cochair
Yantis Acoustical Design, 720 Olive Way, Ste. 1400, Seattle, WA 98191-1878

Byron W. Harrison, Cochair
The Talaske Group, Inc., 105 N. Oak Park Dr., Oak Park, IL 60301

The Technical Committee on Architectural Acoustics of the Acoustical Society of America and the National Council of Acoustical Consultants are sponsoring this Student Design Competition that will be professionally judged at this meeting. The purpose of this design competition is to encourage students enrolled in architecture, architectural engineering, and other university curriculums that involve building design and/or acoustics to express their knowledge of architectural acoustics in the design of a facility in which acoustical considerations are of significant importance. Submissions will be poster presentations. This competition is open to undergraduate and graduate students from all nations.

The submitted designs will be displayed in this session and they will be judged by a panel of professional architects and acoustical consultants. An award of $1,250 will be made to the submitters of the entry judged ''First Honors.'' Up to four awards of $700 each will be made to submitters of entries judged ''Commendation.''

Session 2pAB

Animal Bioacoustics, Acoustical Oceanography, and Underwater Acoustics: Effects of Anthropogenic Sounds on Fishes

Arthur N. Popper, Cochair
Univ. of Maryland, Dept. of Biology, College Park, MD 20742

Richard R. Fay, Cochair
Loyola Univ., Parmly Hearing Inst., 6430 N. Kenmore, Chicago, IL 60626

Chair's Introduction—1:00

Invited Papers

1:10

2pAB1. Anthropogenic sound—Introduction and overview of the ambient and anthropogenic environment.　Arthur N. Popper (Dept. of Biol., Univ. of Maryland, College Park, MD 20742) and Richard R. Fay　(Loyola Univ., Chicago, IL 60626)

Interest in anthropogenic sound in the marine environment has been directed at concerns about the effects of sound on marine mammals. However, fish make up a far larger and more diverse portion of the oceans than do marine mammals, and they hold considerable economic importance internationally. This has led to a growing interest in the effect of anthropogenic sounds on fish in environments ranging from shallow waters near in-shore shipping lanes to the deep sea. Concerns for effects on fish are parallel to those for marine mammals, and include issues ranging from the death of individual animals to the potential for behavioral changes that could lead to impacts on the survival of populations or species. While the body of data on the effects of anthropogenic sounds on fish is still small, and much of the work is not in the peer-reviewed literature, the number of peer-reviewed studies is growing. The purpose of this special session is to provide an overview of what we currently know about the effects of anthropogenic sound on fish, help define the major outstanding questions on these effects, and to provide the basis for discussion of current and future research in order to help resolve these questions.

2pAB2. Attempting to define sound exposure metrics for fish. Mardi C. Hastings (Office of Naval Res., Life Sci. Res. Div., 875 N. Randolph St., Arlington, VA 22203-1995, mardi_hastings@onr.navy.mil)

One of the big issues in addressing the biological effects of sound on humans and animals is metrics. For fishes the issue is further complicated because their inner ear responds directly to acoustic particle motion and in some species, indirectly to pressure through multiple pathways. So, metrics based only on pressure can be misleading in developing sound exposure criteria. The decibel scale also confounds the problem because it is used to express not only levels of sound pressure, but also levels of sound power, intensity, and energy. Even for just sound pressure, decibels are used to express peak, peak-to-peak, and time-averaged values, which can hamper extrapolation of laboratory data to anthropogenic sources in the field. Add to that the changes in decibel reference values when going from air to water, more than 25 000 different fish species, and the fact that a significant amount of sound exposure research on fish has been done in small enclosures at low frequencies, and one quickly realizes the difficulties in developing meaningful relationships between sound exposure and observed effects. This presentation will focus on relevant sound exposure metrics, their practical definitions, and their application to quantifying the biological effects of sound on fish.

2:10

2pAB3. Effects of high-intensity sonar on fish. Michele B. Halvorsen, Lidia Eva Wysocki, and Arthur N. Popper (Dept. of Biol., Univ. of Maryland, College Park, MD 20742, apopper@umd.edu)

Studies investigated the effects of exposure to low-frequency active (LFA) sonar on rainbow trout, channel catfish, and pumpkin-seed sunfish using an element of the SURTASS LFA. Animals were exposed to sounds that had a received peak level of 193 dB re: 1 uPa (rms) (188.5 dB re: 1 uPa^2 s SEL). Measurements were made of hearing sensitivity using auditory brainstem response and of behavioral responses. Additional studies were done on pathology of the ear and other organ systems. Trout and catfish both showed temporary threshold shifts immediately after sound exposure, while sunfish showed no threshold shift. However, the amount of threshold shift and frequencies concerned varied. In addition, trout and catfish showed immediate behavioral responses to the sound onsets and in relative distribution during sound presentation. Details of the responses differed between species. There was no mortality in any species attributable to sound exposure during or after experiments. SEM examination of hair cell ciliary bundles of the inner ears did not show any damage. Similarly, there were no effects on the swim bladder or nonauditory tissues. Overall, the effects of LFA on fish depend upon species and appear to be temporary and relatively modest. [Work supported by Chief of Naval Operations.]

2:35

2pAB4. Effects of riverine seismic air-gun exposure on fish hearing. David A. Mann (Univ. of South Florida, College of Marine Sci., 140 7th Ave. S., St. Petersburg, FL 33701, dmann@marine.usf.edu), Peter Cott, Bruce Hanna (Dept. of Fisheries and Oceans, Yellowknife, NT, Canada X1A 1E2), Alex MacGillivray, Melanie Austin (JASCO Res. Ltd., Victoria BC, Canada V8Z 7X8), Michael Smith, and Arthur N. Popper (Univ. of Maryland, College Park, MD 20742)

Seismic airguns produce considerable amounts of acoustic energy that have the potential to affect marine life. This study investigated the effects of exposure to an airgun array in the Mackenzie River Delta on the hearing of three fish species: northern pike (*Esox lucius*), broad whitefish (*Coregonus nasus*), and lake chub (*Couesius plumbeus*). Fish were placed in cages in 1.9 m of water and exposed to 5 or 20 airgun shots, while controls were placed in the same cage but without airgun exposure. Hearing in both exposed and control fish were then tested using auditory evoked potentials (AEPs). Threshold shifts were found for exposed fish as compared to controls in the northern pike and lake chub, with recovery within 24 h of exposure, while there was no threshold shift in the broad whitefish. It is concluded that these three species are not likely to be substantially impacted by exposure to a 2D airgun array used in a river seismic survey. Care must be taken in extrapolation to other species in other environments and to fishes exposed to airguns in 3D tests or where the animals are exposed to a larger number of airgun shots over a longer period of time.

3:00−3:15   Break

3:15

2pAB5. Effects on fish of pile driving, wind turbines, and other sources. Anthony Hawkins (Loughine Ltd., Kincraig, Blairs, Aberdeen AB12 5YT, UK)

The sounds associated with construction work in the sea are reviewed. The installation of offshore oil and gas platforms, the mounting of offshore wind turbines on the seabed, the construction and renovation of harbors and piers, and the laying and trenching of pipelines and cables can be exceptionally noisy activities. The sounds associated with pile driving, in particular, have the potential to injure fish, induce hearing loss, and evoke avoidance reactions. The sound-pressure levels generated in water by these activities are generally measured, but many fish respond to particle motion rather than sound pressure. Measurements of particle motion and intensity will be necessary in the future if the full effects of these sounds are to be assessed.

3:40

2pAB6. Effects of background sound on fish. Michael E. Smith (Dept. of Biol., Western Kentucky Univ., Bowling Green, KY 42101, michael.smith1@wku.edu), Lidia E. Wysocki, and Arthur N. Popper (Univ. of Maryland, College Park, MD 20742)

There is increasing awareness that underwater anthropogenic sounds may be detrimental to marine organisms. While most of this work has focused on marine mammals, recent studies have shown that sounds higher than background can have detrimental effects on fish as well. This presentation is a review of the current data available on the effects of increased background ambient sound on fish. Loud continuous sounds can cause temporary hearing loss in fish, with the extent of hearing loss increasing with noise exposure duration and intensity, and recovery of hearing occurring within a few days or weeks. The extent of hearing loss is also species

specific—fish with more sensitive hearing thresholds are more prone to hearing loss, whereas nonsensitive fish are barely affected. The latter was shown in a long-term study on the effects of aquaculture noise in rainbow trout. A model termed the linear threshold shift hypothesis (LINTS) has been proposed to predict the extent of noise-induced hearing loss in fish, and this model should be beneficial in mitigating the effects of anthropogenic sound on fish. Other effects of noise include acoustical masking and physiological and behavioral stress responses, which also depend on species and noise type.

4:05

2pAB7. Noise exposure criteria for fish and turtles; lessons learned from criteria for marine mammals. Roger L. Gentry (ProScience Consulting, P.O. Box 177, Dickerson, MD 20842-0177, Roger.gentry@comcast.net)

A committee that is just finishing noise exposure criteria for marine mammals has learned the following lessons that may apply to the writing of similar criteria for fish and turtles. Writing criteria will require scientists from highly diverse backgrounds meeting multiple times over several years. Scientific discourse should prevail throughout; special interests will hinder the writing of science-based criteria. Because empirical data will undoubtedly be inadequate to the task, extrapolation procedures will be needed. Especially lacking will be animal data reporting metrics like SEL and particle motion. The wide diversity of sound types and of animal species will necessitate some grouping within each category. The most efficient format for the criteria will be a matrix of the two groupings. However, existing data will not support a matrix having too many cells. Some criteria will need to be expressed in more than one metric to account for all the relevant aspects of complex signatures. Frequency weighting functions will be needed. Criteria for injury will be easier to write than for behavioral responses due to data availability. Finally, several iterations of the criteria will be needed as science improves. The first version should be written to accommodate change.

4:30

2pAB8. Working group on "The effects of sound on fish and turtles": An update. Richard R. Fay (Parmly Hearing Inst., Loyola Univ. Chicago, IL 60626) and Arthur N. Popper (Univ. of Maryland, College Park, MD)

In order to develop acoustic exposure criteria for fish and marine turtles based on scientific evidence, an ASA Standards Working Group was formed with a membership that includes basic scientists expert on fish and turtle bioacoustics, physical acousticians with special expertise in marine bioacoustics, and several individuals with expertise on effects of anthropogenic sound in the marine environment. This presentation is a summary of the group's progress to date. One of the Working Group's goals is to publish several papers, one on what is known about damage risks due to exposure to underwater noise (e.g., from pile driving, seismic exploration, sonar systems), and one assessing the efficacy of various measures to mitigate adverse effects of noise exposure. The working group has tentatively decided that the fish potentially affected should be categorized with respect to their hearing sensitivity (potential for auditory system damage) and whether the species is known to detect sound pressure or acoustic particle motion. Over the next year to 18 months, the Working Group will be evaluating the known effects of sound sources on fish and turtles, identifying areas in which further research is needed, and beginning to develop noise exposure criteria for both turtles and fish.

4:55–5:25
Panel Discussion

TUESDAY AFTERNOON, 6 JUNE 2006   BALLROOM D, 1:30 TO 4:35 P.M.

Session 2pBB

Biomedical Ultrasound/Bioresponse to Vibration, Physical Acoustics, and Signal Processing in Acoustics: Sensing and Imaging Using Light and Sound

Ronald A. Roy, Cochair
Boston Univ., Dept. of Aerospace and Mechanical Engineering, 110 Cummington St., Boston MA 02215

Todd W. Murray, Cochair
Boston Univ., Dept of Aerospace and Mechanical Engineering, 110 Cummington St., Boston, MA 02215

Invited Papers

1:30

2pBB1. Nanoacoustics: Towards imaging nanostructures using picosecond ultrasonics. T. B. Norris (Ctr. for Ultrafast Optical Sci., Univ. of Michigan, Flint, MI 48502, tnorris@eecs.umich.edu)

Coherent acoustic phonon pulses with a peak frequency and bandwidth of the order of 100 GHz can be generated by ultrafast optical excitation of a thin metal film on a crystalline substrate. We demonstrate submicron resolution imaging of surface-grown nanostructures using these picosecond ultrasonic pulses. Ultrafast optical pulses from a Ti:sapphire oscillator are used to thermoelas-

tically excite a patterned 15-nm-thick Al film on a Si substrate. The spatio-temporal diffracted acoustic strain field is measured by time-delayed optical probe pulses on the opposite side of the substrate, and this strain field is used in a time-reversal algorithm to reconstruct the Al pattern. The image resolution is characterized using lithographically defined 1-micron-period Al structures on Si. Straightforward technical improvements should lead to a resolution of at least 45 nm, which will significantly extend the resolution of acoustic microscopy and provide a novel contrast mechanism for nanoscale imaging. To assist the development of this imaging method, we have also made experimental and theoretical studies of the behavior of these coherent acoustic phonon pulses after propagation through millimeter-scale distances in crystalline Si.

1:55

2pBB2. Applying optics in surface wave laser ultrasonic microscopy. Michael Somekh, Stephen Sharples, and Matthew Clark (Univ. of Nottingham, NG7 2RD, UK, mike.somekh@nottingham.ac.uk)

Laser ultrasonics has many advantages compared to contacting ultrasonics, especially when both optical generation and detection are used. In addition to the obvious advantages of remote inspection there are several other benefits. In this paper we concentrate on the flexibility that can be conferred by employing both optical technology and optical concepts in our laser ultrasonic systems. The papers shows how spatial light modulators (SLMs) can be used to control the optical beam profile used to generate the ultrasonic waves. Initially, SLMs were introduced to improve the signal-to-noise ratio of the system. This is achieved by exciting the tone burst with a grating structure so that large signals can be generated without ablation. The SLMs can also be used to produce optical profiles that generate self-focusing surface waves; this also increases the ultrasonic amplitude. Our recent work has used SLM generation to overcome aberrations or distortions in the ultrasonic field during propagation. To this end an adaptive system using integrated electronics to overcome the phase aberrations during propagation has been developed. We also show how the SLM can be used to perform acoustic wavelength spectroscopy, which has proved invaluable in the characterization of texture in industrially important materials.

2:20

2pBB3. Laser generation and detection of zero-group velocity Lamb mode resonance in thin plates. Claire Prada, Dominique Clorennec, Daniel Royer   (LOA, ESPCI, Universit Paris 7, CNRS UMR 7587, 10 rue Vauquelin, 75231 Paris, Cedex 05, France), Oluwaseyi Balogun, and Todd W. Murray   (Boston Univ., Boston, MA 02215)

Some Lamb modes exhibit a resonant behavior at frequencies where the group velocity vanishes while the phase velocity remains finite. Such a zero group velocity point exists in most isotropic materials for the first-order symmetric mode. Laser sources couple efficiently into this resonance and a sharp peak is observed with source and receiver on epicenter. Moving the detection point away from the source allows us to establish the dispersion curves of the plate and observe the negative phase velocity. We used a modulated diode laser source and a Michelson interferometer to observe the resonance at high frequency. Measurements were made on different material of thickness down to $4\,\mu$m. The resonance peak is sensitive to the thickness and mechanical properties of plates and may be suitable for the measurement and mapping of nanoscale thickness variations. We also observed this behavior with a 20-ns pulsed YAG laser source and a heterodyne interferometer. The low attenuation in duralumin allows the excitation of a very sharp resonance in a single shot. Thickness variations as small as $0.1\,\mu$m have been measured in a 0.50-mm-thick duralumin plate. Furthermore, the time decay of the signal provides an estimation of the local attenuation.

2:45

2pBB4. Fabry Perot polymer film sensors for ultrasound field characterization and biomedical photoacoustic imaging. P. C. Beard   (Dept. of Medical Phys. and Bioengineer., Univ. College London, Gower St., London WC1E 6BT, UK)

A contact ultrasound sensing technique based upon the detection of acoustically induced changes in the optical thickness of a Fabry Perot polymer film sensing interferometer has been developed for ultrasound measurement and imaging applications in medical and industrial fields. The technique provides a broadband ($>$50 MHz) response and excellent detection sensitivities ($<$1 kPa), comparable to those of piezoelectric PVDF transducers. A key feature is that the sensing geometry is defined by the area of the polymer sensing film that is optically addressed. As a consequence, very small element sizes (in principle down to the optical diffraction limit of a few μm) can be obtained without compromising detection sensitivity—a useful advantage over piezoelectric transducers. Additionally, by spatially sampling over a relatively large aperture, a high-density ultrasound receive array can readily be configured. These attributes lend the technique to a variety of applications such as mapping the output of broadband ultrasound sources, transmission ultrasound imaging, and biomedical photoacoustic imaging. The latter is particularly promising, offering the prospect of high-resolution imaging of optically absorbing soft tissue structures such as blood vessels for the study of tumors and other tissue abnormalities characterized by changes in the structure and oxygen status of the microvasculature.

3:10−3:25 Break

3:25

2pBB5. Playing with speckle in acousto-optic imaging. Albert-Claude Boccara, Benoit Forget, Francois Ramaz, Emmanuel Bossy (CNRS and Laboratoire Optique, ESPCI, 10 rue Vauquelin, F75005 Paris, France), Michel Gross   (ENS Laboratoire, F75005 Paris, France), Pedro Santos, Florence Jean, and Michael Atlan   (CNRS and Laboratoire Optique, F75005 Paris, France)

Using light to image in-depth structures and to quantify them in terms of optical (absorption, scattering) properties is a goal for physicists. We will recall the approaches which overcome the strong loss of the photons directional memory in scattering media and their limitations in terms of depth and resolution. Then we will consider hybrid methods which couple acoustics and optics to reach the optical contrast with the acoustic resolution: opto-acoustics, where a locally absorbed laser pulse generates an acoustic signal whose position and depth can be localized by an array of piezoelectric detectors, and acousto-optics (A.O.), where an ultrasonic beam

2p TUE. PM

modulates the speckle. We will focus on the principle of A.O. signal generation for tagging spatially selected photons and describe various approaches based on numerical heterodyne holography or heterodyne holography using a photo-refractive beam splitter. To get enough signals through thick tissues (e.g., breast) in a short time we use these techniques at the shot noise limit. Finally we show images revealing optical contrasts in artificial structures or in tissues.

Contributed Papers

3:50

2pBB6. Characterization of nano-electro-mechanical systems (NEMS) using photoacoustic microscopy. A. SampathKumar, K. L. Ekinci, and T. W. Murray (Dept. of Aerosp. and Mech. Eng., Boston Univ., 110 Cummington St., Boston, MA 02215, twmurray@bu.edu)

A photoacoustic microscopy system has been developed for the non-contact characterization of materials at the micro- and nanometer scale. The system uses an electroabsoption modulated laser source for material excitation and a path-stabilized interferometer to detect the resulting displacements. Using this system, we describe the optical excitation and detection of high-frequency nanoelectromechanical systems (NEMS) at ambient temperature and pressure. Using a tightly focused modulated laser source, we have actuated the out-of-plane flexural resonances of bi-layered doubly clamped nanomechanical beams. The optically detected displacement profiles in these beams are consistent with a model where the absorbed laser power results in a local temperature rise and a subsequent thermally induced bending moment. The excitation technique allows probing and actuation of NEMS with excellent spatial and temporal resolution. From a device perspective, the technique offers immense frequency tunability and may enable future NEMS that can be remotely accessed thus obviating the need for electronic coupling. [The authors gratefully acknowledge funding from National Science Foundation under Grants ECS-0304446 and ECS-0210752.]

4:05

2pBB7. Focusing by use of time reversal of photoacoustic waves. Emmanuel Bossy (Laboratoire d'Optique Physique, ESPCI-CNRS UPR5, 10 rue Vauquelin 75005 Paris, France), Gabriel Montaldo, Jean-Francois Aubry, Michael Tanter, Mathias Fink (ESPCI-CNRS Universite Paris 7, 75005 Paris, France), and Claude Boccara (ESPCI-CNRS UPR5, 75005 Paris, France)

Time reversal of ultrasonic waves allows focusing of ultrasound through complex media, such as highly aberrating media. On one hand, focusing with time reversal can be performed by backpropagating detected ultrasonic waves generated by a localized ultrasound source. This source classically consists of high acoustic contrasts echoing ultrasonic waves generated by an incident ultrasonic field, or directly by pointlike transduc-ers inserted in the medium. On the other hand, the photoacoustic effect consists of the emission of ultrasound caused by the absorption of light by an optical absorber. In particular, this effect can be used to ultrasonically image a distribution of optical absorbers simply by recording photoacoustic signals on an imaging transducers array. In this work, we perform time reversal of ultrasound generated by a pointlike photoacoustic source in order to automatically focus ultrasound back to an optical contrast. A Q-switched pulsed Nd:YAG laser was used to illuminate tissue phantom with nanosecond laser pulses. Conventional ultrasonic transducers arrays (in the MHz frequency range) and a time-reversal electronics were used to detect, record, and time reverse photoacoustic signals. We show that this allows focusing of ultrasound through highly aberrating medium, and perform acoustical imaging with good resolution within the isoplanetic region surrounding the optical contrast.

4:20

2pBB8. Laser pulse impact characterization using ultrasonic transducers. Subash B. Jayaraman, Mark A. Bohenick, Vladimir Semak, Bernhard R. Tittmann (The Penn State Univ., 212 Earth & Eng. Sci. Bldg., University Park, PA 16802), Jill R. Welsch, and Nicholas C. Nicholas (The Penn State Univ., University Park, PA 16802)

Experiments were conducted with an excimer laser (wavelength 248 nm) to cause damage to gelatinous materials, which have optical properties similar to skin tissue. The laser pulses, when they impact the specimen, generate an elastic wave that propagates through the material. A 150-MHz broadband ultrasonic transducer was placed underneath the specimen to observe the material response and the pressure generated by the laser pulse. Different incident laser energies and laser beam spot sizes were utilized to obtain a range of input parameters, and the corresponding transducer responses were recorded. Fourier analysis of the signals was performed to identify the frequency response from the laser pulse. Additional tests were carried out to observe the effects due to a train of laser pulses. Earlier tests were performed with 200-kHz hydrophone-calibrated transducer but due to the shortness of the laser pulses ($\sim$20 ns), a higher frequency transducer was used to accurately characterize the laser impact. The initial results indicate that there is a dominant frequency response around the 25–30-MHz regime.

Session 2pEA

Engineering Acoustics and Physical Acoustics: Joe Blue Memorial Session II: Transduction, Linear and Nonlinear

Thomas G. Muir, Cochair

Univ. of Mississippi, National Ctr. for Physical Acoustics, Coliseum Drive, University, MS 38677

Joseph F. Zalesak, Cochair

2359 US Hwy. 70 SE, #325, Hickory, NC 28602-8300

Chair's Introduction—1:00

Invited Papers

1:05

2pEA1. Dr. Blue at the Underwater Sound Reference Division. Joseph F. Zalesak (2359 US Hwy. 70, SE #325, Hickory, NC 28602-8300)

At the Underwater Sound Reference Division Dr. Blue had many varied interests including acoustic metrology, acoustic materials research, enginering acoustics, underwater transducers, and bioacoustics. He performed the first thorough study on the reproducibility of calibrations of standard hydrophones. He supported the development of traveling-wave tubes and large pressure vessels for calibrating Navy sonar transducers and for the measurement of acoustic materials. The Acoustic Pressue Tank Facility is a 30 000-gal acoustic test vessel ideally suited for acoustic measurements at hydrostatic pressures up to 2700 psi and at frequencies as low as 1.0 kHz and can accommodate acoustic devices up to 6 ft in diameter. Dr. Blue supported research for improved measurement techniques for determining the properties of acoustic materials. Dr. Blue actively supported the development of new elastomeric materials used in the construction of underwater transducers. The Sonar Transducer Reliability Improvement Program was an extinsive program to properly specify a sonar transducer for Navy applications. He first suggested that coupler reciprocity might be used to perform ultra-precise calibrations of hydrophones. He also supported research in manatee vocalizations and dolphin echolocation clicks. Low-noise hydrophones with noise levels better than sea-state zero were developed for general use.

1:25

2pEA2. A perspective of Navy transducer calibration standards. Kirk Jenne (Office of Naval Res., Code 03D&I, 875 N. Randolph St., Arlington, VA 22203-1995)

Research and development efforts to establish underwater electroacoustic transducer standards for use in a variety of environmental conditions, performed concurrently with research on calibration methodologies, began nearly 60 years ago. This research and development effort successfully generated a series of standards now made available under the auspices of the U.S. Navy's Underwater Sound Reference Division in Newport, RI (a.k.a. USRD). Because of consistent and insightful investment of resources made over the years by Navy technical agents and advocates like Dr. Joseph Blue, scientists and engineers from government (U.S. and select foreign), academia, and industry can use Navy transducer calibration standards to perform accurate measurements of acoustic energy underwater. Definition and characteristics of a standard are considered. Performance characteristics of transducer standards currently used, along with some discussion of applications, are presented.

1:45

2pEA3. Development of 1-3 piezocomposites for U. S. Navy applications. Thomas R. Howarth and Kim C. Benjamin (Naval Sea Systems Command Div. Newport, 1176 Howell St., Newport, RI 02841, Howarthtr@npt.nuwc.navy.mil)

Efforts to develop 1-3 piezocomposite materials for U.S. Navy applications began in the early 1990s at the former Naval Research Laboratory's Underwater Sound Reference Detachment in Orlando, FL. These initial investigations concentrated on evaluating the materials properties and parameters from a number of different contractors and manufacturing techniques. This was followed by efforts to initially fabricate receiving devices to replace traditional piezoceramic sensor prototypes, followed by development of devices with both receiving and transmitting capabilities. The presentation will discuss the initial NRL-USRD efforts and follow the development of 1-3 piezocomposite materials from these humble beginnings to devices that are currently being used and further developed by the U.S. Navy. [Work supported by the NRL, ONR, DARPA, and NAVSEA.]

2p TUE. PM

2:05

2pEA4. Use of complex source points to simplify numerical Gaussian beam synthesis. Stephen Forsythe (Naval Undersea Warfare Ctr., 1176 Howell Ave., Newport, RI 02841)

It is often desirable to generate the acoustic field due to a so-called Gaussian beam. One way to do this is to use the free-space Green's function for the acoustic field and to sum small area sources over a circular plate with the appropriate shading for the desired Gaussian beam. For very high frequencies and narrow beams, the computation time to give an accurate sum can be large when calculating the sum for many points in the acoustic field. An alternate approach comes from the use of a single point source with complex coordinates $R = (Xr + i*Xi, Yr + i*Yi, Zr + i*Zi)$. When this complex source point is used in the free-space Green's function, the formal expressions for pressure and particle velocity can be used if careful attention is paid to the interpretation of the complex distance, r, that arises in the $\exp(ikr)/r$ term. The singularity is no longer a single point in the case of a complex source, but a circular disk. The far field of a complex source point is a good approximation to a Gaussian beam. Several computational uses of the technique will be demonstrated. Extension to the shear wave Green's function will be explored. [Work supported by ONR.]

2:25–2:45   Break

2:45–2:50   Introduction

2:50

2pEA5. Joe Blue and nonlinear acoustic transduction: Lessons learned and forgotten, some even ignored; and practical issues to revisit, anew. Thomas Muir (Natl. Ctr. for Physical Acoust., Univ. of Mississippi, One Coliseum Dr., University, MS, 38677)

The 20th century was ripe with major advances in the understanding of nonlinear acoustics and its practical applications, in the American as well as the European and Soviet schools of thought. Some of us were privileged to study this art and contribute to its science, guided by the world's finest educators and mentors. The results originated in physics and were applied to aircraft acoustics and underwater sound, and later to biomedical acoustics, and other arenas, with great success. Joe Blue was a contributor and long-time advisor in many of these endeavors. Some early nonlinear acoustics topics will be reviewed, to refresh memories and introduce newcomers to the field. This includes parametric transmitters, the effects of shock formation and harmonic generation, parametric receivers, various theoretical models, classic experiments in air and water, as well as practical applications. In the early days, ideas for new nonlinear acoustic applications were often met with the comment, "That's crazy." Perhaps many of these ideas were not crazy enough. At the present time, there appears to be some lack of boldness and innovation, coupled with abundant mistakes, in what is proposed and reported. Some comments on the state of the art are offered. [Work supported, throughout the years of the mid to late 20th century, by the U.S. Navy.]

3:10

2pEA6. Investigation of media with large, negative parameters of nonlinearity and their application to the enhancement of a compact, omnidirectional, parametric source. Peter H. Rogers, Etienne Dufour, and David H. Trivett (Woodruff School of Mech. Eng., Georgia Inst. of Technol., Atlanta, GA 30332)

This paper reports on two media whose acoustic properties are highly pressure dependent and, thus, have large nonlinear parameters ($B/A \sim -3000$ to -5000). One is Xanthan gelatin filled with, 10%, Expancel microspheres. The other is a polyurethane rubber, Polytek 74, with 20% Expancel microspheres. The microspheres have a high bulk modulus at ambient pressure due to the hoop stiffness of the spherical shell. As pressure is increased some of the microspheres buckle, leading to significantly decreased bulk modulus. The decreasing bulk modulus with increasing hydrostatic pressure (negative dc/dp) results in both negative B/A and low sound speed (150 m/s at the minimum), which further enhances nonlinear distortion. These media are used to enhance the efficiency of an underwater parametric source. Experiments on a spherical source with a thin layer (2 cm thick) of the Xanthan gelatin material resulted in a generated difference frequency source level 16 dB higher at 300 Hz, than would be obtained driving the transducer directly at the difference frequency. Experiments on the same source with a thin layer of the microvoided, polyurethane rubber are in progress. This material has a smaller coefficient of nonlinearity but operates over a broader range of hydrostatic pressures.

3:30

2pEA7. Nonlinear acoustic detection of buried landmines: A tribute to Joe Blue. Murray S. Korman (Phys. Dept., U. S. Naval Acad., Annapolis, MD 21402)

As a graduate student at Brown University ($\sim$1978), this researcher was very fortunate to have met Joe Blue at Acoustical Society meetings. As I had little knowledge of the members in the society and an extremely limited grasp of the practical technology behind nonlinear acoustics, Joe was extremely friendly and took the time to answer a long list of questions about nonlinear underwater transducer designs with applications to the parametric array [Muir, T.G. & Blue, J.E. "Experiments on the acoustic modulation of large-amplitude waves," J. Acoust. Soc. Am. **46**, 227–232 (1969)]. This influenced this researcher's Ph.D. thesis involving experiments on the scattering of sound by sound in the presence of turbulence. Recently, experiments on the nonlinear acoustic detection of buried landmines [Korman, M.S. & Sabatier, J.M., "Nonlinear acoustic techniques for landmine detection," J. Acoust. Soc. Am. **116**, 3354–3369 (2004)] has brought some attention to understanding the difference between classical and nonclassical nonlinearity. As a tribute to Joe Blue, the latest results will be presented involving the connections to mesoscopic nonlinear elasticity and slow dynamics in nonlinear buried landmine detection. Here, tuning curve behavior of the soil-surface vibration over a landmine exhibits a linear decrease in resonant frequency with increasing amplitude. Two-tone test excitation reveals a rich spectrum of combination frequency components useful in detection schemes.

2pEA8. The application of acoustic standards to sound sources and the subsequent levels in the ocean. William M. Carey (Aero and Mech. Eng., College of Eng., Boston Univ., Boston, MA 02215)

The American National Standards on Acoustics are applied to common sound sources used in both acoustics research and in naval sonar systems operation. Metrics are quantified for both continuous and transient sources of sound. The standard definitions are reviewed with theoretical sound source models and applied to examples of energy sources of sound such as transients from a small omni explosive, an air gun, a light bulb, and a dolphin. A qualitative model of a typical surface ship sonar system is discussed and active sonar transmissions are analyzed with the requisite quantitative metrics required to characterize these emissions. The relative role of peak pressure, time spread, intensity, and energy flux is discussed for deep- and shallow-water environments. These results should be useful in environmental assessments, biological experiments, and in the system design.

Contributed Papers

4:10

2pEA9. Calibrating parametric sonars for backscattering applications. Kenneth G. Foote (Woods Hole Oceanogr. Inst., 98 Water St., Woods Hole, MA 02543)

The status of parametric sonar calibration is reviewed. For backscattering applications, the standard-target method, which is widely used in calibrating ultrasonic scientific echo sounders and multibeam sonars that provide the water-column signal, can be adapted. Two problems generic to parametric sonars are addressed. (1) The virtual nature of the array may require controlled measurements at a number of ranges, spanning the near field in the general case. (2) Calibration at rather low difference frequencies requires a suitably powerful target. Both measurement protocols and target design are considered for a particular parametric sonar based on primary frequencies in the band 15–21 kHz, with difference frequencies in the band 0.5–6 kHz and sum frequencies in the band 30–42 kHz. Solid elastic spheres are specified for both frequency bands. For the lower band, this is an aluminum sphere of diameter 280 mm, with immersed weight of 190 N.

4:25

2pEA10. Evaluating the significance of interface nonlinearity for shallow-water parametric array sonar. Roger Waxler and Thomas G. Muir (NCPA, University, MS 38577)

Difference frequency generation in shallow water is considered for the case in which wave steepening in the primary beam is not significant. A general formula for the effective difference frequency source strength is presented. The Pekeris shallow water model, in which sound speeds and densities in both the water column and sediment are assumed to be constant and both interfaces are assumed to be planar, is considered. In this model, the propagating difference frequency pressure field is computed. Contributions from the interface nonlinearity are compared to those from nonlinearity in the water column.

4:40

2pEA11. High frequency acoustic wave propagation. Mohsen Badiey (Ocean Acoust. Lab., Univ. of Delaware, Newark, DE 19716)

A program studying high frequency acoustics was established in the mid-1990's by the Office of Naval Research that is still going on and has evolved into a substantial basic and applied research initiative to study many areas, including underwater communication. This abstract explains the genesis of that program, and Joe Blue's involvement in making it happen. During the course of the initial experiments under that program, a series of issues were raised dealing with the effects of acoustic transmissions on marine mammals. Again, Joe was a key person in resolving those issues, which could have brought the program to a halt. His insight into the technical problems, as well as, his calm nature helped to bring an excellent new program in acoustic research into being.

4:55

2pEA12. Ship strike acoustics: A paradox and parametric solution. Edmund Gerstein, Joseph Blue (Leviathan Legacy Inc., 1318 SW 14th St., Boca Raton, FL 33486), and Steve Forsythe (Naval Undersea Warfare Ctr., Newport, RI 02841)

Marine mammals are vulnerable to ship collisions when are near the surface. Here, acoustical laws of reflection and propagation can limit their ability to hear and locate the noise from approaching vessels. Defining the physics of near-surface acoustical propagation as it relates to ship noise and hearing is central to understanding and mitigating ship strikes. Field data from controlled ship passages through vertical hydrophone arrays demonstrate a confluence of acoustical factors that poses detection challenges including (i) downward refraction; (ii) spreading loss; (iii) Lloyd's mirror effect; (iv) acoustical shadowing; and (v) masking of approaching ship noise by ambient noise and distant ships. A highly directional, dual-frequency parametric sonar has been developed to mitigate these challenges and alert marine mammals of approaching vessels. The system projector is a planar array, comprised of 45 elements, band centered to transmit a high carrier frequency along with a lower sideband signal. The nonlinearity of water is used to demodulate the mixed high-frequency carrier into a lower-frequency waveform audible to both manatees and whales. The bow-mounted arrays project a narrow beam directly ahead of vessels, and fill in acoustical shadows in an effort to alert marine mammals of the approaching danger. [Research supported by DOD Navy Legacy Program.]

Session 2pMU

Musical Acoustics and Structural Acoustics and Vibration: Finite Element and Finite Difference Methods in Musical Acoustics I

Rolf Bader, Cochair
Univ. of Hamburg, Inst. of Musicology, Neue Rabenstr. 13, 20354 Hamburg, Germany

Uwe J. Hansen, Cochair
Indiana State Univ., Dept. of Physics, Terre Haute, IN 47809

Chair's Introduction—1:25

Invited Papers

1:30

2pMU1. Mode studies in a triangular bell plate. Uwe J. Hansen, James Garner (Dept. of Phys., Indiana State Univ., Terre Haute, IN 47809), Thomas D. Rossing (Northern Illinois Univ., DeKalb, IL 60115), and Suzanne Hogg (Univ. of Technol., Sydney, Australia)

A triangular bell plate serves as a poor man's hand bell. A clapper affixed to the handle strikes the plate, resulting in a sound somewhat resembling the tone of a hand bell. A similar arrangement has been used with slotted Eames tubes. This paper will present a comparison of modes of vibration of a triangular bell plate as observed with holographic interferometry, modal analysis, and as calculated using finite element methods. The holographic interferometry studies were done in the Acoustics laboratory of NIU, the modal analysis work was carried out at ISU, and the FE calculations were done at the UT in Sydney. The similarity of mode shapes and mode frequencies lends credence to the validity of the techniques.

2:00

2pMU2. Finite-element calculation of a bass drum. Rolf Bader (Inst. of Musicology, Univ. of Hamburg, Neue Rabenstrasse 13, 20354 Hamburg, Germany)

The bass drum of a drum kit used with rock or jazz music is modeled as a 3D geometry in a transient finite-element formulation. The geometry was taken from a Gretch drum set called ''Vinnie Colaiuta'' with no cushion damping within the drum or at the resonance membrane. The 12 cross layers or wood used for the drum body was assumed to result in an isotropical Young's modulus all over the wood. The drum was modeled with air enclosed, with the beating and resonating membranes being coupled to the air and to the wooden rim of the drum body. Also, as a starting point of the transient calculation, the strain of the wooden body caused by the tension of the membranes was taken into consideration. The transient model started with a beating on the front membrane and the radiation of the system was integrated over the radiating area. The results are compared to recordings of a real bass drum of this Gretch drum set.

2:30

2pMU3. Finite-element transient calculation of a bell struck by a clapper. Bjrn-Andre Lau, Peter Wriggers (Institut fr Baumechanik und Numerische Mechanik, Univ. of Hannover, Appelstrae 9a, 30167 Hannover, Germany), Albrecht Schneider, and Rolf Bader (Univ. of Hamburg, 20354 Hamburg, Germany)

A finite-element calculation of a bell struck by the bell clapper was performed as a transient contact problem. The geometry of the bell was taken from a real bell which was carved especially in behalf of this research. The eigenmodes and eigenfrequencies of the bell were calculated in advance, being in good agreement with the measured frequencies. The clapper/bell contact time and impulse shape show the expected behavior. The impulse shape is a Gaussian impulse with a tendency of a sharper attack slope and a softer decay slope. The periodical shape of the spectrum of this impulse caused by its finite width with zero values for some frequencies is shown. Nevertheless, this spectrum also drives the frequencies of the bell within the low-amplitude region of the driving impulse with expected and measured strength. The reason for this mode energy transfer is discussed. Furthermore, the sound of the bell is calculated as radiation from its surface and analyzed with signal-processing tools like wavelet transforms or autoregressive models.

3:15

2pMU4. Finite-difference modeling of the piano. N. Giordano (Dept. of Phys., Purdue Univ., 525 Northwestern Ave, West Lafayette, IN 47907, giordano@purdue.edu)

The construction of a complete computational model of the piano is described. This model treats the three major subsystems of the instrument: the hammers and strings, the soundboard, and the air (the sound radiation). Hammer nonlinearities and hysteresis are included, along with realistic stiff strings with frequency-dependent damping, following the work of Chaigne and Askenfelt. [Chaigne and A. Askenfelt, ''Numerical simulations of piano strings. I. A physical model for a struck string using finite difference methods,'' J. Acoust. Soc. Am. **95**, 1112–118 (1994); A. Chaigne, J. Acoust. **5**, 181 (1992)]. The strings are coupled to a soundboard model that includes the added thickness and stiffness of ribs and bridges. The vibrating soundboard drives the room air, with walls modeled through a complex acoustic impedance following Botteldooren. [D. Botteldooren, ''Acoustical finite-difference time-domain simulation in a quasi-Cartesian grid,'' J. Acoust. Soc. Am. **95** 2313–2319 (1994); **98**, 3302 (1995)]. All of the calculations take a finite difference approach, producing a time domain computation. Calculated tones are presented and future directions discussed. [Work supported in part by NSF grant PHY-9988562.]

3:45

2pMU5. Finite-element simulation of the acoustic guitar. Gregoire P. Derveaux, Eliane Becache, Patrick Joly (INRIA, domaine de Voluceau, BP105, 78153 Le Chesnay Cedex, France, gregoire.derveaux@inria.fr), and Antoine Chaigne (ENSTA, 91761 Palaiseau Cedex, France)

The purpose of this work is a 3-D time-domain numerical modeling of the acoustic guitar. The model involves the transverse displacement of the string excited by a force pulse, the flexural motion of the soundboard, and the sound radiation. A specific spectral method is used for solving the Kirchhoff-Love's dynamic plate model for a damped, heterogeneous orthotropic material. The airplate interaction is solved with a fictitious domain method, which allows for making a good approximation of the geometry of the instrument while using a regular mesh for the calculation of the 3-D sound radiation. A conservative scheme is used for the time discretization. Frequency analysis is performed on the simulated sound pressure and plate velocity waveforms in order to evaluate quantitatively the transfer of energy through the components of the coupled system. The effects of structural changes in soundboard thickness and cavity volume on the produced sounds are presented. Simulations of the same guitar in three different cases are also performed: ''*in vacuo*,'' in air with a perfectly rigid top plate, and in air with an lastic top plate. This allows comparisons between structural, acoustic, and structural acoustic modes of the instrument.

<div style="text-align: right">2p TUE. PM</div>

TUESDAY AFTERNOON, 6 JUNE 2006　　　　　　　　　　　　ROOM 556AB, 1:00 TO 3:30 P.M.

Session 2pNSa

Noise, ASA Committee on Standards, and Psychological and Physiological Acoustics: New Loudness Standard

Rhona P. Hellman, Chair

Northeastern Univ., Speech, Language, Pathology and Audiology, 360 Huntington Ave., Boston, MA 02115-0000

Chair's Introduction—1:00

Invited Papers

1:05

2pNSa1. Rationale for a new loudness standard. Rhona P. Hellman (Dept. of Speech-Lang. Path. and Audiol. and Inst. for Hearing, Speech, & Lang. (106A FR), Northeastern Univ., Boston, MA 02115, hellman@neu.edu)

In 2005, a new loudness standard was approved by ANSI for calculating the loudness of steady sounds. The new standard is based on a recent loudness model [Moore *et al.*, J. Audio Eng. Soc. **45**, 224–240 (1997)] derived from an earlier model devised by Zwicker [Acustica **8**, 237–258 (1958)]. It replaces and supersedes the more limited computational procedure used in the old ANSI S3.4-1980 standard that had its origins in the protocol developed by S.S. Stevens [J. Acoust. Soc. Am. **33**, 1577–1585 (1961)]. Unlike the old ANSI S3.4-1980 procedure, the newly adopted ANSI standard can be applied to sounds with sharp line spectral components as well as to broadband spectra. The new standard also extends the dynamic range for loudness calculations to loudness levels below 40 phons. Hence, it can be used to estimate loudness, or loudness level, with reasonable accuracy at levels close to threshold. Moreover, in accord with the revision of ISO 226 [International Organization for Standardization, Geneva (2003)], it enables the loudness of complex sounds containing spectral energy below 500 Hz to be determined. Empirical examples that illustrate the broader and more accurate scope of the new loudness standard are presented.

1:30

2pNSa2. The Cambridge loudness model: Design principles and predictions. Brian C. J. Moore, Brian R. Glasberg, and Tom Baer (Dept. of Exp. Psychol., Univ. of Cambridge, Downing St., Cambridge CB2 3EB, UK, bcjm@cam.ac.uk)

The Cambridge loudness model, which forms the basis for the new ANSI standard (ANSI S3.4-2005), is similar in structure to loudness models proposed earlier by Fletcher and by Zwicker, but differs in many of the details. The input signal to each ear is specified in terms of its spectrum, which may be defined in 1/3-octave bands, or as the sum of a series of sinusoidal components and one or more pink-or white-noise bands. The first stage of the model is a filter to account for the transfer of sound from the sound field or headphone to the eardrum. The second stage is a filter to account for the transfer of sound through the middle ear. The third stage transforms the effective spectrum reaching the cochlea to an excitation pattern, calculated as the output of an array of level-dependent auditory filters. In the final stage, the excitation at each center frequency is converted to specific loudness, by a nonlinear frequency-dependent transformation. The specific loudness is summed across frequency to give the loudness for each ear, and is summed across ears to give overall loudness. The model accounts well for the effects of level, frequency, and bandwidth on loudness.

1:55

2pNSa3. Reexamining loudness at and near threshold. Michael Epstein (Auditory Modeling and Processing Lab., Inst. for Hearing, Speech & Lang., Dept. of Speech-Lang. Pathol. and Audiol., Northeastern Univ. Boston, MA 02115) and Mary Florentine (Northeastern Univ. Boston, MA 02115)

This talk reviews two sets of data that support the contention that loudness at threshold is not zero, consistent with the new loudness standard. Both sets of data examine the form of the loudness function in normal listeners at low levels. The first set of data was derived from loudness-matching experiments between equally loud tones and tone complexes (i.e., spectral integration of loudness). Listeners matched the loudness of tone complexes comprised of subthreshold components with pure tones above threshold. Results show that a tone complex composed of components at threshold can easily be heard and can be relatively loud. This result is only possible if all tones had some probability of detection (some contribution to loudness), even those below threshold. The second set of data consists of the slopes of the loudness function at threshold, which average 1.3. These data are consistent with loudness functions obtained with a wide variety of methods that yield a slope close to unity. A slope close to unity indicates a positive loudness at threshold in accord with the new loudness standard. [Work supported by NIH/NIDCD Grant R01DC02241.]

2:20

2pNSa4. The new International Organization for Standardization (ISO) standard for the equal-loudness contours: Comparison to earlier contours and procedures. Yôiti Suzuki (Res. Inst. of Elec. Commun., Tohoku Univ., Katahira 2-1-1, Aoba-ku, Sendai 980-8577, Japan, yoh@riec.tohoku.ac.jp) and Hisashi Takeshima (Sendai Natl. College of Technol., Sendai 989-3128, Japan)

Equal-loudness contours exhibit the intensity-frequency characteristic of the auditory system. Since 1956 such contours based on the free-field measurements of Robinson and Dadson [Br. J. Appl. Phys. **7**, 166–181 (1956)] have been widely accepted as an international standard in ISO 226. However, in 1985 some questions about the general applicability of these standard contours were raised to ISO TC/43 WG1 by Fastl and Zwicker. Consequently, a new international effort to measure the equal-loudness contours was undertaken. In 2003, this effort produced a revision of ISO 226. The newer experimental data and their analyses were subsequently published by Suzuki and Takeshima, [J. Acoust. Soc. Am. **116**, 918–933 (2004)]. The revised contours more closely agree with the classic contours measured by Fletcher and Munson [J. Acoust. Soc. Am. **5**, 82–108 (1933)] than with those produced by Robinson and Dadson. This finding is especially true below 500 Hz at moderate loudness levels. Moreover, the revised contours are in good overall agreement with the contours calculated according to the procedure proposed by Moore *et al.* [J. Audio Eng. Soc. **45**, 224–240 (1997)]. In 2005, this procedure was adopted by ANSI for loudness calculations of steady sounds to replace ANSI S3.4-1980. Comparative examples are given.

2:45

2pNSa5. Future directions in loudness standardization: Time-varying loudness. Patricia Davies and Andrew Marshall (Purdue Univ., 140 S. Intramural Dr., West Lafayette, IN 47907-2031, daviesp@purdue.edu)

The loudness of many sounds varies with time and the temporal processing of the human hearing system affects loudness perception. Existing loudness algorithms that are standardized address loudness of stationary sounds. Two algorithms that predict time-varying loudness have been developed: Zwicker's, and Moore and Glasburg's time-varying loudness; variants of the former are available in most sound-quality analysis systems. For continuous sounds with repetitive impulses or beating tones loudness exceeded 5% of the time correlates well with perceptions of overall loudness. The focus here is on predictions of time-varying loudness algorithms for isolated impulsive sounds caused by supersonic aircraft, because recent developments in supersonic aircraft design could result in aircraft producing much quieter booms that may be more acceptable to the general public. Loudness is a major factor in community response; thus, the research focus is to understand how sound modifications (amplitude reduction, signal shaping) affect the loudness of transients. Also, playback systems involve signal filtering; how this may affect loudness perception is of interest. The predictions of the two algorithms are compared for a series of sonic-boom recordings and modified recordings to determine how filtering and signal modification affect loudness predictions.

3:10–3:30

Panel Discussion

Session 2pNSb

Noise and Architectural Acoustics: Audio-Visual Design in Soundscapes II

Brigitte Schulte-Fortkamp, Cochair

Technical Univ. Berlin, Inst. of Technical Acoustics, Einsteinufer 25, 10587 Berlin, Germany

Bennett M. Brooks, Cochair

Brooks Acoustics Corporation, 27 Hartford Turnpike, Vernon, CT 06066

Continuation of session 2aNSa. Field measurements will be discussed and evaluated.

Session 2pPA

Physical Acoustics: Nonlinear Acoustics, Flow, and Miscellaneous Topics

Vera Khokhlova, Chair

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

Contributed Papers

1:30

2pPA1. Assessment of incipient material damage and remaining life prediction with nonlinear acoustics. Dimitri Donskoy (Intelligent Sensing Technologies, LLC, P.O. Box 6457, Fair Haven, NJ 07704-6457), Andrei Zagrai (Stevens Inst. of Technol., Hoboken, NJ 07030), Alexander Chudnovsky, Edward Golovin (Univ. of Illinois at Chicago, Chicago, IL 60607), and Vinod S. Agarwala (Naval Air Systems Command, Patuxent River, MD 20670)

Subjected to a wide spectrum of operational and environmental loads, engineering structures exhibit first signs of material deterioration at micro and meso scales. In contrast to nondestructive assessment of the macro-scale damage that resulted from material fracture, a limited number of technologies have tackled the complex task of detecting and monitoring the small-scale incipient damage. This paper discuses application of the nonlinear acoustic vibro-modulation technique for assessment of the micro/meso scale fatigue damage and remaining life prediction of metallic samples. A nonlinear acoustic damage index (DI), which indicates strength of the damage-induced nonlinear interaction of the high-frequency ultrasonic wave and the low-frequency structural vibration, was monitored during a set of strain-controlled fatigue tests. Stable growth of the damage index was observed at an increasing number of fatigue cycles showing correlation between DI and the micro/meso scale damage accumulation. This correlation allowed for introducing DI in the remaining life prediction methodology based on an entropy density criterion of local failure. Experimental testing confirmed the effectiveness of the proposed prediction approach. [Work supported by NAVAIR.]

1:45

2pPA2. Wave vector resonance in a nonlinear diffuse elastodynamic field. Alexei Akolzin and Richard L. Weaver (Dept. of Theoretical and Appl. Mech., Univ. of Illinois, 104 S. Wright St., Urbana, IL 61801)

Nonlinear coupling within an elastic body leads to spectral energy redistribution. The coupling strength can exhibit sharp discontinuities as a function of frequencies, due to wave vector coincidences between constituent waves of different wave speeds. The phenomenon is investigated for an ensemble of weakly nonlinear elastic bodies of a generic ray-chaotic shape, with the elastodynamic field inside being treated as diffuse and described in a statistical fashion. Comparison is made between theoretical estimates of the average mode coupling strength and results of direct numerical simulations performed for an ensemble of 2D plane-strain elastic squares with rough boundaries and a model quadratic nonlinearity representative of elastodynamic waves. Existence of the wave vector resonance and its dependence upon frequency ratios is verified.

2:00

2pPA3. Dynamic characteristics of nonlinear interaction in elastic medium. Nikolai Zagrai (Taganrog State Univ. of Radio-Eng. GSP-17A, 44 Nekrasovsky St., Taganrog, 347928, Russia)

Traditionally, an acoustic wave process is analyzed in terms of sound pressure and/or sound velocity. The dynamic distortion of the wave profile, however, is complicated by contribution of both quadratic and cubic nonlinearities of the elastic medium. This condition necessitates expanding a range of variables to include particle and wave accelerations. As dynamic characteristics, particle and wave accelerations fully complement

analysis of dynamic and inert properties of the elastic medium. The Riemanns solution, in which the quadratic and cubic nonlinearities are accounted for through respective derivatives of the sound velocity, allows for a new and different perspective on the aforementioned properties. For a simple wave with contribution of the quadratic nonlinearity only, the acceleration profile describing dynamics of wave deformation varies symmetrically relative to the discontinuity region. The cubic nonlinearity leads to more complex acceleration profile with asymmetric properties, which can be discerned and assessed using a combination of numerical and graphical methods. Such an approach is quite effective and vivid in separating contributions of quadratic and cubic nonlinearities. An integral nonlinear parameter N, introduced on the basis of this analysis, is proposed for estimating ability of the fluidlike medium to support effective propagation and nonlinear interaction of acoustic waves.

2:15

2pPA4. A frequency-domain algorithm for the propagation of finite-amplitude acoustic signals. Lauren E. Falco (Grad. Prog. in Acoust., Penn State Univ., 217 Appl. Sci. Bldg., University Park, PA 16802, lfalco@psu.edu), Philip J. Morris, and Anthony A. Atchley (Penn State Univ., University Park, PA 16802)

It has been shown that the quantity Q_{p^2p}, the cross-spectral density of the square of the pressure and the pressure, is related to the strength of nonlinear effects in a propagating acoustic signal. Morfey and Howell [AIAA J. **19**, 986–992, (1981)] derive from the Burgers equation an expression that relates Q_{p^2p} to the rate of change of the power spectral density (PSD) of a signal. They suggest a propagation scheme based on this expression that makes assumptions about the Gaussian nature of the signal. These assumptions are invalid for many finite-amplitude signals, so a new propagation scheme is proposed and validated. This scheme uses an expression for the rate of change of the phase of the signal as well as its amplitude. These two expressions are combined into a finite-difference equation that is used to step the signal to the desired propagation distance. The method is shown to compare favorably with the Fubini solution for lossless plane waves. Comparisons are also made with a numerical split-step algorithm for the nonlinear propagation of a broadband source, and with experimental data obtained in a plane wave tube. [Work supported by ONR, GE.]

2:30

2pPA5. Thermal gradient effects on the propagation of nonlinear pressure waves in a gas. Nicholas DiZinno, George Vradis, and Volkan Otugen (Mech. & Aerosp. Eng., Polytechnic Univ., 6 Metrotech Ctr., Brooklyn, NY 11201, dizinno@yahoo.com)

A numerical study of wave propagation through gases with nonuniform temperature distributions will be presented. The main objective of this study is to determine the impact of temperature gradients on high-intensity, initially sinusoidal pressure waves. Particular emphasis is paid to wave reflection, transmission, and any influence a high-temperature region may have on nonlinear behavior. Ultimately, the performance of thermal barriers in attenuating nonlinear waves is evaluated. The concept of using regions of hot gas inside an ambient environment has potential in aeroacoustic applications, such as jet screech mitigation. This analysis considers one-dimensional compressible unsteady Euler equations with an ideal gas state equation, applied over a uniform domain. The domain is composed of two regions with uniform and equal gas properties (ambient conditions) separated by a third region with higher gas temperature and, accordingly, lower density. Pressure is uniform throughout the domain. We introduce a high-intensity sinusoidal wave into this medium. The shape and extent of the high-temperature zone is varied to study the effect of this region on wave propagation. Further, wave reflection and transmission are studied for a range of wave and thermal field parameters. Results for nonlinear pressure waves are compared to linear acoustic waves.

2:45

2pPA6. Evanescent time reversal: Super-resolution and communications applications. Stephane G. Conti, Philippe Roux, and William A. Kuperman (SIO/UCSD, Marine Physical Lab., 9500 Gilman Dr., La Jolla, CA 92093-0238)

The spatial resolution of the focused field with a classical time-reversal mirror in the far field has a wavelength-order diffraction limit. Previously reported results for super-resolution beyond the diffraction limit require either the full knowledge of the original source, or the evanescent waves in the near field. Here, we show that subwavelength focusing can be achieved without *a priori* knowledge of the original probe source. In the wave-number domain, the contributions for the spatial localization and for the resolution of the focus can be separated from each other in the time-reversed field recorded within one wavelength of the original source. The evanescent time-reversal field is then obtained from the combination of the spatial localization contribution from the far-field time reversed with the spatial resolution for a point source. Further, we show that different signals can be transmitted simultaneously with subwavelength separation. Super-resolution of the order of 1/20 of the wavelength is experimentally demonstrated with acoustic wave fields. Applications for communications are presented using multiple receivers a fraction of the wavelength apart.

3:00–3:15   Break

3:15

2pPA7. Area oscillations of a bias-flow aperture, with application to phonation. Richard S. McGowan (CReSS LLC, 1 Seaborn Pl., Lexington, MA 02420, rsmcgowan@earthlink.net) and Michael S. Howe (Boston Univ., Boston, MA 02215)

A thin baffle plate in a mean flow duct partitions the tube into high- and low-pressure regions. The plate spans the whole cross section of the duct and is pierced by a small aperture, through which a nominally steady bias flow is maintained by the mean pressure differential. Sound is generated when the flow is disturbed by imposed, small amplitude cross-sectional area fluctuations of the aperture. For irrotational flow the radiation is determined by the input impedances of the tube on each side of the aperture, and energy flowing into the acoustic field is equal to the work done on the fluid by the suction force at the edge of the aperture. For high Reynolds number flow of a real fluid (modeled by the imposition of a Kutta condition) much of the work goes into the production of vorticity, the unsteady volume flow is greatly reduced, and at low frequencies there can be a substantial oscillating drag force on the plate [M. S. Howe, Proc. R. Soc. London, Ser. A **366**, 205–233 (1979)]. The application of these results to phonation indicates that the unsteady drag is most important in the lower harmonics of voice. [Work supported by NIDCD-004688 to UCLA.]

3:30

2pPA8. Simulation of fluid flow in and around a porous windscreen. D. Keith Wilson (U.S. Army Engineer Res. and Development Ctr., 72 Lyme Rd., Hanover, NH 03768, d.keith.wilson@erdc.usace.army.mil) and Michael J. White (U.S. Army Engineer Res. and Development Ctr., Champaign, IL 61826)

Fluid dynamic simulation of wind noise (turbulent pressure fluctuations) around porous microphone windscreens in an atmospheric flow is made difficult by the broad range of spatial scales, from large atmospheric eddies down to the pore size of the windscreen, that is involved. Here, we describe a simplified simulation approach that is based on two new insights: (1) the ability of linear fluid flow approximations to describe wind noise [Z.C. Zheng and B. K. Tan, J. Acoust. Soc. Am. **113**, 161–166 (2003)] and (2) a new set of macroscopic, time-domain relaxational equations for the linear interaction of a fluid flow with a porous material [D. K. Wilson, V. E. Ostashev, and S. L. Collier, J. Acoust. Soc. Am. **116**, 1889–1892 (2004)]. An upstream flow condition consisting of quasiwavelets

(analogous to turbulent eddies) is also used to create well-developed, realistic turbulence. This simulation approach is shown to provide detailed simulations of the production of pressure fluctuations in and around a porous windscreen.

3:45

2pPA9. Diffraction pattern of an axisymmetric aperture from axial measurements. Edward H. Pees (Dept. of Phys., Univ. of Rhode Island, 2 Lippitt Rd., Kingston, RI 02881-0817)

An axisymmetric aperture can be represented by a circular hole in an infinite rigid screen, a circular piston undergoing harmonic motion, or a more complicated source composed of annular ring transducers. Ordinarily, the radiation pattern for this type of aperture is measured by moving a sensor along a circular arc about the center of the aperture, measuring the intensity of the radiated field as a function of angle. In the present case, the angular spectrum of plane wave representation of the diffracted field is used to demonstrate that the diffraction effect leads to the presence of a wave number distribution, observable along the normal axis of the aperture, that can be used to reconstruct the far-field radiation pattern from measurements taken only along this angular view. Because this diffraction pattern has a direct relationship to the boundary condition on the aperture,

it follows that measurements of the axial field can potentially provide useful information regarding the size and shape of the aperture.

4:00

2pPA10. Transient total reflection and transient evanescent wave. Hailan Zhang (Inst. of Acoust., Chinese Acad. of Sci., Beijing 100080 China)

The total reflection of a steady plan wave is well known. The transmitted wave is an evanescent wave that propagates along one direction and evanesces along a direction perpendicular to the propagation direction. Evanescent waves also appear in many other acoustic problems, such as the surface wave along a free boundary of a solid medium. In this report, the total reflections with different incident waves are studied, including an incident beam with a limited width, an incident pulse wave with a limited duration, and an incident bulletlike wave package. These incident waves are represented by superposition of plane waves based on the angle spectrum synthesis. Their transmitted wave, therefore, can be synthesized from the transmitted waves of the plane incident wave. Snapshots of the acoustic field are presented that show the procedure of the interaction of the incident wave with the interface of the media. [Work supported by National Natural Science Fundation of China, No. 10534040.]

TUESDAY AFTERNOON, 6 JUNE 2006

ROOM 556AB, 3:45 TO 5:20 P.M.

Session 2pPP

Psychological and Physiological Acoustics: Spatial Hearing

Norbert Kopco, Chair
Boston Univ., Cognitive and Neural Systems, 677 Beacon St., Boston, MA 02215

Chair's Introduction—3:45

Contributed Papers

3:50

2pPP1. Spatial audio target acquisition in the frontal horizontal plane: Effects of feedback, mobility, and index of difficulty. Georgios Marentakis and Stephen Brewster (Dept. of Computing Sci., Univ. of Glasgow, 17 Lilybank Gardens, G12 8QQ UK)

The results of an empirical study investigating the effect of feedback, mobility, and index of difficulty on a deictic spatial audio target acquisition task in the horizontal plane in front of a user are presented. Audio feedback marked display elements are found to enable usable deictic interaction that is performed with high accuracy and can be described using Fitts' law. Feedback does not affect perceived workload or preferred walking speed compared to interaction without feedback. Without audio feedback, spatial audio is not found to support adequately the corrective submovements necessary for high selection accuracies. Interaction with the display becomes therefore highly inefficient. Mobility is found to degrade interaction speed and accuracy by 20%. Participants were able to perform deictic spatial audio target acquisition when mobile while walking at 73% of their preferred walking speed. The proposed feedback design is examined in detail and the effects of variable target widths are quantified. Deictic interaction with a spatial audio display is found to be a feasible solution for future interface designs.

4:05

2pPP2. Effect of roving on spatial release from masking for amplitude-modulated noise stimuli. Norbert Kopčo, Jaclyn J. Jacobson, and Barbara G. Shinn-Cunningham (Hearing Res. Ctr., Boston Univ., 677 Beacon St., Boston, MA 02215, kopco@bu.edu)

A previous study investigating detection of broadband signals embedded in broadband noise showed that detection thresholds depended both on the temporal structure of the target and masker and their simulated spatial locations [Kopco and Shinn-Cunningham, J. Acoust. Soc. Am. **117**, 2396 (2005)]. The current study extends this work by introducing a rove in overall intensity to reduce the reliability of overall intensity cues for detecting the target. Detection thresholds were measured for a broadband noise target spectrally and temporally centered within a broadband noise masker. Thresholds were measured for all combinations of six spatial configurations of target and masker and five modulation conditions (all combinations of target modulated and unmodulated and of masker modulated and unmodulated; when both target and masker were modulated, the modulation could either be equal or π out of phase). The amplitude modulation, if present, had a rate of 40 Hz and depth of 0.5. The masker level was roved by ± 5 dB between intervals within a trial. Results suggest that the cues underlying target detection depend on both the spatial configura-

tion of target and masker and their temporal structure, but that, when available, overall intensity cues also contribute to a target detection. [Work supported by NSF and ONR.]

4:20

2pPP3. Experiments in speech localization using headphones and loudspeakers. Casey Canfield, Michelle Vu (Garrison Forest School, 300 Garrison Forest Rd., Owings Mills, MD 21117), Ilene J. Busch-Vishniac, and James E. West (Johns Hopkins Univ., Baltimore, MD 21218)

This experiment studies the ability of humans to localize sound as recording and playback methods are varied. Further, because subjects are asked to indicate personal preference, we are able to determine the importance of localization in setting personal preference. The stimulus used in this experiment was speech (6 stanzas of a poem, each recorded at a different location). Three recording methods were used: monaural, stereo, and stereo through an artificial head. Two playback methods were used: stereo speakers and stereo headphones. Results of the work are applicable to the design of conference facilities and the design of headphones and speakers. [This work is a Wise Women project. Wise Women partners high school students from Garrison Forest School with Johns Hopkins engineering faculty.]

4:35

2pPP4. Evaluating the principal spectral components positioning a virtual sound source on a cone of confusion. David H. Benson and William L. Martens (CIRMMT, Faculty of Music, McGill Univ., 555 Sherbrooke St. West, Montreal, QC H3A-1E3, Canada, david.benson@mcgill.ca)

Although many studies have attempted to identify spectral cues to sound-source direction for the entire sphere of possible incidence angles, few studies have focused their attention on directional hearing for virtual sources positioned on cones of confusion. These regions of space, defined by relatively constant interaural time and intensity differences, provide an ideal subset of incidence angles for testing the importance of the spectral cues to direction allowing for up/down and front/rear distinctions between sources at constant angular distance from the median plane. In this study, head-related transfer functions (HRTFs) measured on a well-lateralized cone of confusion were decomposed into principal spectral components putatively responsible for positioning a virtual sound source, and their related angle-dependent scores that describe the relative contribution of each shape to the total spectral variation. The findings can be summarized as follows: For the set of ipsilateral HRTFs the most significant set of scores shows sinusoidal variation with angle, and could potentially encode a front-back cue, though with the extrema rotated by about 30 deg. The most significant score from an analysis of interaural spectral differences seems to capture a head-shadowing effect which is most extreme in the lower rear.

4:50

2pPP5. The effect of spectral differences on the ability to judge the laterality of a simulated source followed by a simulated echo. Joseph B. Boomer, Raymond H. Dye, Jr., Stanley Sheft, and William A. Yost (Parmly Hearing Inst., Loyola Univ., 6526 N. Sheridan Rd, Chicago, Chicago, IL 60626)

The purpose of this study was to examine listeners' abilities to lateralize a source click based on interaural difference of time (IDTs) when followed by a spectrally different echo click. The first click was centered at 3000 Hz and the second was centered at 1500, 2000, 3000, 4000, or 5000 Hz. A correlational analysis was carried out to assess the relative weight given to the IDTs of source and echo clicks for echo delays of 8–128 ms. Also measured were proportions correct and proportions of responses predicted from the weights. Listeners were instructed to indicate the laterality of the source click. The IDTs of the source and echo clicks were selected independently from Gaussian distributions ($= 0$ s, $= 100$ s). When the second click had energy in a spectral region above that of the first, source weights and proportions correct were higher than when the spectral region of the second was lower than the first click. For some observers, this low-frequency dominance was evident for echo delays as long as 128 ms. The effect of echo frequency was constant across all echo delays, although source weights and proportions correct were highest at an echo delay of 8 ms.

5:05

2pPP6. Learning of interaural-level-difference discrimination with a 0.5-kHz tone in human adults. Yuxuan Zhang and Beverly A. Wright (Dept. of Commun. Sci. and Disord., and Northwestern Univ. Inst. for Neurosci., Northwestern Univ., 2240 Campus Dr., Evanston, IL 60208)

Sound-source location on the horizontal plane is primarily determined by two cues: interaural time differences (ITDs) for low-frequency sounds and interaural level differences (ILDs) for high-frequency sounds. Though not available in natural listening environments, ILDs in low-frequency sounds, when artificially delivered to listeners through headphones, also can be used to lateralize sounds. Little is known about the neural processing of such low-frequency ILDs. Here, the modifiability of this processing was examined by training human adults on ILD discrimination with a 0.5-kHz tone on the midline. After nine 1-h daily training sessions, trained listeners ($n = 6$) improved significantly more than untrained controls ($n = 8$) on the trained condition. This learning generalized to a 4-kHz tone and a 0.5-kHz tone amplitude modulated at 30 Hz, but appeared not to generalize to a 1-kHz tone, an off-midline position, or ITD discrimination with the trained tone. This overall learning pattern differs from those previously obtained for ILD at 4 kHz and ITD at 0.5 kHz using the same training paradigm [Wright and Fitzgerald, Proc. Natl. Acad. Sci. U.S.A. **98**, 12307–12312 (2001)], suggesting that the processes involved in low-frequency ILD discrimination differ from those encoding high-frequency ILDs and low-frequency ITDs, at least at the neural site(s) affected by training. [Work supported by NIH/NIDCD.]

Session 2pSAa

Structural Acoustics and Vibration: Structural Acoustics of Vehicles

Gerard P. Carroll, Cochair
Naval Surface Warfare Ctr., Carderock Div., 9500 MacArthur Blvd., West Bethesda, MD 20817

Hari S. Paul, Cochair
Acoustical Foundation, 101/20 First Main Rd., Gandhi Nagar Adyar, Chennai 600020, India

Contributed Papers

1:15

2pSAa1. Unsteady surface pressures on a ship hull. Michael Goody, Theodore Farabee, and Yu-Tai Lee (Signatures Directorate, Naval Surface Warfare Ctr., Carderock Div., 9500 MacArthur Blvd., West Bethesda, MD 20817-5700, Michael.Goody@navy.mil)

Unsteady surface pressure measurements at locations distributed on the surface of a ship model hull and flow visualization of the bow wave and free surface are discussed. The measurements were performed in the David Taylor Model Basin using a ship model that is approximately 40 ft. long and representative of a naval combatant at Froude numbers from 0.14 to 0.44. The pressure measurement locations are distributed over the hull surface from 15 surface pressure spectrum and limited spatial coherence measurements are discussed. Additionally, the measured surface pressure spectra are compared to predictions done using an RANS-based formulation and predictions done using an empirical formulation that is based on historical, flat-plate, surface pressure spectra. [Work sponsored by Office of Naval Research, Code 334.]

1:30

2pSAa2. Power flow and wave propagation characteristics of surface ship inner-bottom structure. Robert C. Haberman (BBN Technologies, Old Mystic Mill, 11 Main St., Mystic, CT 06355, rhaberman@bbn.com)

The construction of the inner bottom of a surface ship consists primarily of deck plating connected to hull plating by longitudinal vertical plates. Foundations supporting equipment typically line up with the vertical plates, thereby providing a direct path of mechanical excitation to the hull resulting in forced acoustic radiation. The purpose of this paper, therefore, is to provide an overview of the physics that controls wave propagation in surface ship inner-bottom structure along with a description of the power flow associated with direct force excitation of the vertical plating. Two theories will be presented: (1) an approximate theory based on the vertical plate modeled as a beam coupled to the deck and hull modeled as plates containing flexural waves and (2) an exact theory based on all three structures modeled as plates containing flexural, compressional, and shear waves. Numerical results will be presented that show the importance of full wave coupling along with some interesting phenomena associated with power flow and leakage of energy into deck and hull plating. The implication of these results in statistical energy analyses modeling will be briefly discussed. [Work partially funded by ONR, Code 334 Dr. S. Schreppler.]

1:45

2pSAa3. Radiated acoustic power of model ship hull sections estimated from scanning laser Doppler vibrometer vibration response measurements. Gerard P. Carroll (Naval Surface Warfare Ctr. Carderock Div., 9500 MacArthur Blvd., Bethesda, MD 20817)

This paper will present radiated acoustic power estimates for ship hull models which are based entirely on scanning laser Doppler vibrometer measurements of the underwater vibrating surface. The radiated power estimates are compared to direct measurements of the radiated acoustic power obtained using hydrophones in the reverberant tank. These radiation estimates are based on Rayleigh integral formulations, one for which the vibrating surface is assumed to be baffled, and the other for which it is unbaffled. For the latter case, the pressure in the plane of the vibrating surface is assumed to be zero outside the vibrating surface. This corresponds to a mixed boundary condition in which the velocity is known on the plate and the pressure is known off the plate. An iterative application of the Rayleigh integral is used to determine the velocity and the pressure over the entire plane. For the purposes of calculating the radiated power, this formulation is shown to iterate much more quickly by iteratively calculating the surface intensity. This is possible since the pressure and therefore the surface intensity is zero outside the plate boundary. Utilizing these surface radiation intensity calculations, it is possible to then investigate the sources and mechanisms of the hull radiation.

2:00

2pSAa4. Numerical investigation of small-scale surface ships by energy finite-element analysis. Kuangcheng Wu (Dept. of Signatures and Hydrodynamics, Northrop Grumman Newport News, 4101 Washington Ave., Newport News, VA 23607, kcwu@ngc.com) and Nick Vlahopoulos (Univ. of Michigan, MI 48109)

Experimental investigations have been performed to understand the structural acoustic characteristics of two small-scale surface ship models representing sections of conventional and advanced double-hull ship designs [Craun *et al.*, "Experimental and analytical investigation of surface shiphull structure acoustics using physical models," J. Acoust. Soc. Am. **116**, 2569 (2004)]. Recently, new developments have been added to the energy finite-element analysis (EFEA) for surface ships [Wu and Vlahopoulos, "Vibratory response of surface ships predicted by energy finite element approach," J. Acoust. Soc. Am. **116**, 2569 (2004)]. The presentation will compare hull responses from EFEA predictions to test data for the two small-scale models to validate the new development in EFEA. Furthermore, the impact of the hull responses due to the different model designs will be discussed. [Support for this research is provided by ONR code 334.]

2:15

2pSAa5. Application of a hybrid finite-element formulation for structure-borne vibration in an automotive vehicle structure. Sang-Bum Homg (Cummins, MC 50182, 1900 McKinley Ave., Columbus, IN 47201) and Nickolas Vlahopoulos (Univ. of Michigan, Ann Arbor, MI 48109)

The hybrid FEA method combines the conventional FEA method with energy FEA (EFEA) for analyzing systems that contain both flexible and stiff members. Conventional FEA models are employed for modeling the behavior of the stiff members in a system. Appropriate damping elements are introduced in the connections between stiff and flexible members in order to capture the presence of the flexible members during the analyses of the stiff ones. The component mode synthesis method is combined with

3297   J. Acoust. Soc. Am., Vol. 119, No. 5, Pt. 2, May 2006

151st Meeting: Acoustical Society of America 2006   **3297**

2p TUE. PM

analytical solutions for determining the driving point conductance at joints between stiff and flexible members and for defining the properties of the damping elements which represent the flexible members when analyzing the stiff components. Once the vibration of the stiff members and the amount of power dissipated at the damping elements has been identified, an EFEA analysis is performed in order to determine the amount of vibrational energy in the flexible members. The new developments are validated by comparing results of the hybrid FEA with results from very dense conventional finite-element analyses for a typical body-in-white vehicle structure. [Work sponsored by the Automotive Research Center, supported by the U.S. Army NAC.]

2:30

2pSAa6. The role of structural acoustics in the space shuttle flowliner crack investigation. Michael J. Allen and Robert C. Haberman (BBN Technologies, Old Mystic Mill, 11 Main St., Mystic, CT 06355, mjallen@bbn.com)

During normal inspection of the Space Shuttle main propulsion fuel line system small cracks were identified in a slotted flowliner just upstream of the main propulsion engines. The cause is believed to be resonance excited by pressure fluctuations originating from the downstream fuel pump. This presentation highlights the analyses that were performed to characterize the structural acoustic environment surrounding the flowliner during waterflow and liquid hydrogen laboratory experiments, as well as in-flight conditions. Analyses employed a combination of analytical and finite-element methods to model fluid structure interaction and inverse methods to reconstruct equivalent acoustic sources representative of the fuel pump. Numerical results will be presented that compare experi-

mental strain and pressure data with theoretical counterparts. Implications will be drawn regarding the applicability of the theory to the actual shuttle fuel line system. [Work was sponsored by NASA Marshall Flight Center.]

2:45

2pSAa7. Structural monitoring from noise cross correlation. Karim G. Sabra, William A. Kuperman (Marine Physical Lab., Scripps Inst. of Oceanogr., U.C. San Diego, 9500 Gilman Dr., La Jolla, CA, 92093-0238, ksabra@mpl.ucsd.edu), Eric S. Winkel, Dwayne A. Bourgoyne, David R. Dowling, Steve L. Ceccio, and Marc Perlin (Univ. of Michigan, Ann Arbor, MI 48109-2133)

It has been demonstrated theoretically and experimentally that an estimate of the Green's function between two receivers can be obtained from the long-time average of the cross correlation of ambient noise at these two receivers in various environments and frequency ranges of interest: ultrasonics, underwater acoustics, and seismology. This result provides a means for structural monitoring using the ambient structure-borne noise only, without the use of active sources. The coherent wave-fronts emerge from a correlation process that accumulates contributions over time from noise sources whose propagation path passes through both receivers. We present experimental results using random vibration data collected on eight accelerometers mounted on a hydrofoil or plate at the Navy's William B. Morgan Large Cavitation Channel. The tests were conducted at high Reynolds number, with the primary excitation source being turbulent boundary-layer pressure fluctuations on the upper and lower surfaces of the plate or foil. Identical deterministic time signatures emerge from the noise cross-correlation function based on robust and simple signal processing between passive sensors. These time signatures can be used to estimate and monitor the structural response of the test models.

TUESDAY AFTERNOON, 6 JUNE 2006

ROOM 551B, 3:15 TO 4:45 P.M.

Session 2pSAb

Structural Acoustics and Vibration: Damping and Materials

Courtney B. Burroughs, Chair
1241 Smithfield St., State College, PA 16801

Contributed Papers

3:15

2pSAb1. A strain energy approach to the flexural damping of layered plates. Richard J. Deigan and Kin Ng (NSWCCD, 9500 MacArthur Blvd, W. Bethesda, MD 20817-5700)

The flexural damping factor of multilayered plates is developed using a strain energy approach. The derivation is based on the exact, steady-state solution for a plate subjected to boundary conditions that are harmonic in time and position along the plate. The distribution of elastic strain energy contained within the individual layers of the plate is developed, from which an effective, flexural damping factor is determined. Comparisons to the widely used RKU analysis [D. Ross, E. E. Ungar, and E. M. Kerwin Jr., ''Damping of Plate Flexural Vibrations by Means of Viscoelastic Laminae,'' ASME, Section 3, pp. 49–87 (1959)] are made for selected cases of constrained and unconstrained layer damping. Good agreement with RKU theory is demonstrated for the selected cases. Advantages of the present analytic approach include: (1) The ability to model the effect of fluid loading; (2) Unlimited number of plate layers; and (3) The handling of compressible layers and associated loss mechanisms. Analytical estimates of damping factors are compared with measurements made on steel bars treated with an unconstrained damping layer. Stress-state assumptions are shown to be a major factor affecting theory-data agreement.

3:30

2pSAb2. Dynamic modulus measurement using the Dynamic Mechanical Analyzer. Kin Ng, Richard Deigan, and Samedi Koch (Naval Surface Warfare Ctr., Carderock Div., 9500 MacArthur Blvd., West Bethesda, MD 20817-5700)

The Navy has used the Dynamic Mechanical Thermal Analyzer (DMTA) as a tool to characterize material properties over the last 20 years. It is now evaluating the Dynamical Mechanical Analyzer (DMA) as a replacement for the DMTA, with the goal to insert it into operation in 2006 with minimal disruption to current acquisition programs. This paper compares the two instruments with regard to their capabilities and looks at some of the transition issues.

3:45

2pSAb3. Space-dependent damping for reducing vibration of continuous systems. Mark H. Holdhusen (Univ. of Wisconsin—Marathon County, 518 S 7th Ave., Wausau, WI 54401)

Structural vibration in continuous systems is a common source of problems for many mechanical systems. Much research has considered ways to reduce the vibration in continuous systems. Some research uses passive techniques that are simple to implement, whereas other methods

---

implement active control that can be quite complex. The research presented here investigates the effect space-dependent damping has on the vibration of continuous systems. Specifically, the work at hand will focus on the flexural vibration of beams. In a distributed damping beam system, the damping is variable as a function of position along the length of the beam. Such a system could be physically realized through the use of magneto-rheological or electro-rheological fluids. Along its length, a beam could be divided into several isolated sections containing such a fluid. By controlling the magnetic or electric field across each section of the beam, the damping would vary along the length of the beam. The work presented here optimizes the damping as a function of beam position to reduce the broadband frequency response of the beam. A number of support conditions for the vibrating beam are considered.

4:00

2pSAb4. Multiple-particle tracking rheometry of gelling polymer solutions. Ryan Braun (NSWCCD, 9500 MacArthur Blvd, W. Bethesda, MD 20817-5700) and Dennis Wirtz (Johns Hopkins Univ., Baltimore, MD 21218)

Many polymer materials used for damping are generated utilizing polycondensation reactions. These polymer reactions may start in the liquid phase, undergo gelation, and then solidify. The microscopic, three-dimensional, heterogeneous network produced during the chemical reaction greatly affects the bulk material properties. The polymer architecture is mainly locked in at the point of gelation, when polymer mobility is drastically reduced. This study investigates the steady-state microstructure and micromechanics of gelling polymer materials using the recently developed method of multiple-particle tracking microrheometry (MPTM). Unlike existing methods, MPTM quantitates, in real time, the microrheology and microheterogeneity of complex fluids undergoing gelation. Brownian trajectories of probe particles are transformed into families of mean-squared displacements, from which distributions in local, frequency-dependent viscoelastic moduli or time-dependent creep compliance are readily extracted. To understand the effects the microscopic, three-dimensional architecture of a gel has on the macroscopic, bulk properties, a model polysaccharide gel system was studied. The steady-state properties of the polysaccharide gels were evaluated as a function of concentration.

4:15

2pSAb5. Metal-polymer-ceramic constrained layer damping system. Liya Bochkareva (UIIP Natl. Acad. of Sci. of Belarus, Filatova 7-28, Minsk 220026, Belarus) and Maksim Kireitseu (Univ. of Sheffield, UK)

The focus of this paper is directed toward the investigation into multilayered metal-polymer-ceramic damping coatings and their FEM-based computer modeling that can provide an alternative spacer material in a constrained layer damping configuration so as to enable design of three

and more layered damping structures, but with controlled structure and mechanical properties of the gradient damping layers. An advanced measuring system for the analysis of vibrating structures is based on a laser scanning vibration interferometer. The position of a ceramic layer in a constrained layer configuration does matter. The outward position is most favorable for high strength by ceramic layer and damping by constrained aluminum-polymer layer. Modal loss factors obtained in the 60%–80% partial-coverage test cases of cylindrical shell are higher than the maximum modal damping obtained in the case of 100% coverage. The modal damping for the target modes is higher than in the complete-coverage case. An effective damping design involves selecting a proper combination of area coverage, relative thickness and stiffness values of the layers of the damped configuration. [Dr. Bochkareva is currently continuing her research work under 2-year EU INTAS 2005-2007 postdoctoral fellowship Ref. No. 04-83-3067.]

4:30

2pSAb6. Carbon nanotube-reinforced damping materials: Design and experiments. Maksim Kireitseu (INDMASH Natl. Acad. of Sci. of Belarus, Filatova 7-28, Minsk 220026, Belarus)

The CNT-reinforced material damping phenomenon is complex because of the variety of energy dissipation/fracture mechanisms involved, the complex nature of the nanoparticles themselves, etc. that affect damping/dynamics. The focus of this paper is directed toward the experimental validation of the next-generation vibration damping materials, providing a road map to experimental calibration and measurement of these materials. Modified modal strain energy method and FEM-based model created in MSC.visualNastran is employed to support an experiment. SWNT-reinforced polymeric matrix composites are prepared using an extrusion procedure. SWNT particles produced by Dynamic Enterprises, UK (2–5 nm in diameter and 70–90 vol % sample purity) are added in concentrations of 3%–6% by weight along with a surfactant (polyoxyethylene) to aid in dispersion. Resonance frequencies, the mode shape, and damping at each mode were determined by laser vibrometry at standard vibration shaker tests and Bruel & Kjaer vibration system. Data acquisition and control of the electro-dynamic system is based on the computer measurement dystem that is highlighted. Resonance frequency is determined from peaks on a frequency response curve for each of the experimental methods. [Work supported by Marie-Curie Fellowship Ref. No. 021298-Multiscale Damping 2006-2008 at University of Sheffield, UK with Prof. G.R. Tomlinson.]

Session 2pSC

Speech Communication: Speech Production (Poster Session)

Rajka Smiljanic, Chair
Northwestern Univ., Dept. of Linguistics, 2016 Sheridan Rd., Evanston, IL 60208

Contributed Papers

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

2pSC1. Finite-element simulations of self-oscillation mechanisms in human phonation. Douglas Cook, Luc Mongeau, and Paul Smith (Purdue Univ., 585 Purdue Mall, West Lafayette, IN 47907)

The fluid-structure interactions which occur during human phonation were investigated using a commercially available finite-element program. Both lumped mass and continuous structural models were coupled with viscous fluid flow. The simulated behavior was analyzed. The addition of viscous flow was not sufficient to induce self-oscillation of a single degree-of-freedom lumped mass model but did contribute to the steady vibration of the continuous model. This observation reinforces the conclusion that structural complexity is an essential mechanism of self-oscillation. The influence of false folds on the oscillation patterns of the continuous model was also examined.

2pSC2. Characteristics of phonation in women who produce voiced versus voiceless /h/. Laura L. Koenig (Haskins Labs., 300 George St., New Haven, CT 06511 & Long Island Univ., Brooklyn, NY 11201)

Production of the consonant /h/ typically involves a vocal-fold abduction gesture. Although /h/ is traditionally described as voiceless in English, voiced allophones also occur. Our recent work has described the phonatory behavior of women who produce a mixture of voiced and voiceless /h/; our results suggested speaker-specific patterns of the factors that lead to devoicing. This study characterizes the behavior of normal women who produce exclusively voiced /h/ and compares it to those who produced both voiced and voiceless /h/. Measures of baseline (dc) airflow are used as an index of abduction degree; peak intraoral pressures during adjacent /p/ closures provide a measure of subglottal pressure; and pulse-by-pulse measures are made of $f0$ and pulse amplitude (ac flow) around the /h/ abduction gesture. Multidimensional analyses are carried out within speakers to determine how these vocal parameters are related. We expect that women who produce only voiced /h/ will differ from those who devoice some /h/ productions in the parameters that facilitate phonation, such as higher subglottal pressures, smaller abduction degrees, and reduced vocal-fold tension as reflected in $f0$ measures. Based on our previous findings, however, we also expect individual variation in the phonatory settings that produce voiced /h/. [Work supported by NIH.]

2pSC3. Contextual effects on the continuancy of /ð/. Sherry Zhao (MIT, 77 Massachusetts Ave., Rm. 36-545, Cambridge, MA 02139)

This research examined the segmental-contextual effects on the continuancy of the voiced dental /ð/ in American-English and whether certain acoustic attributes were preserved despite possible modification. Word-initial /ð/ cases, extracted from continuous speech of 146 speakers from the TIMIT database, were frequently (averaging 65%) stoplike when it was in utterance-initial position or when its preceding phoneme was voice-less and/or [-continuant]. This stoplike modification occurred less frequently (averaging 39%) when /ð/ was preceded by a voiced fricative and rarely (averaging 13%) when preceded by a vowel or liquid consonant. A comparison of stoplike /ð/ and /d/ cases under similar contexts showed that the burst peak location, burst spectrum shape, and F2 at vowel onset averages were all statistically different between the two groups. In addition, the acoustic data suggested that the dental place of articulation was preserved for the modified /ð/. Preliminary automatic classification experiments involving salient acoustic attributes seemed to indicate that F2 at vowel onset may be a reliable cue for the dental place of articulation in /ð/. [Work supported by NIH DC02978 and T32DC00038.]

2pSC4. Modeling the influence of acoustic loading on laryngeal self-sustained oscillations. Matías Zañartu, Luc Mongeau, and George R. Wodicka (Ray W. Herrick Labs. Purdue Univ., West Lafayette, IN 47907)

Three-way interactions between sound waves in the subglottal and supraglottal tracts, the vibrations of the vocal folds, and laryngeal flow were investigated. Nonlinear coupling was investigated analytically, with a focus on the effects of a realistic acoustic loading on the self-sustained oscillation of the vocal folds. MRI-based shapes for the supraglottal tract [Story *et al.*, J. Acoust. Soc. Am. **100**, 537–554 (1996)] and an equivalent acoustical model for the subglottal tract [Harper *et al.*, IEEE Trans. Biomed. Eng. **48**, 543–550 (2001)] were considered. The model was based on the wave reflection analog technique, along with a novel single degree of freedom model of the vocal folds to describe the source. The effects of a mucosal wave on the vocal folds were incorporated through a time-varying discharge coefficient and a Bernoulli-type flow model. The relative importance of acoustic loading and time-varying flow resistance for fluid-structure energy transfer was compared for various configurations. [Work supported by NIH Grant RO1-DC05788.]

2pSC5. Patterns of interarticulator coordination within the syllable. Patrizia Bonaventura and Joseph Sand (Dept. of Commun. Sci., Case Western Reserve Univ., 11206 Euclid Ave, Cleveland, OH 44106, patrizia.bonaventura@case.edu)

Interarticulatory coordination patterns were observed in movements of the crucial articulator for the production of place in selected syllables containing labiodental and alveolar consonants. The coordination of the lower lip with upper lip and tongue tip, and the coordination of the tongue tip with tongue blade and lower lip have been observed. The goals of the study were (a) to verify whether stable patterns of speed with respect to excursion of movement [Fujimura, O., "Temporal organization of articulatory movements as a multidimensional phrase structure," Phonetica, **38**, 66–83 (1981a); "Icebergs revised," ASA meeting, 2aSC (1996b)], were

3300   J. Acoust. Soc. Am., Vol. 119, No. 5, Pt. 2, May 2006

151st Meeting: Acoustical Society of America 2006   **3300**

found in concurrent articulators to the crucial one for place, under different prosodic conditions; (b) whether such stable portions would vary proportionally to the increase in amplitude of the vertical jaw movement, selected as a measure of the magnitude of the syllable; (c) to observe relative timing at the iceberg thresholds in the selected consonantal movements with respect to the center of the syllable. Analyses were performed on articulatory recordings from cinefluorographic data, based on read dialogues [Erickson *et al.*, ''Articulatory correlates of prosodic control: Emotion and emphasis,'' Lang. Speech, **41**, 399–417 (1998)]. Iceberg patterns were found in the concurrent articulator movements, showing a linear dependence of slope on the total excursion of the demisyllabic movement. Timing of concurrent articulator movements with respect to the center of the syllable is shown to be relatively stable.

2pSC6. Modulation of frication noise in a dynamic mechanical model of the larynx and vocal tract. Anna Barney (ISVR, Univ. of Southampton, Southampton, SO17 1BJ, UK, ab3@soton.ac.uk) and Philip J. B. Jackson (Univ. of Surrey, Guildford, GU2 7XH, UK)

A dynamic mechanical model of the larynx and vocal tract was used to investigate the aerodynamics and acoustics of voiced fricatives. The model had typical dimensions for a human adult male. Glottal excitation arose from driven shutters representing the vocal folds; sinusoidal and square-wave driver functions were investigated. Glottal amplitude, fundamental frequency ($f0$), and volume velocity were independently controlled. Frication noise was produced by an orifice plate with a sharp-edged obstacle on its downstream side. Two different positions of constriction within the vocal tract, each coupled with a different constriction-obstacle distance, were investigated. Vocal-tract wall pressure was measured at three locations and a microphone at the vocal-tract exit measured radiated sound. The sound radiated from the open end of the vocal tract was found to have both harmonic and noise components. The noise was modulated by the $f0$ of the glottal vibration. The depth of modulation was independent of the shape of the glottal excitation. It showed an overall decreasing trend with increasing $f0$ in the range 20 to 150 Hz, but with a local maximum of modulation depth around 80 Hz. The noise bursts had maxima coincident with the middle of the glottal open phase.

2pSC7. Acoustic characteristics of clearly produced fricatives. Kazumi Maniwa, Allard Jongman (Linguist. Dept., Univ. of Kansas, 1541 Lilac Ln., Blake Hall, Rm. 427, Lawrence, KS 66044-3177), and Travis Wade (Posit Science Corp., San Francisco, CA 94104)

Research suggests that speakers can adopt a speaking style that allows them to be understood more easily when confronted with difficult communication situations, but few studies have examined the acoustic properties of clearly produced consonants in detail. This study attempts to characterize the type and magnitude of adaptations in the clear production of English fricatives in a carefully controlled range of communication situations. Ten female and ten male talkers produced nonsense syllables containing the eight English fricatives in VCV contexts, both in a conversational style and in a clear style (elicited by means of feedback consisting of simulated recognition errors received from an interactive computer program). Acoustic measurements were taken for spectral, amplitudinal, and temporal properties known to influence fricative recognition. Results illustrate that (1) there were some consistent overall clear speech effects, several of which (consonant duration, spectral peak location, spectral moments) were consistent with previous findings and a few (notably consonant-to-vowel intensity ratio) which were not; (2) ''contrastive'' differences related to eliciting prompts were observed in a few key comparisons; and (3) talkers differed widely in the types and magnitude of acoustic modifications.

2pSC8. Articulation of fricatives: Evidence from X-ray microbeam data. Christine H. Shadle, Khalil Iskarous, and Michael I. Proctor (Haskins Labs., 300 George St., New Haven, CT 06511)

Articulatory changes in /s/ over time give rise to the onset and offset of turbulence noise, but the small tolerances involved make it difficult to separately consider the effects of movement, phonetic context, and subject variation. Aerodynamic estimates of the constriction area have been shown to demonstrate a marked asymmetry, with the /s/ onset more rapid than offset [C. Scully, Speech Commun. **11**, 1992]. Previous articulatory studies led us to expect an invariant tongue tip position, with jaw raising dependent somewhat on the following vowel. In order to investigate the timing of the constriction formation in fricatives, X-ray microbeam data [Westbury *et al.*, 1994] for citation forms of /sVd/ for 15 vowels were studied. In fact, the tongue marker patterns during [s] vary both by following vowel and subject; subjects seem to have different apical-laminal strategies. The jaw reaches maximum height during [s] for each subject studied; that height does not appreciably vary with vowel. This indicates that jaw height is important specifically for /s/ production. One possibility is that the constriction is formed by the tongue, and turbulence noise is then maximized by the lower teeth being brought into position. [Work supported by NIH NIDCD R01 DC 006705.]

2pSC9. A range of intonation patterns produced in an elicitation task. Alejna Brugos, Jonathan Barnes (Program in Appl. Linguist., Boston Univ., 96 Cummington St. #246, Boston, MA 02215, abrugos@bu.edu), Stefanie Shattuck-Hufnagel (MIT, Cambridge, MA 02139), and Nanette Veilleux (Simmons College, Boston, MA 02115)

A production task was designed to elicit a specific intonation contour, characterized by an exaggerated rise-fall-rise in $f0$ ($L+H^*L-H\%$ in ToBI terms), conveying an attitude of ''dismayed surprise.'' Speakers were given models, a short practice period, and a preceding context for each utterance. A targetlike rise-fall-rise contour (including $H^*L-H\%$ variants) was successfully elicited for each of the contexts, but in less than half the utterances: Of 1134 utterances, 631 (56%) were judged by a panel of 3 experienced ToBI labelers as different from the intended contour, e.g., $L^*H-H\%$. Of these, 261 were produced with contours perceptually similar to the target, with a sustained flat or minimally rising $f0$ after the fall from the pitch accent, or with the rise occurring early in the next phrase. Alignment characteristics and individual speaker data may determine whether these are one contour with a continuum of realizations, or separate contours. These results (a) suggest that speakers have more than one way of signaling a given attitude and resist being limited to using only one, and (b) support the need for careful postexperiment transcription to characterize which contours subjects have actually produced. [Work supported by NSF 0345627 and NIH DC00075.]

2pSC10. Factors affecting phonetic variation of American English /k/. Stefanie Shattuck-Hufnagel (Speech Group, Res. Lab. of Electron., MIT, Cambridge, MA 02139) and Nanette Veilleux (Simmons College, Boston, MA 02115)

Word-final consonant reduction is commonly observed in American English, but variation in word-initial consonants is less thoroughly studied. Several considerations suggest that initial consonants might resist reduction; a more nuanced hypothesis is that initial consonants resist reduction in prosodically strong positions (i.e., at phrase edges and pitch accents) but may be reduced in weaker positions (e.g., phrase medially, in nonaccented syllables), particularly in un-self-conscious speech. A corpus of 16 spontaneous dialogues (8 speakers), prosodically labeled using the ToBI transcription system, provides a testbed for this view: the speakers were pairs of close friends engaged in an absorbing task which distracted them from paying close attention to their speech. Preliminary inspection of initial /k/'s from 3 speakers revealed a range of pronunciations (as described in earlier work), from normal prevocalic /k/ (with a silent closure period followed by release noise), to ''leaky'' /k/'s with some noise during the closure period and higher-amplitude release noise, to fricated /k/'s with

constant-amplitude noise throughout, and offered some support for reduction in weak prosodic contexts. Further analysis will compare prosodic context with other factors governing phonetic variation (e.g., word frequency); implications for speech recognition and production will be discussed. [Work supported by NIH DC00075 and DC02978.]

2pSC11. Errors and strategy shifts in speech production indicate multiple levels of representation. Bryan Gick (Dept. of Linguist., Univ. of BC, E270-1866 Main Mall, Vancouver, BC V6T1Z1 Canada)

An earlier pilot study [Gick, J. Acoust. Soc. Am. **112**, 2416 (2002)] observed that speakers producing challenging sequences of articulatory movements exhibit categorical shifts in production strategies. For example, a speaker may initiate all two-flap sequences with an upward tongue movement, then shift to a downward movement, then shift again. The present study attempts to determine how these strategy shifts can inform phonologists as to levels of representation. Ten subjects produced sequences of flap allophones of English /t/ and /d/ in an experimental paradigm similar to the pilot study. Results show strategy shifts in the speech of 7/10 subjects, such that: (1) shifts are more likely to occur following speech errors; (2) errors are more likely to occur in sequences with more flaps and more conflicts; (3) errors in response to articulatory conflicts occur at multiple levels of representation (segmental, lexical, etc.); (4) strategy shifts occur at multiple levels of representation (subsegmental, segmental, whole word); (5) word frequency does not affect strategy shifts. These results suggest that speech processing is active in parallel at multiple levels associated with phonological representation. [Work supported by NSERC.]

2pSC12. 3-D acoustic analysis based on magnetic resonance imaging of a "retroflex" and "bunched" American English rhotic sound. Xinhui Zhou, Carol Espy-Wilson (Dept. of Elec. and Comput. Eng., Univ. of Maryland, 2205 AV Williams Bldg., College Park, MD 20742, espy@glue.umd.edu), Mark Tiede (Haskins Labs, New Haven, CT 06511), and Suzanne Boyce (Univ. of Cincinnati, Cincinnati, OH 45202)

A variety of articulatory configurations can produce an American English /r/. Common across all configurations is a large front-cavity volume that results directly or indirectly in the characteristically low third formant. Earlier work [Zhang *et al.*, ICASSP2005 1, pp. 893–896] using 2-D finite-element modeling of the front cavity with stylized bunched and retroflexed shapes showed that, even at 2000 Hz, cross modes are generated in the front cavity. Thus, the front cavity should be modeled as a single tube with a side branch, regardless of whether there is space underneath the tongue. This paper extends the earlier work by (a) considering the full 3-D vocal tract geometry reconstructed from magnetic resonance images, (b) performing 3-D finite-element analysis, and (c) using contrasting tongue shapes from similar-sized subjects. Harmonic analysis showed the frequency response is comparable to the spectrum of the acoustic signal, suggesting good agreement between the actual and reconstructed vocal tracts. As before, wave propagation in the front cavity is not planar. 3-D vocal tract analysis shows that the cross modes have 3-D properties. Application of the 3-D wave propagation property to area function extraction and simple-tube modeling of the vocal tract will be discussed. [Research supported by NIH.]

2pSC13. Syllable structure effects on velum-oral coordination evaluated with real-time MRI. Stephen Tobin, Dani Byrd (USC Linguist., 3601 Watt Way, GFS 301, Los Angeles, CA 90089-1693, dbyrd@usc.edu), Erik Bresch, and Shrikanth Narayanan (USC Los Angeles, CA 90089)

Seminal work by Krakow (PhD. Diss., Yale University, 1989, Phonetics and Phonology **5**, 87–116, 1993) has indicated that intrasegmental coordination of articulatory gestures varies systematically as a function of syllable position. Specifically, onset and coda nasals display different lag times between the velum and oral gestural components. In onset nasals, velum lowering is achieved roughly synchronously with oral constriction, while in codas, velum lowering precedes oral constriction. However, Krakow's study was limited by its reliance on indirect measures of velum movement. Nor did her work examine nasals other than [m] or syllable organizations other than singleton onsets and codas. Our study employs real-time magnetic resonance imaging [Narayanan *et al.*, J. Acoust. Soc. Am. **115**, 1771–1776 (2004); Mathiak *et al.*, Int. J. of Lang. & Comm. Disorders **35**, 419–425 (2000)] to acquire direct information about the key vocal tract variables of velum aperture and oral constriction, thereby allowing the calculation of relative gestural timing. We investigate the effects of syllabification on interarticulatory timing in [n], and, in addition to examining codas and onsets, further consider juncture geminates spanning word boundaries. Preliminary real-time MRI results from one of three anticipated subjects do find the predicted timing difference for [n]. Further, we find that the relative timing in juncture geminates mirrors that of coda [n] but presents a velum aperture intermediate between coda and onset [n]. [Work supported by NIH.]

2pSC14. Adaptation of speech production to palatal perturbation: Kinematic analysis using electromagnetometry. Douglas M. Shiller (School of Commun. Sci. and Disord., McGill Univ., 1266 Pine Ave. West, Montreal, QC, Canada H3G 1A8, doug.shiller@mail.mcgill.ca), Mark K. Tiede (Haskins Labs., New Haven, CT), and Vincent L. Gracco (McGill Univ., Montreal, QC, Canada)

The extent to which adults are able to flexibly adapt their speech production to altered sensory-motor conditions has considerable implications for oral-motor rehabilitation, yet it remains poorly understood. Previous studies have explored the effects of different physical and perceptual perturbations on speech production, yielding mixed results with respect to the timecourse and extent of adaptation. A major limitation of these studies has been their reliance on acoustical measures as the sole index of motor adaptation. The difficulty with such an approach is the potentially complex relationship between acoustic changes and their underlying articulatory patterns. In the present study, speech adaptation to a palatal perturbation (5-mm thickened alveolar ridge) was investigated by directly measuring articulatory kinematics using electromagnetometry. In contrast to previous claims, subjects were found to produce a significant change in speech-related tongue movement immediately upon insertion and removal of the artificial palate. In addition, subjects gradually fine-tuned their tongue position with continued practice. The observed patterns are consistent with the idea of two independent adaptation processes: a rapid adaptation process that is driven by the mere presence of the false palate and a slower adaptation process driven by the continued presence of the palatal perturbation.

2pSC15. Production of American English word-initial stops in the vicinity of a voiceless stop. Elina Eydlin (Dept. of Speech-Lang. Pathol. and Audiol., New York Univ., 719 Broadway, Ste. 200, New York, NY 10003) and Nassima Abdelli-Beruh (New York Univ., New York, NY 10003)

This study examined how native-American English speakers produced the stop voicing distinction when word-initial stops were positioned before a voiceless stop (/t/) in the sentence repeat CVC ten times. Sentence durations, durations of vowel (/i/) before word-initial stops, number of produced stop closures (/t/, /c/), durations of voicing in closures and VOT were measured. Preliminary analyses on three speakers show that they produced only one closure in most cases. Speakers occasionally produced two closures when slowing down their speaking rate. Vowels preceding /b, d, g/ were barely longer than vowels before /p, t, k/. When speakers produced one closure, closures were occupied by voicing for both /b, d, g/ and /p, t, k/. However, the average durations of voicing in closure were consistently longer for /b, d, g/ than for /p, t, k/. In addition, /b, d, g/ were produced in most cases with short-lag VOT, except when there were two closures, in which case, /b, d, g/ were produced with lead VOT. All /p, t,

k/ were produced with VOT greater than 30 ms. Articulatory studies are needed to assess how the sequence of stops is produced and whether the gestures for /t/ are completely or partially eliminated.

2pSC16. Modeling the effects of the lower airway on vowel spectra.
Steven M. Lulich (MIT Speech Commun. Group, 77 Massachusetts Ave., MIT Bldg. 36-511, Cambridge, MA 02139, lulich@mit.edu)

Computational models of the acoustics of the lower airway (below the glottis) have been implemented by researchers in the speech and respiratory sciences. In general, these models are symmetrical and are studied independent of acoustic coupling to the vocal tract. We have constructed a new model that is asymmetrical and based on human morphometric data. The acoustic properties of the model are explored and compared with symmetrical models, and the effects of the model on vowel spectra are studied for varying amounts of lower airway and vocal-tract coupling via the glottis. Cochlear excitation patterns are also computed for the vowel spectra, and the effects of the lower airway acoustics on these excitation patterns are discussed.

2pSC17. Effect of tone on voice onset time in laryngeal and esophageal speech of Mandarin.
Manwa Ng (Commun. Sci. and Disord., Long Island Univ., Brookville, NY 11548-1300) and Hanjun Liu (Northwestern Univ., Evanston, IL 60208)

Defined as the time between the release of burst of a stop and the onset of the following vowel, voice onset time (VOT) directly indicates the interarticulator timing between the voicing source and the articulatory movement in the vocal tract. A number of studies have investigated the VOT characteristics of different languages. Yet, few have studied how VOT is affected by tone level in a tone language, and by the type of phonation. The present study investigated the possible effect of tone level and phonation type on such interarticulator timing characteristics. VOT values were measured from the syllables /$p^h a$/, /$t^h a$/, and /$k^h a$/ produced at four Mandarin tone levels by eight laryngeal (NL) and seven esophageal (SE) speakers who were native speakers of Mandarin. Results indicated that significant differences in VOT were found in the NL speaker group, but not in the SE speaker group. With respect to phonation type, SE speakers showed significantly shorter VOT values than NL speakers. Such differences may be related to the use of pharyngoesophageal (PE) segment as a new sound source. SE speakers appear to take a shorter time to start PE segment vibration as compared to NL speakers.

2pSC18. Intrinsic vowel pitch in tracheo-esophageal speech.
Michael Kiefte, Natalie Didrachnik, and Jillian Roswell (School of Human Commun. Disord., Dalhousie Univ., 5599 Fenwick St., Halifax, NS B3H 1R2 Canada)

The tendency for high vowels to have higher fundamental frequency (f_0) than low vowels, or intrinsic vowel pitch (IF_0), has been found in both laryngeal and esophageal speech [e.g., Gandour and Weinberg, Phonetica **25**, 140–164 (1980)]. In the present study, eight tracheo-esophageal speakers were asked to produce five repetitions of each of the four point vowels and f_0 was measured. Results showed significantly higher f_0 for high vowels than for low vowels, consistent with previous findings for laryngeal and esophageal speakers. In addition, f_0 was measured from two laryngectomees over a period of 4 months immediately following surgery to determine whether IF_0 is produced automatically or whether it develops over time. Neither of these subjects produced differences in IF_0 at any time within the 4-month period. These preliminary data appear to indicate that IF_0 is not an automatic consequence of either speech physiology or acoustics. More data from additional subjects will be presented. [Work supported by SSHRC.]

2pSC19. Computational aeroacoustics of flow through static model of human vocal tract.
Jungsoo Suh, Steven H. Frankel, Luc Mongeau, and Michael W. Plesniak (School of Mech. Eng., Purdue Univ., West Lafayette, IN 47907)

Computational aeroacoustics studies of flow through the human vocal tract, here modeled as a planar channel with an orifice, hence referred to as the glottis, were conducted using large-eddy simulation (LES). Comparisons were made between LES predictions and experimental wall pressure measurements and particle-imaging-velocimetry flow fields. Rigid models of both converging and diverging glottal passages, each featuring a 20-deg included angle and a minimal glottal width of 0.04 cm with transglottal pressure of 15 cm H_2O, were studied. The compressible Navier-Stokes equations were accurately and efficiently integrated for the low Mach number flow through the use of an additive semi-implicit Runge-Kutta method and high-order compact finite-difference schemes for spatial discretization. Characteristic-based nonreflecting boundary conditions were used together with an exit zone in the context of a multiblock approach. Asymmetry of the flow due to a Coanda effect, and transition to turbulence were observed. An acoustic analogy based on the Ffowcs Williams-Hawkings equation was applied to decompose the near-field acoustic source into its monopole, dipole, and quadrupole contributions to assess glottal geometry effects on far-field sound. The results showed that dipole sources due to the unsteady forces exerted on the duct wall are dominant. [Work funded by NIH Grant RO1 DC03577.]

2pSC20. Modal analysis of a time-varying vocal tract with losses: A comparison of left and right eigenfunctions.
Gordon J. Ramsay (Haskins Labs., 300 George St., New Haven, CT 06511-6624)

Sound propagation during speech can be modeled by a linear wave operator describing the spatiotemporal evolution of acoustic pressure and velocity distributions on a bounded domain representing the interior of the vocal tract. The eigenvalues of the wave operator determine the formant frequencies and bandwidths. The right eigenfunctions describe the standing wave patterns associated with each formant, whereas the left eigenfunctions describe how an arbitrary source distribution projects onto each formant. In speech research, it has always been implicitly assumed that left and right eigenfunctions are identical. This only holds if the wave operator is self-adjoint, which cannot be true when losses due to viscous damping, radiation effects, and wall vibration are taken into account. To examine this issue, a time-domain finite-volume simulation of acoustic wave propagation in the vocal tract was developed, including the above loss mechanisms. Eigenvalues and eigenfunctions were calculated from the resulting state-space recursion, for time-varying geometries representing VCV sequences containing fricatives and stops. Left and right eigenfunctions are similar in shape, but there are differences in the zero-crossing points and the amplitudes of the different lobes. Predictions based on right eigenfunctions alone will be incorrect. [Work supported by NIH.]

2pSC21. The interaction of linguistic and affective prosody in a tone language.
Chinar Dara and Marc D. Pell (Commun. Sci. and Disord., McGill Univ., 1266 Pine Ave. West, Montreal, PQ H3G 1A8, Canada, chinar.dara@mail.mcgill.ca)

To address how a common set of acoustic properties of speech prosody modulate to convey linguistic and affective meanings concurrently, this study investigated the influence of phonemic tones on the expression of emotion (happy, sad, angry) and linguistic modality (declarative, interrogative) in a tone language, Punjabi. Base stimuli consisted of neutral sentences with either of the three contrastive tones in Punjabi (falling, level, high-rising) varying at the object position of the sentence only. Each of these sentences was elicited as a statement or a question for each of the three emotions by 6 native Punjabi speakers. Utterances were validated for adequate representation of the target emotion, and reliable exemplars for each of the emotions and linguistic modality were subjected to detailed acoustic analysis. Fundamental frequency and duration measures were analyzed for the stressed vowel of the keywords (subject, object, verb) and

the whole utterance for each of the tokens. Results demonstrated ways that the acoustic correlates of tone at the word level interact with that of the intonation and emotion at the sentence level, which were further compared with previous results on the effect of focus on intonation and emotional prosody [M. D. Pell, J. Acoust. Soc. Am. **109**, 1668–1680 (2001)].

2pSC22. The scope of effect of prosodic boundaries in articulation. Jelena Krivokapic (Dept. of Linguist., USC, 3601 Watt Way, GFS 301, Los Angeles, CA 90089-1693)

An articulatory study of the scope of prosodic boundary effects across boundaries of varying strengths is presented. Sentences in each of three prosodic conditions contain the following string: C1VC2VC3#VC4VC5VC6, where C is an alveolar consonant, V is a vowel, and # is a prosodic boundary. The boundaries are of three degrees of strength. Using articulator movement-tracking data (EMA) for the tongue tip articulator, consonant constriction formation and release duration, acceleration duration, and spatial magnitude are measured. Leftward and rightward temporal effects of the boundary on the consonants are investigated. Based on earlier studies (e.g. Cambier-Langeveld, Linguistics in the Netherlands, 13–24, 1997; Shattuck-Hufnagel and Turk, Proceedings 16th International congress on Acoustics and 135th Meeting Acoustical Society of America, 1235–1236, 1998; Berkovits, J. of Phonetics **21**, 479–489, 1993) and predictions of the prosodic gestural model (Byrd and Saltzman, J. of Phonetics, **31**, 149–180, 2003), it is expected that (1) the amount of articulatory lengthening will increase with boundary strength and (2) the degree of lengthening will decrease with distance from the boundary. Three subjects participated. The results show that at the boundary, gestures have longer constriction and acceleration durations, and the effect of the boundary increases with boundary strength, distinguishing up to three levels of boundary strength. The effect is local and diminishes with distance from the boundary. These results support the initial hypotheses. [Work supported by NIH.]

2pSC23. Acoustic and fiberscopic studies of the voice characteristics of school teachers under vocal abuse conditions with the loud voices of children. Noriko Kobayashi (School of Allied Health Sci., Kitsato Univ., 1-15-1 Kitasato, Sagamihara 228-8555 Japan, noriko@ahs.kitasato-u.ac.jp, Takashi Masaki, and Koichiro Nishiyama (Kitasato Univ, Sagamihara 228-8555, Japan)

It has been known that the incidence of vocal pathology was high in school teachers as they had to use a great amount of voice and loud phonation in noisy school settings with the children's loud voices. A proper vocal hygiene program and voice therapy are effective treatments for these cases. The final goal of the treatment should be to obtain the skills to use effective and healthy phonation even in noisy environments. In our study, three school teachers with vocal nodules, three speech therapists who were trained to produce nonconstricted voice even in noisy environments, and ten college students with no laryngeal pathology (control group) spoke eight sentences with one target word in two environments: a quiet environment and one with meaningful multi-talker babble (MMB). Acoustic analyses and fiberscopic examination revealed higher sound pressure and F0 levels and laryngeal constriction in MMB for the teachers before therapy and the normal talkers, but two speech therapists kept similar F0 levels and laryngeal conditions in both environments. After voice therapy, teachers produced less constricted voice under noise. These results suggested the efficacy of voice therapy with MMB for teachers to obtain effective and healthy voice in their noisy work environments.

2pSC24. Realizing question intonation in Mandarin with neutral tone. Fang Liu (Dept. of Linguist., Univ. of Chicago, 1010 E. 59th St., Chicago, IL 60637, liufang@uchicago.edu) and Yi Xu (Univ. College London, London, NW1 2HE, UK)

Previous research has found that postfocus pitch range is lowered in Mandarin even in questions, although the amount of lowering is smaller than in statements. This study investigates whether postfocus pitch range is raised in questions where the postfocus words carry the neutral tone, which, being conventionally considered targetless, may be better suited for manifesting intonation. Eight native speakers of Mandarin each produced 80 statements and 80 questions. The sentences ended either with a sequence of 5 neutral tones or with 3 neutral tones followed by 2 high tones. The sentences had two alternative focus locations, one on the full tone (high, rising, low, or falling) immediately preceding the neutral tone sequence, and the other on the low tone before the full tone. Results show that (1) postfocus lowering occurred in the neutral-tone-final sentences in both statements and questions, just as in the full-tone-final sentences, and (2) sentence-final neutral tone had falling F0 even in questions, thus contrasting with sentence-final high tone, which had rising F0 in questions. The findings are interpreted as indication that the neutral tone, being associated with weak articulatory effort, is not as effective as full tones in realizing the statement/question contrast.

2pSC25. Acoustic and articulatory correlates of contrastive focus. Melanie Thibeault, Lucie Menard, Annie Leclerc, Anthony Calabrino (Laboratoire de phonetique, UQAM, Case postale 8888, succursale Ctr. ville, Montreal QC, Canada, H3C 3P8, thibeault.melanie.2@courrier.uqam.ca), and Annie Brasseur (UQAM, Montreal QC, Canada, H3C 3P8)

This paper aims at examining production of French speakers in marking focus-induced prominence at the articulatory and acoustic levels. The corpus consisted of CVC syllables, where C corresponds to one of the stops /p t k/ and V is one of the vowels /i a u/. Target words were embedded in carrier sentences elicited in two prosodic conditions: neutral (unfocused) and under contrastive focus. Four adult speakers (all native speakers of French) pronounced ten repetitions of each sequence. The audio signal and tongue shapes were recorded using a digital camera and a SONOSITE 180 ultrasound. Formant frequencies, rms values, duration, and tongue contours corresponding to each vowel were extracted. Analyses show that prosodic context has a significant effect on acoustic and articulatory data for all vowels, increasing F1 and F2 under contrastive focus, compared to the neutral context. However, this stable acoustic pattern is achieved by various articulatory strategies across subjects. For example, 2 subjects produced /u/ under focus with a lower tongue body than in the neutral context, whereas the remaining 2 subjects had a higher tongue body in focused syllables. Results are compared to previous studies on articulatory and acoustic correlates of prosodic structure in French and English.

2pSC26. Rhythm in English clear speech. Rajka Smiljanic, Josh Viau, and Ann Bradlow (Linguist., Northwestern Univ., 2016 Sheridan Rd., Evanston, IL 60208, rajka@northwestern.edu)

This study investigates the effect of hyperarticulated, intelligibility-enhancing clear speech on English speech rhythm. Ramus et al. [Ramus et al., "Correlates of linguistic rhythm in the speech signal," Cognition **72**, 265–292 (1999)] showed that temporal properties of a speech signal such as the percentage of vocalic intervals (%V) and variability of consonantal and vocalic intervals (C, V) can be related to phonological properties such as presence/absence of unstressed vowel reduction and syllable structure complexity, and are consequently quite successful at grouping languages into the traditional rhythmic classes (stress-, syllable-, and mora-timed). Here, we explore whether/how clear speech affects stress-timed characteristics of English sentences in terms of these measures. Results revealed that the proportion of vocalic intervals (%V) within sentences remained stable across speaking styles, i.e., consonants and vowels

were lengthened equally in clear speech. However, variability of both vocalic and consonantal intervals (V, C) was higher in clear than in conversational speech. The increase in V and C was achieved primarily through insertion/strengthening of short vocalic and consonantal segments that were dropped or coarticulated with surrounding sounds in conversational speech, making word and syllable boundaries more salient. These results suggest that increased intelligibility of clear speech can in part be attributed to prosodic structure enhancement by means of enhanced syllable and word boundaries demarcation.

2pSC27. Sensitivity of Mandarin speech to loudness-shifted voice feedback perturbation. Hanjun Liu, Qianru Zhang, Charles R. Larson (Northwestern Univ., 2240 Campus Dr., Evanston, IL 60208), and Yi Xu (Univ. College London, London, NW1 2HE, UK)

Recent studies have demonstrated that loudness-shifted voice auditory feedback leads to compensatory adjustments in voice amplitude during production of vowels. In this study we asked the question of whether Mandarin-speaking subjects would respond to loudness-shifted voice feedback during speech. Twenty-two native-speaking subjects with normal hearing and neurological background served as subjects. They said the Mandarin phrase ba ma, with the high tone on both syllables, as answers to three randomized prerecorded questions. The questions induced focus on the first or the second syllable, or on both syllables. During each production a ± 3 dB, 200- or 400-ms perturbation in voice feedback was presented 250 ms following voice onset. Randomized control trials were intermixed with test trials. Digitized voice records were converted to an rms trace representing voice amplitude. Traces were aligned with stimulus onset for event-related averaging. Results demonstrated that subjects responded to the loudness perturbation with compensatory adjustments in voice amplitude. The data extend recent findings indicating that voice feedback is used to help control amplitude during speech. [This work was supported by NIH Grant No. DC006243-01A1.]

2pSC28. Gradient alternations and gradient attraction. Katherine Crosswhite (Dept. of Linguist., Rice Univ., 6100 Main St., Houston, TX 77005, crosswhi@rice.edu)

Many phonological alternations have been shown to be sensitive to lexical "gradient attraction," defined by Burzio ["Surface-to-surface morphology: When your representations turn into constraints," Rutgers Optimality Archive #341-0999 (http://roa.rutgers.edu) (1998)] as follows: (A) The overall structure of a word W (in both phonological and semantic components) is influenced by that of other words in the lexicon to which

W is independently similar, and which are thought of as "attractors" of W. (B) Attraction is stronger where independent similarity is greater. Since most strong attractors will be morphologically related to W, this results in a limitation on phonological alternations that would produce allomorphy. So far, gradient attraction has been studied for categorical alternations, such as stress placement in English (cf. compArable versus cOmparable). However, many alternations previously believed to be categorical are in fact gradient [as reviewed in, e.g., Port and Leary, "Against formal phonology," Language **81**, 927–964 (2005)]. This study documents the influence of lexical attraction on two gradient phenomena of English: polysyllabic shortening and flapping. Attractor strength was varied in two manners. First, each base used in the study was paired with both derivational and inflectional derivatives (i.e., rose-roses-rosy for polysyllabic shortening, and wait-waiting-waiter for flapping), and second by analyzing the effect of between-item variation in semantic relatedness for the derivationally related forms (cf. ice-icy versus nose-nosy).

2pSC29. Resonance tuning in soprano singing and vocal tract shaping: Comparison of sung and spoken vowels. Shrikanth Narayanan, Erik Bresch (USC Viterbi School of Eng., 3710 S. McClintock Ave, RTH 320, Los Angeles, CA 90089, shri@sipi.usc.edu), Stephen Tobin, Dani Byrd, Krishna Nayak, and Jon Nielsen (USC, Los Angeles, CA 90089)

Singing at high pitches by sopranos involves complex orchestration of vocal-fold mechanisms and vocal-tract shaping. A specific phenomenon is the tuning of vocal-tract resonances to the fundamental at high pitches. Measurement of vocal-tract resonances has shown that this is particularly true for the first resonance [Joliveau *et al.* (2004)]. Higher resonances show some increase in their values perhaps reflecting incidental effects rather than active tuning. We investigate vocal-tract shaping in soprano singing using a real-time MRI technique [Narayanan *et al.* (2004)] with synchronized noise-cancelled audio recording. A professionally trained Western-classical singer spoke and sang the vowels /a, i, u, o, e/ preceded by the consonant /l/ over an ascending scale (from B-flat below middle C, rising two octaves). Acoustic analysis shows clear evidence of F1 tuning to F0 in the higher-octave singing of the vowels. The vocal-tract shape was distinctly different at the higher pitches compared to vowels spoken at normal pitch and those sung at the lower octave, with fairly large oral and pharyngeal cavities, created by a highly bunched and retracted tongue, and a raised larynx. At the highest pitch, the vocal-tract shape differences between the vowels are considerably lessened, although some vestigial canonical tongue shapes appear to remain. [Work supported by NIH, USC Annenberg Center.]

2p TUE. PM

Session 2pSP

Signal Processing in Acoustics: Topics in Seismic Signal Processing

Alan W. Meyer, Cochair
665 Dunhill Dr., Danville, CA 94506

Max Deffenbaugh, Cochair
Exxon Mobile Research and Engineering Co., 1545 Route 22 East, Annandale, NJ 08801

Chair's Introduction—1:00

Invited Papers

1:05

2pSP1. Short-term non-poissonian temporal clustering of magnitude 4+ earthquakes in california and western nevada. John E. Ebel (Weston Observatory, Boston College, 381 Concord Rd., Weston, MA 02493, ebel@bc.edu), Daniel W. Chambers, Alan L. Kafka, and Jenny A. Baglivo (Boston College, Chestnut Hill, MA 02467)

The M4+ mainshocks throughout California and western Nevada from 1932 to 2004 show non-Poissonian temporal clustering over time periods of a few days. The short-term clustering is independent of the distance between earthquake epicenters. It implies that some of the M4+ mainshocks are mutually triggered by some unknown regional cause. In southern California, more short-term clustering is found for M4+ earthquakes east of the San Andreas Fault. In central California, most M4+ mainshocks at Long Valley, CA have occurred within 10 days of M4+ mainshocks around the San Francisco Bay area. The clustering implies predictable behavior in the occurrences of M4+ mainshocks. We propose a hidden Markov model (HMM) as an earthquake forecast method for the region. Our HMM assumes a hidden sequence of interevent time states associated with observations of earthquake occurrences (times, locations, and magnitudes) with transition probabilities between states determined with the Baum-Welch algorithm and the past earthquake data. Given the seismic history up to the latest earthquake, the probability of another earthquake within the next few days is estimated. Tests of our HMM with two, three, and four temporal states show some modest success. We plan to extend the model to forecast magnitude and spatial parameters.

1:25

2pSP2. Trends in detection in earthquake seismology. David B. Harris (Lawrence Livermore Natl. Lab., P.O. Box 808 L-205, Livermore, CA 94551, harris2@llnl.gov)

Detection methods in earthquake seismology have been dominated in the past by simple energy detectors, often combined with array processing (beamforming) to detect at lower thresholds. Increasingly, correlation detectors now are being employed to exploit the additional gain possible with coherent detection methods. Coherent detection of transient seismic signals is complicated by the fact that earthquake signals generally cannot be predicted with certainty in advance, except for those cases where observations of previous events are available. One approach to solving this problem is to employ subspace detectors that allow variation in the correlation template to match the range of signals anticipated from a specific source region. Subspace templates have been produced empirically from past observations of repeating sources. The current challenge is to compute the templates in areas without past event observations. One approach to doing so calculates the target signal space by forward simulation through many earth realizations drawn from a stochastic earth model. In this talk, we present the results of empirical subspace detection with arrays of sensors (combining array gain with coherent matched-signal gain) and discuss preliminary results from the use of computed subspaces. [This work was performed under the auspices of the U.S. Department of Energy by the University of California, Lawrence Livermore National Laboratory under Contract No. W-7405-Eng-48.]

1:45

2pSP3. Electromagnetic-to-seismic conversion: A new direct hydrocarbon indicator. A. H. Thompson (Retired, 13602 Peachwood Ct., Houston TX 77077)

In electroseismic (ES) exploration, electromagnetic (EM) energy converts to seismic energy at depth in the Earth. The amplitude of the converted seismic wave is large at porous, permeable, resistive bodies such as hydrocarbon reservoirs. ES is the only known exploration tool that, applied from the surface of the Earth, may discriminate between water, gas, and oil in the pores of reservoir rock. ES measures rock electrical properties with seismic resolution. The seismic conversion may be linear, at the same frequency as the EM stimulus. Electrokinetic effects, involving the relative motion of the rock solids and fluids, describe linear ES well. Nonlinear conversion may create all harmonics of the EM stimulus. Conventional descriptions of ES and SE conversion do not account for nonlinear conversions. New models, that couple EM and seismic energies through modulation of electrochemical potentials in the Earth, do explain the observations. ES field tests detected the hydrocarbon-sensitive, linear response to 1000 m and the nonlinear response to 1500 m deep. The results suggest that greater depth penetration, to 7000 m, is feasible.

2:05

2pSP4. Waveform design for electroseismic exploration. Scott C. Hornbostel, Arthur H. Thompson, and Warren S. Ross (ExxonMobil Upstream Res. Co., P.O. Box 2189, Houston, TX 77252-2189, scott.c.hornbostel@exxonmobil.com)

Electroseismic (ES) exploration remotely identifies the presence of hydrocarbons using the conversion of electromagnetic energy to seismic energy. These conversions are relatively larger in a porous, permeable resistive body (such as an oil or gas reservoir) when compared with background conversions. Typical ES signals, however, may be several orders of magnitude below the expected ambient noise levels, thus requiring repetitions of long source sequences. In addition, the ES signals may vary linearly or nonlinearly with the input current. Two classes of coded waveforms are presented for the detection of these linear and nonlinear conversions. The linear conversions are detected using a Golay sequence pair with side lobes that cancel when the correlated pair is summed. A set of pseudo-random binary sequences (PRBSs) can be modified to detect a nonlinear squared response with minimal side lobes while removing undesired linear conversions. These source waveforms are implemented in the power waveform synthesizer (PWS) that is capable of switching three-phase power to create the specified waveforms. The PWS is an important part of the overall electroseismic exploration system that includes source electrodes, pickup-free receivers, and acquisition and processing features aimed at effective detection of small ES signals.

Contributed Papers

2:25

2pSP5. Small-scale geophysics inversion using surface wave extracted from noise cross correlation. Pierre Gouedard, Philippe Roux, and Michel Campillo (LGIT—BP 53—38041 Grenoble cedex 9, France, pierre.gouedard@ujf-grenoble.fr)

Green's functions between receivers can be retrieved from the correlation of ambient noise or with an appropriate distribution of sources. We demonstrate this principle in geophysics at small scales. We used a 22.5-m-long line of 16 geophones and recorded noise signals in the bandwidth of 5 to 150 Hz. Noise was generated by human steps for a 10-min walk in the alignment of the geophones line (5 min on each side). The correlations of the records gave us the Green's functions associated with each pair of geophones, which exhibit two Rayleigh modes. We applied a frequency wave number transformation to separate those two modes and we measured the dispersion curve of each mode. We used these dispersion curves to perform an inversion and obtain the 1D velocity profile of the near surface (about 10 m deep).

2:40

2pSP6. Passive correlation imaging of a buried scatterer. Oleg I. Lobkis, Eric Larose, and Richard L. Weaver (Dept. of Theoretical and Appl. Mech., Univ. of Illinois, 104 S. Wright St., Urbana, IL 61801)

Waveforms obtained by correlation of diffuse fields are now widely used to estimate times of flight and ballistic Green's functions. Theory, however, predicts that such passively obtained waveforms should equal the full Green's function of a medium, and thereby convey information on the details of a structure and the presence of isolated scatterers. Here, we investigate diffuse field correlations in a reverberant elastic body and demonstrate that their shear and Rayleigh waves allow to image internal and surface features. Resolution in the resulting images is comparable to standard echographic and seismic images. The experimental setup is analogous to a real seismic experiment.

2:55–3:10   Break

Invited Papers

3:10

2pSP7. Anisotropic seismic waves in systems with fluid-saturated fractures. James Berryman (Lawrence Livermore Natl. Lab., P.O. Box 808 L-200, Livermore, CA 94551-9900, berryman1@llnl.gov)

The seismic wave speeds observed in exploration reflection surveys of oil and/or gas reservoirs may be dominated by the elastic properties in the dry state of very flat cracks or fractures, containing relatively little porosity. In contrast, the majority of work on effective medium theories (both estimates and bounds) has concentrated on systems whose elastic behavior depends on the volume fractions of the fluid constituents. Although a considerable amount of work has already been done in this area, there remain controversies regarding the use of the differential and self-consistent schemes for property estimation in these systems, and these controversies are known and well documented. When the reservoirs contain vertical fractures, these systems exhibit anisotropic seismic wave speeds. Methods are known that permit relatively simple calculation of these anisotropic properties from theoretical estimates made on randomly isotropic systems of fractures. When these methods are appropriately combined with weak anisotropy approximations associated with the Thomsen seismic-wave parameters, it is possible (at least in principle) to estimate fracture orientations and densities. The relationships between these forward and inverse problems will be treated and discussed with special emphasis on the question of uniqueness of the resulting interpretations.

3:30

2pSP8. Finite-frequency seismic tomography. Yang Shen (Grad. School of Oceanogr., Univ. of Rhode Island, Narragansett, RI 02882)

The use of 3-D sensitivity kernels in tomographic inversions has received much attention in the seismological community. Conventional tomographic studies are based on ray theory, which assumes that the arrival time of a body wave depends only on the velocity along the ray path between the source and receiver. Observed seismic waves are finite frequency signals, and their travel times are sensitive to a 3-D volume around the geometrical ray and subjected to wavefront healing, scattering, and other diffractive effects. As a result, seismic waves passing through velocity heterogeneities with dimensions smaller than the width of the Fresnel zone undergo significant wave-front healing, which results in reduced travel-time shifts compared to the values predicted by ray theory. 3-D sensitivity kernels take into account the effects of wave diffraction and scattering. This presentation will provide a brief review of

2p TUE. PM

finite-frequency tomography, which is in a stage of rapid development. With the advancement of computer technology, it has become increasingly practical to construct 3D sensitivity kernels based on 3D reference models using full waveform simulations. This holds the promise of a new level of the integration and joint interpretation of finite-frequency waves propagating in complex media.

3:50

2pSP9. High-resolution imaging of fault zone structures. Zhigang Peng and Yehuda Ben-Zion (Dept. of Earth and Space Sci., Univ. of California, 595 Charles E. Young Dr. East, Los Angeles, CA 90095, zpeng@ess.ucla.edu)

Damaged fault zone (FZ) rocks with high crack density are expected to produce several indicative wave propagation signals. These include scattering, anisotropy, nonlinearity, and FZ guided head and trapped waves. These signals can be used to obtain high-resolution imaging of the FZ structures and to track possible temporal evolution of FZ properties. Results associated with systematic analysis of such signals recorded at several large strike-slip FZs can be summarized as follows: The observed FZ trapped waves are generated by relatively shallow structures that extend only over the top $\sim 3--4km$ of the crust. The shallow trapping structure in the North Anatolian fault (NAF) is surrounded by broader anisotropic and scattering zones that are also confined primarily to the top 3 km. Systematic analyses of anisotropy and scattering around the NAF do not show precursory temporal evolution of properties. However, the scattering results show clear co-seismic changes and post-seismic logarithmic recovery. These effects are likely to reflect mostly changes in properties of the shallow crust in response to strong shakings of nearby major earthquakes. Analyses of FZ head waves along the San Andreas Fault point to the existence of material interfaces that extend to the bottom of the seismogenic zone.

4:10

2pSP10. Multichannel inversion of scattered teleseismic waves for migration imaging and separation of noise. Stephane Rondenay and Xander Campman (MIT, 77 Massachusetts Ave., 54-512, Cambridge, MA 02109, rondenay@mit.edu)

In recent years, a substantial increase in the availability and fidelity of broadband seismographs has led to new opportunities for multichannel processing of scattered teleseismic body waves, allowing for high-resolution imaging of subsurface structure. Here, we discuss the theoretical foundation and the implementation of such a technique. The problem is posed for forward- and back-scattered wave fields generated at discontinuities in a 2-D isotropic medium, with the backprojection operator cast as a generalized radon transform (GRT). The approach allows for the treatment of incident plane waves from arbitrary backazimuths, and recovers estimates of perturbations in material property (P and S wave velocities) about a smoothly varying reference model. In its current implementation, the method can be applied to low-pass filtered data (0.3-Hz cutoff) for imaging structure in the lower crust and lithospheric mantle. Examples from the Cascadia and Alaska subduction zones are shown. Ongoing work is being conducted to render the method applicable to higher frequency data by identifying and canceling noise generated by near-surface scatterers (e.g., mountains, sedimentary basins).

Contributed Papers

4:30

2pSP11. Multichannel near-array corrections for short-period teleseismic data. Xander Campman and Stephane Rondenay (Earth Resources Lab., Massachusetts Inst. of Technol., 42 Carleton St., Cambridge, MA 02142, xander@erl.mit.edu)

The increased availability of data from broadband and short-period seismometer arrays has stimulated the development of multichannel high-resolution imaging algorithms for scattered teleseismic waves. One of the main factors limiting the resolution of teleseismic images is the interference from unwanted phases at higher frequencies (0.3–1.5 Hz). These unwanted phases are generally associated to scattering at near-surface features. We describe initial results from a multichannel inversion-based method designed to suppress scattered waves from surface topography. Such an approach is possible because the dense spatial sampling of short-period arrays allows us to sample these phases well enough to characterize them. Using a subset of the data, we first estimate a model for the distribution of near-surface scatterers. Next, we predict scattered waves in the entire data set using this scatterer distribution model. Both the inversion and prediction steps are based on an integral representation of the scattered-wave field which takes into account the near-field character of body-to-surface wave scattering. Finally, we subtract the predicted scattered waves from the entire data set. Data from the LARSE93 experiment are used to illustrate and validate the present algorithm.

4:45

2pSP12. An earthquake detection, identification, and location system for the northeastern U.S. Based on the wavelet transform. John E. Ebel (Weston Observatory, Boston College, 381 Concord Rd., Weston, MA 02493)

In order to optimize the ability of the New England Seismic Network (NESN) to detect all local earthquakes to the smallest possible magnitude, Weston Observatory of Boston College has developed an automated event detection and identification system based on the wavelet transform. This software system performs a wavelet transform on the data from each station being received and uses the time, frequency content, and energy of the first arrival, the highest energy arrival, and the end of the detection to compute the Bayesian probability that the detection was a distant earthquake (teleseism), regional earthquake, local earthquake, quarry blast, Rg wave from a quarry blast, or transient noise. At each station, these measured parameters are used to estimate the origin time, epicentral distance, and magnitude for each detection. The systems attempts to associate the detections from different stations that have a common event identification as a teleseism, regional/local earthquake, or quarry blast. If three or more stations associate, an automatic event location and magnitude (if a regional or local earthquake) are computed. This wavelet-transform detector and identifier has greatly increased the sensitivity of detecting and locating small earthquakes and quarry blasts in the New England region.

Session 2pUW

Underwater Acoustics: Scattering of Sound at the Sea Surface

Duncan P. Williams, Chair

DSTL, Physical Sciences, Winfrith Technology Centre, Dorchester, DT2 8WX, UK

Chair's Introduction—1:00

Invited Papers

1:05

2pUW1. Measurements and modeling of low-frequency air-sea-interface scattering.　Roger C. Gauss, Joseph M. Fialkowski, and Daniel Wurmser　(Naval Res. Lab., Code 7144, Washington, DC 20375-5350)

The acoustic scattering response of the undersea air-sea interface can be quite sensitive to ocean surface conditions, scattering geometry, and acoustic frequency. This paper presents both measurements and modeling of low-frequency (<1500 Hz) scattering strength. The modeling is based on a small slope approximation (SSA) that accounts for directional surface-wave roughness spectra with low-wave-number cutoffs. To provide an operationally useful predictive tool, a set of surface-wave spectral parameters (strength, exponent, and peak) are determined by nonlinear, multi-parameter fits to broadband, vertically bistatic surface backscattering strength data, leading to a semi-empirical bistatic model environmentally parameterized solely by the wind speed. The same SSA theory is also used to estimate the coherent-reflection loss given the frequency, angle, and wind speed. These broadband models are then used to explore the dependence of both incoherent and coherent sea-surface scattering on frequency, angle, and wind speed. [Work supported by ONR and PEO C4I & Space (PMW-180).]

1:25

2pUW2. How significant is the wave height of a moving surface on low-frequency propagation?　Duncan P. Williams　(Dstl. Winfrith, Winfrith Technol. Ctr., Rm. N11 Bldg. A32, Dorchester DT2 8WX, UK)

The ocean can carry low-frequency acoustic waves to surprisingly long range, but the ocean is ever changing and acoustic propagation is rarely unaffected by its interaction with the moving sea surface. The clarity of low-frequency acoustic communications at long range can be corrupted by this interaction. This is unsatisfactory for reliable communication and it is necessary to explain the effects introduced by the nature of the sea-surface scattering. This paper looks at the relationship between the power spectra of the received surface scattered signal and the sea surface. The surface can be conveniently characterized by its swell and wind-sea parts that can, in turn, be expressed as a function of their significant wave height and peak frequency. Basic properties of a surface spectral model are reviewed. A new model-based approach to computing the scattering from a moving sea surface is introduced. This approach is based on ray theory and a simple physical model for the response function of the surface. The model is compared with long-range experimental data and then results are shown to illustrate the effects of wave height and peak frequency on simple broadband waveforms.

1:45

2pUW3. Gravity wave focusing in the littoral zone.　Grant Deane　(Scripps Inst. of Oceanogr., Code 0238, UCSD, La Jolla, CA 93092-0238)

One of the unique environmental factors impacting acoustic propagation in the littoral zone is the presence of shoaling surf. Surface gravity waves act as acoustic mirrors, creating high-intensity cusps and caustics in the reflected sound field. The focal points of sound move through the water column as the gravity waves shoal, and can disrupt the performance of underwater communications systems and impact sonar performance. Analytical and numerical calculations of the focusing properties of shoaling gravity waves are described and used to predict the motion of the focal regions through time. [Work supported by ONR.]

2:05

2pUW4. A brief review of modeling in support of bubble-enhanced scattering from the air/sea interface.　Richard Keiffer (NRL-SSC, Stennis Space Ctr., MS 39529)

By 1995 or so, the long-standing disparity (at low to moderate frequencies) between sea surface (backscatter) reverberation data and interface scattering models was solved by the incorporating a bubble cloud scattering mechanism into reverberation models. At that time (and still today), a comprehensive description of the spatially and temporally varying bubble population was lacking, so critical environmental parameters needed for the bubble cloud scattering models were inferred from the differences in the scattering levels predicted by interface scattering models and measured scattering strengths. This modeling development generally ignored another mechanism through which the bubbles may influence the reverberation: the bubbles will also cause upward refraction near the air/sea interface and this upward refraction may significantly increase the scattering from the air/sea boundary. In this presentation, a brief review of the published modeling supporting the bubble-enhanced interface scattering phenomenon will be presented. [Work supported by ONR/NRL.]

2p TUE. PM

2pUW5. Wavefront modeling of scattering from surface waves. Chris Tindle (Phys. Dept., Univ. Auckland, Auckland, New Zealand)

The scattering of sound from any particular smooth realization of ocean surface waves can be calculated using wavefront modeling. The method uses ray tracing to locate the wavefronts and stationary phase analysis to determine the phase and amplitude of the sound field. Waveforms due to reflection of short pulses can be found directly and compare well with experimental results. Reflection at a curved surface leads to foci and caustics and the field is determined numerically from Pearcey and Airy functions, respectively. The method uses the time domain and is fast and accurate. [Work supported by ONR.]

2:45–3:00   Break

3:00

2pUW6. Time domain modeling of acoustic Doppler from 2-D, developing, and swell-contaminated seas displaying a bimodal azimuthal dependence. Richard Keiffer (NRL-SSC, Stennis Space Ctr., MS 39529)

The typical sea-surface acoustic scattering experiment is often modeled invoking an environment and measurement scenario containing a number of idealizations. Almost universal is the assumption that the rough pressure-release air/sea boundary can be described using a roughness spectrum that belongs to spatially uniform, fully developed seas. Often this roughness spectrum is assumed to have a simple cosine-squared or some other canonical azimuthal dependence. The air/sea boundary is assumed to be an overly homogeneous isospeed ocean and the source and receiver are assumed stationary or in uniform rectilinear motion. Because the data collected at sea and the results from numerical simulations are often compared in order to infer the acoustic attributes of bubbles and bubble clouds, it is important to have a good understanding of the limitations that these common idealizations impose on the data/model comparison. In this paper, a newly developed and tested time domain model for acoustic scattering from moving 2-D surfaces [J. Acoust. Soc. Am. **118**(3), 1283–1299 (2005)] is used to examine the implications that wave growth, swell contamination of a bimodal azimuthal dependence, and near-surface upward refraction have on the Doppler of acoustic signals backscattered from the sea surface. [Work supported by ONR/NRL.]

3:20

2pUW7. Modeling of glints due to surface curvature using the virtual source technique. Ahmad T. Abawi, Michael B. Porter, and Martin Siderius (Heat, Light, and Sound Research, 12730 High Bluff Dr., San Diego, CA 92130)

The virtual source technique, also known as the method of superposition or the auxiliary source technique, has been widely used to model scattering from targets. In this technique the target is replaced by a collection of sources of unknown complex amplitudes. The source amplitudes are calculated by applying the appropriate boundary conditions at the surface of the target. In this paper, the virtual source technique is used to compute scattering from rough surfaces. For sound scattering from the rough ocean surface, this is done by solving the wave equation for a half-space ocean overlaying a bottom and forcing the solution to satisfy the pressure-release boundary condition at a desired surface. The method is applied to study the effect of focusing of energy (glints) due to reflection of sound from a gently sloping, but concave ocean surface for various ocean conditions. Additionally, the method is applied to the study of scattering of waves from an infinite periodic surface.

Contributed Papers

3:40

2pUW8. High-frequency limit under the Rayleigh hypothesis. Tanos Elfouhaily and Thomas Hahn (Univ. of Miami, RSMAS, 4600 Rickenbacker Cswy., Miami, FL 33149)

The Rayleigh hypothesis (RH) is traditionally invoked to compute the scattering amplitude in rough surface scattering. This hypothesis simplifies the analytic derivation of the scattering problem and has been proven to be accurate for the determination of the Volterra series for the small perturbation method (SPM). The SPM development physically corresponds to the scattering of low-frequency acoustical waves. In this study we focus on the high-frequency limit under the Rayleigh hypothesis (RH). We validate the consistency of the RH with the high-frequency limit to the order invoked in the approximation. This means that the first-order geometrical optics (GO1) is reached even though the Rayleigh hypothesis consists of the summation of solely upward scattered fields.

3:55

2pUW9. Scattering by surfaces with sub-Bragg structure. Frank Henyey and Dajun Tang (Appl. Phys. Lab., Univ of Washington, 1013 NE 40th St., Seattle, WA 92105)

The scattering from a rough pressure release surface, where the spectrum of the surface only contains wavelengths considerably shorter than half the acoustic wavelength, is considered. The amplitude of the surface

may be larger than that for which perturbation theory would be appropriate. We show that the surface can be replaced by an effective surface containing waves longer than half the acoustic wavelength. This effective surface is close to the envelope of the actual surface; this effective problem is likely to be much easier to solve than the original problem. We investigate the relationship between the problems and adjust parameters of the effective surface, by solving both problems with the integral equation.

4:10

2pUW10. Finite-difference time-domain simulation of acoustic propagation through a dispersive oceanic bubble cloud. Guy Norton (Naval Res. Lab., Acoust. Div., Stennis Space Ctr., MS 39529, guy.norton@nrlssc.navy.mil)

Accurate modeling of pulse propagation and scattering is a problem in many disciplines. For the case of an acoustic wave propagating in a two-dimensional non dispersive medium, a routine second-order in time and space finite-difference-time-domain (FDTD) scheme representation of the linear wave equation can be used to solve for the acoustic pressure. However, when the medium is dispersive, one is required to take into account the frequency-dependent attenuation and phase speed. Until recently, to include the dispersive effects one typically solved the problem in the frequency domain and then Fourier transformed into the time domain. However, by using a theory proposed by Blackstock [D. T. Blackstock, J. Acoust. Soc. Am. **77**, 2050 (1985)], the linear wave equation has been modified by adding an additional term that takes into account the disper-

sive nature of the medium. In this work a fourth-order (in time and space) 2-D FDTD scheme, which includes attenuation and dispersion via the convolution operator, is used to model backscattering of broadband signals from a dispersive composite subsurface bubble cloud composed of two different plumes underlying a random rough sea surface. [Work supported by ONR/NRL.]

4:25

2pUW11. A time domain approach to sea-surface scattering. Martin Siderius, Michael B. Porter, and Ahmad T. Abawi (HLS Res. Inc., 12730 High Bluff Dr., Ste. 130, San Diego, CA 92130)

Solutions to scattering problems are often formulated in the frequency domain which, for ocean acoustics, implies the surface is frozen in time. This may be reasonable for short-duration signals but breaks down if the surface changes appreciably over the time of the transmission. Additionally, frequency-do-main solutions are impractical for source-receiver ranges and frequency bands typical for applications such as acoustic communications (e.g., several kilometers, 3–50-kHz band). The time-varying nature of the sea surface adds to the complexity, often leading to a statistical description for the variations in received signals. A purely statistical description likely limits the insight that modeling generally provides. In this presentation we describe a time-domain modeling approach to the sea-surface scattering problem. The Helmholtz integral equation is used with the Kirchhoff approximation. The Green's functions representing the impulse response from all scattering points on the surface to the receivers are computed in terms of arrival amplitudes and time delays. This time-domain framework allows for including both broadband signals and the time-varying sea surface. The method will be compared with exact solutions for a subset of problems (frozen, rough surfaces) to determine limitations. Further, scattering results will be compared with those using Gaussian beam tracing.

4:40

2pUW12. Further work on the relationship between the coherent-to-incoherent ratio (CTIR) and the sea surface wave height for one-way propagation in a shallow ocean channel. R. Lee Culver, David L. Bradley (Appl. Res. Lab and Acoust. Grad. Program, The Penn State Univ., P.O. Box 30, State College, PA 16804, rlc5@psu.edu), Philippe Roux (LGIT-Maison des Gosciences, Grenoble cedex 9, France), William A. Kuperman (Univ. of California at San Diego, La Jolla, CA 92093-0238), and Stevenson Mark (NATO Undersea Res. Ctr., La Spezia, Italy)

A 29-element source array and 32-element receiver array have been deployed 8.6 km apart in a 120-m-deep ocean channel. Both arrays spanned most of the water column. Linear frequency-modulated (LFM) pulses 150 ms long and 4 kHz wide centered at 3.5 kHz were transmitted from one source at a time and recorded at all receivers during a several day period. During the recordings, the sound speed profile was approximately iso-velocity and the significant wave height varied from 0.3 to 1.1 m. We average the received signal coherently and incoherently for each source-receiver pair, average over all pairs, and take the coherent-to-incoherent ratio (CTIR). Log (CTIR) is found to depend linearly on time after the initial arrival, with slope proportional to the significant wave height. An earlier talk showed that the average reflection coefficient times the number of surface bounces N was a good predictor of the slope, but subsequently we find that it alone is not sufficient. Here we report a more complete theory that combines N, the average reflection coefficient, and the density of the arrivals. A predictive model for slope versus surface wave height is presented and compared with the measurements. The method can incorporate ray tracing and be extended to non-iso-velocity environments.

4:55

2pUW13. Detection in a nonstationary waveguide using orthogonal codes to measure the invariants of the time reversal operator. Franck Philippe, Claire Prada, Dominique Clorennec, Julien De Rosny (LOA, ESPCI, 10 rue Vauquelin, 75005 Paris, France), and Mathias Fink (LOA, 75005 Paris, France)

In underwater acoustics, time reversal (TR) techniques using an array of acoustic sources are commonly known for their efficiency in applications such as communication and detection. However, as TR methods assume stationary conditions, they are clearly affected by fluctuations of the medium. Among them, the decomposition of the time reversal operator (DORT method), although highly efficient for separating the echo from two close targets through a complex medium, suffers from the long acquisition time required. Indeed the whole interelement responses are needed meaning that N signals are acquired for an N-transducer array. The measurement is then at least N times longer than travel time. To reduce the acquisition time, orthogonal codes can be used [Folgot *et al.*, J. Acoust. Soc. Am. **117**, 3757–3765 (2005)]. These codes are tested in a laboratory detection experiment: a small-scale model of a shallow-water wave guide in which we add artificial surface waves of different frequencies and different amplitudes. The perturbations induced by the surface waves on the invariant of the time reversal operator are analyzed. Different methods which greatly reduce the acquisition time like Kasami and AIRS codes are compared.

Meeting of Accredited Standards Committee (ASC) S1 Acoustics

J. P. Seiler, Chair S1

U.S. Dept. of Labor, Mine Safety and Health Admin., P.O. Box 18233, Bldg. 38, Cochrans Mill Road, Pittsburgh, PA 15236

G. S. K. Wong, Vice Chair S1

Inst. for National Measurement Standards, National Research Council, Montreal Road, Bldg. M36, Ottawa, Ontario K1A OR6, Canada

Accredited Standards Committee S1 on Acoustics. Working group chairs will report on the status of standards currently under development in the areas of physical acoustics, electroacoustics, sonics, ultrasonics, and underwater sound, etc. Consideration will be given to new standards that might be needed over the next few years. Open discussion of committee reports is encouraged.

People interested in attending the meeting of the TAGs for ISO/TC 43 Acoustics and IEC/TC 29 Electroacoustics, take note – those meetings will be held in conjunction with the Standards Plenary meeting at 10:30 a.m. on Tuesday, 6 June 2006.

Scope of S1: Standards, specifications, methods of measurement and test, and terminology in the field of physical acoustics, including architectural acoustics, electroacoustics, sonics and ultrasonics, and underwater sound, but excluding those aspects which pertain to biological safety, tolerance and comfort.

Meeting of Accredited Standards Committee (ASC) S12 Noise

R. D. Hellweg, Chair S12

Hewlett Packard Co., Acoustics Lab., MR01-2/K15, 200 Forest St., Marlborough, MA 01752

W. J. Murphy, Vice Chair S12

NIOSH, 4676 Columbia Parkway, Mail Stop C27, Cincinnati, OH 45226

Accredited Standards Committee S12 on Noise. Working group chairs will report on the status of noise standards currently under development. Consideration will be given to new standards that might be needed over the next few years. Open discussion of committee reports is encouraged.

People interested in attending the meeting of the TAG for ISO/TC 43/SC 1 Noise, take note – that meeting will be held in conjunction with the Standards Plenary meeting at 10:30 a.m. on Tuesday, 6 June 2006.

Scope of S12: Standards, specifications and terminology in the field of acoustical noise pertaining to methods of measurement, evaluation and control, including biological safety, tolerance and comfort and physical acoustics as related to environmental and occupational noise.