Session 4aAA

Architectural Acoustics and Noise: Preserving Acoustical Integrity in the Course of Renovation

Daniel R. Raichel, Chair
2727 Moore Ln., Fort Collins, CO 80526

Chair's Introduction—8:30

Invited Papers

8:35

4aAA1. Renovating Teatro alla Scala Milano for the 21st century, Part I. Higini Arau (Travesera de Dalt, 118, 08024 Barcelona, Spain, h.arau@arauacustica.com)

Teatro alla Scala of Milan, known simply as La Scala throughout the world, is an old but venerable opera house that achieved legendary status in the world of music. A great number of singers of Olympian status have sung there; and major operas, among them Verdi's Falstaff and Otello and Puccini's Turandot, premiered there. The 227-year-old theater is beloved with a passion by the Milanese and the Italians, but it has suffered the ravages of time. It needed to be renovated in order to reverse material decay, meet current fire codes and security requirements, incorporate a new HVAC system, and to accommodate badly needed modern stage machinery. This renovation project took 3 years during which the theater was closed, and it included the construction of an elliptical 17-floor fly tower, designed by architect Mario Botta, for housing rehearsal rooms and serving as a scenery changing facility. The renovation proposal originally aroused a strong sense of melodrama among the extremely excitable Italian opera buffs who feared the desecration of their beloved edifice, but the acoustics and the beauty (carried out by Elisabetta Fabbri Architect) of the auditorium were preserved (and even enhanced). In this paper we explain how this project was achieved.

8:55

4aAA2. Renovating Teatro alla Scala Milano for the 21st century, Part II. Higini Arau (Travesera de Dalt, 118, 08024 Barcelona, Spain, h.arau@arauacustica.com)

The acoustic phase of La Scala renovation began in September 2002, after the main stalls and other sections of the theater were demolished. This assignment was twofold: (a) design of the auxiliary building with architect Mario Botta, and more importantly, (b) collaboration with architect Elisabetta Fabbri in restoration of the auditorium through acoustic analyses of proposed solutions. Only one set of acoustical measurements was known to be taken before demolition; and reliance had to be placed on hearsay from audience members. The author used his own computer program that included some of the salient features of other programs such as Odeon, Epidaure, Raynoise, etc. but avoided their pitfalls. This program was the only one that correctly predicted the known RT of the auditorium through the use of H. Arau Purchades formula [Arau, H., 1988. Acustica. Hirzel Verlag **65**(4), 163–180] and the authors dimension theory [Arau, H. 1997. Variation of the reverberation time of places of public assembly. Building Acoustics **4**(2).]. A new floor was designed to provide sufficient vibration transmission to the audience, actuating as a radiation box installed to direct sound vertically. Music Director Ricardo Muti pronounced the acoustical results as being excellent.

9:15

4aAA3. Maintaining the acoustics of Boston Symphony Hall. Robert Berens, Benjamin Markham, and Carl Rosenberg (Acentech Inc., 33 Moulton St., Cambridge, MA 02138, rberens@acentech.com)

Boston Symphony Hall celebrated its centennial in 2000; one of its biggest birthday presents was a new organ. More correctly, the partially new and partially refurbished organ made its debut in October 2004, for James Levine's inaugural concert as Music Director of the Boston Symphony Orchestra. Symphony Hall's original organ was replaced in 1949, but its replacement was never regarded as a great concert instrument. While some wanted the new organ work to include turning the Hall into an organ showcase, the BSO management was adamant that the acoustics of the Hall not be affected. Acentech was asked to provide technical oversight to help minimize any such changes. Tests were conducted to assess the potential impacts that removal and renovation of the organ pipes could have on Symphony Hall's acoustics. Reverberation times were measured in the organ chamber, on-stage, and in the Hall. Remedial measures were devised for use, if necessary, to counteract the expected changes in the organ chamber acoustics as the pipes were removed. However, such changes showed little effect on the acoustics of the stage or the hall. This paper describes the work done during the organ renovation process to protect the hallowed acoustics of Symphony Hall.

9:35

4aAA4. Historical preservation of acoustics at Spelman College, Atlanta. Carl Rosenberg, Benjamin Markham (Acentech Inc., 33 Moulton St., Cambridge, MA 02138, crosenberg@acentech.com), and Barbara Kovacs Black (Surber Barber Choate & Hertlein Architects, Inc., Atlanta, GA 30309)

In 1926, Spelman College in Atlanta dedicated its new Sisters Chapel, named after Laura Spelman Rockefeller and Lucy Maria Spelman, mother and aunt of John D. Rockefeller, Jr., the major donor. The Chapel has around 1000 seats and serves as a major venue for a myriad of programs, ceremonies, concerts, and student activities. In 2003, architects Surber Barber Choate and Hertlein were commissioned to renovate and rehabilitate the chapel with air-conditioning, new toilet facilities, new roof, enlarged balcony, and new sound system. The neo-classic pseudo Georgian basilica has a barrel vault running the length of the room. Sometime in the past, probably around 1950, the chapel ceiling was covered with a sound absorbing acoustic tile, presumably to rectify a focusing problem from the curved ceiling. With computer modeling, Acentech reconstructed the apparent acoustical problems and used this information to guide a renovation. The goal was to recapture the visual aspects of the original design, enhance the reverberation for the renovated Holtkamp organ and singing, and at the same time avoid any focusing problems for which the acoustic tile was presumably added. The project included mechanical system noise and vibration control and a new highly directional sound system.

9:55

4aAA5. Barnum Hall—The continuing renovation of a Streamline Moderne theater. Neil A. Shaw (Menlo Sci. Acoust., Inc. P.O. Box 1610, Topanga, CA 90290-1610, menlo@ieee.org), Kenneth Koslow (Santa Monica Malibu Unified School District, Santa Monica, CA 90404), Jean Sedillos (Restore Barnum Hall, Santa Monica, CA 90403), and Jim Mobley (Renkus-Heinz, Foothill Ranch, CA 92610)

Barnum Hall, built by the WPA in 1937 and located on the campus of Santa Monica High School, was the first Santa Monica Civic Auditorium. After 60 years it was closed for renovations. The building has historic significance, which placed some limitation on the renovation design as to what could be done. Balancing the interests of the various stakeholders—the Santa Monica-Malibu Unified School District, the City of Santa Monica Landmarks Commission, fund raisers, high school faculty, and community members, among others—impacted the design and construction process. The first phase included expanding and modernizing the stage house and surrounding support areas. Phase II of the work was concerned with mitigating the orchestra level rear wall reflections, modifying the balcony nose profile, adding an orchestra pit lift, the purchase of a custom acoustical orchestra shell, and improving creature comfort (air conditioning) and amenities (improving restrooms, creating a backstage area, and refurbishing 75% of the theater seats). A study of several sidewall shaping schemes were then analyzed to determine if the improvement in the level of envelopmental sound warranted the cost. Phase III will see the installation of a modern sound reinforcement system and refurbishing the last section seats.

10:15–10:30 Break

10:30

4aAA6. Renovating cultural icons. Gregory Miller (Talaske, 105 N Oak Park Ave., Oak Park, IL 60301, greg@talaske.com)

Three case studies of historic renovations are presented where acoustics were a key component of the renovation process. Each hall is an icon of the cultural life in the surrounding community. The first case study is Troy Savings Bank Music Hall, a late nineteenth century musical gem where recent renovations (adding variable acoustic features) required that the unamplified acoustics be unaltered. The second example, the Coronado Theatre in Rockford, Illinois, illustrates the sensitive modification of the acoustics in a historic vaudeville house adapted to modern multi-use requirements. Finally, The Great Hall at The Cooper Union in New York City, where Abraham Lincoln delivered his great "Right Makes Might" speech in 1860, is presented as an example of a renovation that utterly destroyed a historic acoustic environment, and discusses how this can be avoided.

Contributed Papers

10:50

4aAA7. Returning an acoustic legend to its original glory (and then some). Scott Pfeiffer (Kirkegaard Assoc., 801 W. Adams St., 8th Fl., Chicago, IL 60607)

Hill Auditorium at the University of Michigan broke all of the rules of modern day acoustic design, while still providing a tremendously rewarding experience for large audiences. The combination of diverse and intriguing programming by the University Music Society and the University Music Department, including the use of the Frieze Memorial Organ, provides tremendous opportunities for the community of Ann Arbor, MI to access great performances. The restoration project included air conditioning, elevators, and new sound systems. Additionally, improvements were made with respect to seating; specifically, sightlines, accessible seating, and under balcony seating were improved upon. This restoration project provided opportunities to revitalize the already famous acoustics without sacrificing unique and characteristic qualities of the hall. By drawing on the original architects, Albert Kahn Associates, utilizing the expertise of Quinn Evans Architects, as well as through the help of Gary Steffy Lighting, Fisher Dachs Associates, and many others in design and construction, (most notably a great client in the University of Michigan), the project secured the 2004 AIA National Honor Award.

11:05

4aAA8. Preserving the acoustics of the Mahaiwe Theater. Ronald Eligator (Acoust. Dimensions, 145 Huguenot St., New Rochelle, NY 10801, religator@acousticdimensions.com)

The Mahaiwe Theater, Great Barrington, Massachusetts, opened in 1905. It has been home to vaudeville, traveling Broadway shows, opera, concerts, and movies. In 2003, the theater began a multi-phase renovation project. The renovation project includes restoration of historic finishes,

new seating, installation of air conditioning and heating systems, and installation of sound and projection systems. A stage shell for music performances will also be installed. A key aspect of the renovation is the preservation of what has been widely regarded as excellent acoustics. Some aspects of the original structure contributed to this reputation (e.g., room shaping, finishes), while others undoubtedly were less helpful (e.g., the 1905-era ventilation system). The renovation provides an opportunity to preserve the positive attributes of the room, while correcting the deficiencies. Among the key issues discussed are the design of historically-sensitive finishes that respect the acoustic requirements of the multi-use space, acoustical design of a new quiet air conditioning system, and design of an enclosure for music ensembles.

11:20

4aAA9. Preservation of the acoustics of the Salt Lake Tabernacle, Lessons learned from acoustical characterization of the hall. Sarah Rollins and Timothy Leishman (Acoust. Res. Group, N283 ESC, Brigham Young Univ., Provo, UT 84606, sr223@email.byu.edu)

Current and historical properties of the Salt Lake Tabernacle have been investigated to characterize its unique acoustics. This characterization will be used to help maintain the current acoustical conditions after a seismic renovation. This paper discusses computer models developed for this purpose and impulse response measurements taken to quantify the acoustics in their current state. It also explores spatial variation of room acoustic parameters derived from the models and measurements.

THURSDAY MORNING, 19 MAY 2005 — PLAZA C, 8:15 TO 11:50 A.M.

Session 4aAB

Animal Bioacoustics: Tools for Animal Bioacoustics: New Designs and Directions I

William C. Burgess, Chair
Greeneridge Sciences, 6060 Graham Hill Rd., Felton, CA 95018

Chair's Introduction—8:15

Invited Papers

8:20

4aAB1. A binaural acoustic recording tag reveals details of deep foraging in beaked whales. Mark Johnson, Peter Madsen, Peter Tyack (Woods Hole Oceanograph. Inst., Woods Hole, MA 02543, majohnson@whoi.edu), and Natacha Aguilar de Soto (Univ. La Laguna, Tenerife, Spain)

A new acoustic and orientation recording tag (DTAG) for marine mammals contains two hydrophones, 25 mm apart, sampled at 192 kHz. Data are stored with loss-less compression in 6.6 GB of memory giving a 9.5 h recording capacity. Stereo DTAGs have been attached with suction cups to Cuvier's and Blainville's beaked whales in Liguria and the Canary Islands. Both species make regular clicks centered at 40 kHz during foraging dives. The DTAGs recorded clicks from the tagged whale and other whales nearby as well as echoes from targets in the water. The angle-of-arrival (AoA) of these sounds was determined both by cross-correlation and time-delay fitting of signals from the two hydrophones on the tag. The AoAs of clicks from tagged whales reveal that they turn their heads from side-to-side while foraging while the AoAs of echoes are consistent with a beam-width of about 20. By scanning a narrow acoustic beam, the whales may be able to search a large water volume while reducing clutter from multiple echoes. Frequent clicks from untagged whales with distinct AoAs indicate that several whales forage together during deep dives and such group cohesion may be a contributing factor to strandings of these species related to use of naval sonar.

8:40

4aAB2. Miniature self-contained acoustic recorders applied in a survey of beluga-whale populations in Knik Arm, Alaska. William C. Burgess (Greeneridge Sci., 6060 Graham Hill Rd Ste F, Felton, CA 95018), Michael T. Williams (LGL Alaska Res. Assoc., Anchorage, AK 99518), and Susanna B. Blackwell (Greeneridge Sci., Aptos, CA 95003)

Since 1995, acoustic recording tags attached to marine wildlife have increased our understanding of how animals use and respond to sound. Tagging studies require these instruments to be designed for minimum size and maximum recording capability; however, by designing for ease of use as well, a variety of other acoustic studies become feasible that could not otherwise be attempted for lack of time, staff, or support. As a result, designing for ease of use multiplies the scientific power of acoustic recording tag technology. In late 2004 four acoustic recording tags, known as Bioacoustic Probes and developed by Greeneridge with ONR support, were used as fixed recorders in a primarily visual survey of beluga whales in the Knik Arm of Cook Inlet, Alaska. The study took place with a compressed schedule and was staffed by field biologists with little time available for technical training on the use and maintenance of the instruments. Designing the instruments for ease-of-use enabled the study to include an independent acoustic component for assessing the presence or absence of beluga whales in areas that were being surveyed visually during daylight hours. [Work supported by Knik Arm Bridge and Toll Authority.]

9:00

4aAB3. Modular autonomous array deployments using passive acoustic time synchronization. Aaron Thode (Marine Physical Lab., Scripps Inst. of Oceanogr., 9500 Gilman Dr., MC 0238, La Jolla, CA 92093-0238, thode@mpl.ucsd.edu) and William Burgess (Greeneridge Sci., Felton, CA 95018)

Multi-element acoustic arrays generally use wired cable to transmit signals to a central recording location. While convenient, hardwiring hydrophones together increases array fragility and field costs while decreasing deployment flexibility. It is thus difficult to integrate acoustic array operations with field activities involving marine mammals in remote environments. Economic trends in the cell-phone and consumer electronics industries, combined with recent trends in tagging technology development, have led to the existence of compact low-power autonomous acoustic recorders that store data to either flash memory or small hard drives. In this presentation it is shown how two or more autonomous recorders can be time-synchronized using passive acoustic measurements of the background ambient noise field, effectively creating coherent array processing systems of varying aperture, spacing, and deployment geometry. Configurations tested to date include short and large-aperture vertical arrays off the coasts of Australia and Alaska, bottom-mounted horizontal arrays in San Ignacio Lagoon, Baja California, and instruments installed inside a glider. [Work sponsored by ONR Acoustic Entry Level Faculty Award.]

9:20

4aAB4. High-frequency Acoustic Recording Package (HARP) for long-term monitoring of marine mammals. Sean Wiggins, Chris Garsha, Kevin Hardy, and John Hildebrand (Scripps Inst. of Oceanogr., 9500 Gilman Dr. MPL-0205, La Jolla, CA 92093-0205)

Advancements in low-power and high-data-capacity computer technology during the past decade have been adapted to autonomously record sounds from whales over long time periods. Acoustic monitoring of whales has advantages over traditional visual surveys including greater detection ranges, continuous long-term monitoring in remote locations under various weather conditions, and lower cost. One currently used tool for providing long-term acoustic monitoring of marine mammals is an autonomous acoustic recording package (ARP) which uses a tethered hydrophone above a seafloor-mounted instrument frame. Since 2000, ARPs have been deployed to record baleen whale sounds in the Bering Sea, in the Beaufort Sea, in the Gulf of Alaska, off the coast of southern California, around Antarctica and near Hawaii. ARP data have provided new information on the seasonal presence, abundance, call character and patterns of calling whales. The need for a broader-band, higher-data capacity system capable of recording odontocete whales, dolphins and porpoises for long time periods has prompted the development of a High-frequency Acoustic Recording Package (HARP). The HARP design is described and data analysis strategies are discussed using examples of HARP broad-band (sample rates up to 200 kHz) recorded data.

9:40

4aAB5. A near-real-time acoustic detection and reporting system for endangered species in critical habitats. Christopher W. Clark, Thomas Calupca (Bioacoustics Res. Program, Cornell Lab. of Ornithology, 159 Sapsucker Woods Rd., Ithaca, NY 14850), Douglas Gillespie (Intl. Fund for Animal Welfare, Yarmouth Port, MA 02675), Keith Von der Heydt, and John Kemp (Woods Hole Oceanograph. Inst., Woods Hole, MA 02543)

Passive acoustics is an effective mechanism for detection and recognition of species-specific sounds and can be a more cost-effective approach than visual techniques for monitoring populations of rare or endangered species. A network of moored buoys has been strategically deployed in and around Cape Cod Bay to report detections of northern right whales in critical habitat. Each buoy continuously and automatically monitors for right whale contact calls and transmits detection and ambient noise data by cell or satellite phone to Cornell University on a regular basis. Each day, validated data are automatically unloaded into a Website database to provide on-line graphical and numerical data summaries. The array of three buoys deployed in the Bay will eventually be synchronized to allow localization and tracking of individual animals. [Work supported by funds from the NOAA Right Whale Grants Program and augmented by funds from the Commonwealth of Massachusetts Division of Marine Fisheries.]

10:00–10:10 Break

10:10

4aAB6. Extensible bioacoustical analysis software: Two examples. Harold G. Mills and Harold K. Figueroa (Bioacoustics Res. Program, Cornell Lab. of Ornithology, 159 Sapsucker Woods Rd., Ithaca, NY 14850, hgm1@cornell.edu)

Two extensible software tools for bioacoustical analysis have been developed by the Bioacoustics Research Program at the Cornell Laboratory of Ornithology. Each tool provides a framework for sound visualization and analysis, supporting both manual selection and automatic detection of acoustic events in arbitrarily long recordings, manual annotation and automatic measurement of properties of these events, and management of the created acoustic metadata. The acoustic events of interest might include animal vocalizations and/or anthropogenic sounds. Simple examples of measurable event properties include duration, total energy, and spectrum. More advanced measurements might classify sounds according to the species or individuals that produced them, or estimate the location of a vocalizing animal. Both software tools can be extended in various ways, including the addition of new detectors and measurements. It has been found that this extensibility adds great value to the tools by allowing them to be readily adapted for new applications by bioacousticians or others with programming ability. It has also been found that such tools greatly facilitate the development of new detection and measurement algorithms by allowing prototype algorithms to be rapidly evaluated. Several case studies are presented that illustrate these principles. [Work supported in part by NSF.]

10:30

4aAB7. Measurement of biosonar signals of echolocating bat during flight by a telemetry system. Hiroshi Riquimaroux (Dept. of Knowledge Eng. and Comput. Sci., Doshisha Univ., Kyotanabe, Kyoto 610-0321, Japan)

A telemetry microphone system (Telemike) has been developed, which can be mounted on the bat's head, to measure acoustic characteristics of emitted pulses and returning echoes. The system allows us to monitor what the bat listens to during its flight. A high-speed video camera system has been also adopted together with the Telemike to trace positions in space of the flying bat for analyzing temporal pulse emission patterns. With those devices, how the CF-FM bats execute parallel time-sharing real-time processing during their flight can be investigated. Doppler-shift compensation, echo amplitude compensation and processing for multiple target detections were observed. Some of the evidence found by the Telemike will be introduced and discussed. [Work partly supported by the Innovative Cluster Creation Project promoted by MEXT and by a grant to RCAST at Doshisha University from MEXT.]

Contributed Papers

10:50

4aAB8. 3D modeling and animation techniques to elucidate mechanisms of echolocation-based acoustic perception. Seth Horowitz (Univ. at Stony Brook, Dept Psychiatry, HSC T-10 Rm. 086, Stony Brook, NY 11794, shorowitz@neuropop.com) and James Simmons (Brown Univ., Providence, RI 02912)

Current models of echolocation focus on the presence of specular acoustic surface reflections or ''glints'' as the perceptual cues that provide information about object distance and geometry. Using current material and motion 3D modeling techniques for visual animation, we have developed models of common real-world elements that can be detected by bat biosonar. By mapping acoustic reflectivity data from natural and artificial targets onto luminosity, specularity and reflection coefficients for materials and accurate spatial models of these objects, we have generated visual glint-like reflections from object surfaces that are similar to echolocation data. This allows us to create 3D visual animations of echolocation auditory scenes. Using these animations, human observers can integrate object shapes, sizes and movements based on gestalt grouping, particularly when the target moves or the echolocation source is in moving in relation to the target. These types of modeling tools may help elucidate the perceptual phenomena arising from integration of the individual acoustic glint structures that allow bats and other echolocating animals to create complex umwelts from auditory data.

11:05

4aAB9. The acoustic environment of the southern resident killer whales in Haro Strait: Propagation modeling and analysis of field measurements. Christopher Jones and Michael Wolfson (Appl. Phys. Lab., Univ. of Washington, Seattle, WA 98105)

We will discuss the complicated shallow water acoustic environments of the southern resident killer whales in the Haro Strait of the Puget Sound through combined analysis of field measurements and acoustic propagation modeling in the frequency range of 1 to 20 kHz. Haro Strait is a highly variable acoustic environment with active commercial shipping, whale watching, and Naval activity. Southern resident killer whales are of unique public concern in this area because of potentially high impact by anthropogenic sound sources, either by auditory masking that may interfere with foraging strategies or annoyance and disorientation due to complicated reverberation. Predictive acoustic modeling in combination with field measurements can aid in understanding the mechanisms of impact and better inform assessment of the risk, providing a critical step towards the quantitative evaluation of impact in the context of complicated environments, changing background sound levels, and emerging management issues. Preliminary results of modeling and analysis of data collected in the summer of 2004 will be presented.

11:20

4aAB10. Ship strike acoustics. Edmund Gerstein, Joseph Blue (Leviathan Legacy Inc., 1318 SW 14th St., Boca Raton, FL 33486), and Steve Forsythe (Naval Undersea Warfare Ctr., Newport, RI)

The confluence of Lloyds mirror effect together with acoustical shadowing and spherical spreading pose significant detection challenges for whales and manatees. Direct measurements of approaching vessels using horizontal and vertical hydrophone arrays demonstrate how the noise from vessels can become indistinguishable from ambient noise. Geometric scattering is also estimated for various hull dimensions as the acoustical shadows cast ahead of ships provide no warning to animals near the surface within the shadow zone. Animals at sufficient depths, or outside the shadow boundaries that hear approaching vessels, may seek refuge near the surface or directly in the path of vessels where it is relatively quiet. Speed reductions proposed to reduce collisions do not address the underlying acoustical challenges marine mammals face. Field measurements support predictions that noise intensity is proportional with vessel speed to the 5th power. In multiple ship environments the acoustical masking challenges are greatest and noise from slow vessels operating in marine mammal corridors can mask the sounds of distant faster moving vessels. A low intensity bow-mounted projection system has been designed to selectively fill-in acoustical shadows with modulated ship noise to mitigate masking and near surface effects, and neutralize the dangerous ambiguity posed by acoustical shadows.

11:35

4aAB11. Detection ranges for acoustic based manatee avoidance technology. Richard Phillips (Dept. of Mech. and Aerosp. Eng., Univ. of Florida, Gainesville, FL 32611-6250), Christopher Niezrecki (Univ. of Massachusetts Lowell, Lowell, MA 01854-2881, Chris_Niezrecki@uml.edu), and Diedrich O. Beusse (Univ. of Florida, Gainesville, FL 32610-0126)

The West Indian manatee (*Trichechus manatus latirostris*) has become endangered partly because of watercraft collisions in Florida's coastal waterways. Several boater warning systems, based upon manatee vocalizations, have been proposed to reduce the number of collisions. One aspect of the feasibility of an acoustically based system relies upon the distance at which a manatee vocalization is detectable. The magnitude of environmental noise and manatee vocalizations, as well as the acoustic spreading properties of the habitat will help to estimate the detection range of a vocalizing manatee. This study combines measured source levels of manatee vocalizations with the modeled acoustic properties of manatee habitats to develop a method for determining hydrophone spacing requirements for acoustically reliant manatee avoidance technologies. In quiet environments (background noise 70 dB) it was estimated that manatee vocalizations are detectable at approximately 250 m, with a 6 dB detection threshold. In louder environments (background noise 100 dB) the detection range drops to 2.5 m. Noise generated by boating traffic is also investigated. In a habitat with 90 dB of background noise, a passing boat with a maximum noise floor of 120 dB would be the limiting factor when it was within approximately 100 m of the hydrophone.

Session 4aNS

Noise, ASA Committee on Standards and Engineering Acoustics: Noise Control Feasibility: Technical, Legal and Economic Issues

John P. Seiler, Cochair
Mine Safety and Health Administration, Cochran Mill Rd., Pittsburgh, PA 15236

Angelo J. Campanella, Cochair
Campanella Associates, 3201 Ridgewood Dr., Hilliard, OH 43026-2453

Chair's Introduction—9:00

Invited Paper

9:05

4aNS1. Industrial noise control—Technical and economic feasibility. Stephen I. Roth (Roth Acoust. Assoc., 2352 Norton Rd., Pittsburgh, PA 15241, sroth@rothacoustics.com)

Industrial noise control is too often considered technically not possible or economically infeasible. The use of ear protection devices becomes the fallback approach to attempt to reduce the risk of worker hearing loss. In reality there are noise control opportunities in industrial settings than are low cost and provide significant noise reduction without affecting production and maintenance practices. This paper will present mechanical and pneumatic noise controls that can be successfully applied to industrial environments. This paper will also present information on OSHAs approach to determine whether engineering noise controls are considered economically feasible.

Contributed Papers

9:25

4aNS2. The feasibility of noise and reverberation control for good acoustics in classrooms. Bennett M. Brooks (Brooks Acoust. Corp., 27 Hartford Turnpike, Vernon, CT 06066, bbrooks@brooksacoustics.com)

The acoustics of classrooms has been the subject of increasing scrutiny, as good speech intelligibility is recognized as an essential feature of an effective educational environment. There is consensus on the need for low levels of background noise and reverberation. Acceptable values for these parameters have been published as a standard [ANSI S12.60-2002]. The technical feasibility of achieving these goals is well established. In the US, few jurisdictions require that classrooms meet acoustical standards. The limits to constructing acoustically adequate classrooms are primarily economic. Perception and understanding of the issue are also important, as those who specify and pay for classrooms, school boards and taxpayers, are not those directly served in them, students. As the direct customers and their immediate providers, teachers, have little power, feasibility depends on competing priorities set by interested non-users. Therefore, advocates must sell the cost/benefits to the decision makers. Widely scattered data are available on costs as a percentage of the total school building project. However, detailed cost data for individual mechanical system components that meet noise standards are scarce. Also, long-term economic paybacks must be quantified. Available cost data and areas needing better definition are summarized here.

9:40

4aNS3. Low impact renovations with high performance sound insulation. Kenneth P. Roy and Sean D. Browne (Armstrong Innovation Ctr., 2500 Columbia Ave., Lancaster, PA 17604)

Architectural renovations, although often driven by visual objectives, are a good opportunity to pursue acoustical interventions involving both sound intrusions into a space, and sound quality of the acoustic environment within the space. A high priority is to keep the construction impact low as it affects both project cost and schedule. Obviously, if the architectural intervention can be accomplished using external measures such as the addition of added elements to the existing structure, then that would be a definite advantage. Test data will be presented for the application of sound insulating/absorbing panels to existing drywall structures. It will be shown that such an overlay panel can add sufficient damping to result in essentially "mass law" performance of the system. This will not only result in STC improvement, but more importantly will result in even higher voice insulation between spaces. This effect will be auralized for comparison listening purposes. If the added panel is also of a sound absorbing nature, such as those used in the tested cases, then they will also have the effect of adding to the receiving room sound performance.

9:55

4aNS4. Structural vibration isolation design for a magnetic resonance imaging (MRI) system. Chad Himmel (JEAcoust., 1705 W Koenig Ln, Austin, TX 78756, info@jeacoustics.com)

This structural vibration control case study presents problems, constraints and design solutions for existing magnetic resonance imaging system (MRI) installation in a medical office facility. Objective: Reduce MRI noise received in adjacent, unrelated spaces. Manufacturer's data indicated that airborne MRI sound emissions could exceed permissible noise criteria for nearby occupied rooms. Experience with MRI systems indicated possibility of structure borne vibration that could result in radiated noise in other spaces. Structure borne vibration paths needed attenuation or isolation, to prevent excessive or annoying and distracting noise to adjacent office spaces. In addition, containment design was required for loud noises in the magnet room. Measurements revealed transient vibration of building floors, walls, and ceilings coincident with MRI operation. Measurements also revealed transient airborne noise levels consistent with radiated sound produced by vibration acceleration of large surfaces. Spectral analyses of noise and vibration led to design of a retrofit vibration isolation mounting scheme for the MRI on existing building floor. Design parameters included structural resonance of slab, manufacturer's allowable vibration for MRI, and receiver room permissible criteria. Noise and vibration measurements will show building conditions before and after implementation of retrofit vibration isolation mounts. Plan and section drawings will illustrate design solutions.

10:10–10:30 Break

10:30

4aNS5. MRI system vibration and noise considerations in hospital design. Eric E. Ungar and Jeffrey A. Zapfe (Acentech, Inc., 33 Moulton St., Cambridge, MA 02138, eungar@acentech.com)

Magnetic Resonance Imaging (MRI) systems are increasingly being located on above-grade floors of hospital buildings. The floor structures that support these systems need to provide vibration environments that satisfy the system's criteria, and airborne and structure-borne transmission of noise to neighboring areas needs to be taken into account in order to provide acceptable acoustical environments in these spaces. The fundamental operating principles of MRI systems are reviewed in relation to the systems' susceptibility to vibration and to their sound producing mechanisms. Vibration criteria for some widely used systems are discussed and spectra of airborne and structure-borne noise are illustrated. Means for vibration control and for the attenuation of structurally transmitted sound are delineated, together with architectural considerations for limiting noise transmission to spaces adjacent to MRI suites.

10:45

4aNS6. Pyschoacoustical comparison of active versus passive noise control techniques. Gerard Mangiante and Georges Canevet (Laboratoire de Mecanique et d'Acoustique, CNRS, 31 Chemin Joseph-Aiguier, 13402 Marseille Cedex 20, France)

Comparisons of active versus passive noise control techniques can be found in various papers. However, this comparison mainly concerns economic, operational, or technical considerations. The present contribution aims at describing the psychoacoustical effects produced by some of the classical solutions used in passive and active noise control. The models introduced by Zwicker and his coworkers, and by Moore and Glasberg are used to evaluate the auditory efficiency of passive and active noise control techniques. Several types of signals were examined: (i) test signals obtained with a band of noise embedded in a white noise or a pink noise; (ii) actual environmental noises: noise produced by the turbine of an aircraft or by a car engine, and several interior noises (locomotive, helicopter and car). It is shown that the modifications in the spectrum of a signal that can be produced by active control are sometimes disappointing, because they induce a subjective enhancement of the high-frequency portion of the spectrum. What the listener then commonly reports is that overall the signal has become slightly softer, but also more unpleasant. The use of a hybrid noise control technique, combining active and passive control, can greatly reduce this effect.

11:00

4aNS7. Adaptive multi-modal active noise control. Adam K. Smith, Jeffrey S. Vipperman, and Daniel D. Budny (Univ. of Pittsburgh, Dept. of Mech. Eng., 648 Benedum Hall, Pittsburgh, PA 15261)

Low frequency Active Noise Control (ANC) has been found to work well for sound suppression in acoustic ducts. Feedforward and feedback are the two distinct methods in which to implement (ANC) schemes. A feedforward implementation is realized by utilizing a known disturbance measurement (e.g. with a microphone), and using the measurement to compute the actuator (loudspeaker) signal. Feedback control modifies dynamics of an enclosed sound field to add damping. For example, one can use a resonant filter in the feedback path that is tuned to a selected mode in the enclosed sound field to be controlled. The resonant filter model is similar to those used for positive position feedback (PPF) in structures. A method of compensating the strong speaker dynamics (phase angle) was recently investigated by Bisnette and Vipperman (2004). The method was found to improve the performance and stability of the controller. Here, the above mentioned procedure is expounded upon to provide improved multi-modal performance, using higher order band-bass filters. Methods for adapting the controller parameters (e.g., gain, frequency, bandwidth) are also presented.

11:15

4aNS8. Acoustic power and intensity control by absorptive materials arrangement in an enclosure. Sung-Ho Cho and Yang-Hann Kim (Dept. of Mech. Eng. KAIST, 373-1 Sci. Town, Daejon-shi, Korea)

This paper studies how the sound distribution of room or cavity is affected by the absorptive materials arrangement on the wall. In other words, we want to know, very specifically, how the change of boundary condition affects rooms sound distribution. The boundary condition can be modified by changing not only absorbents position but also its acoustic impedance. The effect of changing boundary condition is expressed in terms of modal admittance on the enclosures surface. What we have gotten from this study is that the absorptive materials placement is closely related to wave length, cavity size and geometry. The authors pay attention to the intensity field, especially reactive intensity divergence in space. The divergence of reactive component of intensity is directly related to acoustic energy distribution which is composed of potential and kinetic energy. Using acoustic energy balance equation, the relation between global noise control performance and absorptive material arrangement is explained. In this point of view, the possibility of global noise reduction will be discussed in terms of acoustic potential energy reduction.

11:30

4aNS9. Sound attenuation by lattices of rigid elliptic cylinders. Daniel Torrent, Andreas Håkansson, and José Sánchez-Dehesa (Nanophotonics Technol. Ctr., Tech. Univ. of Valencia, E-46022 Valencia, Spain, jsdehesa@upvnet.upv.es)

The attenuation properties of acoustic barriers consisting of a few layers of rods with elliptical cross section has been studied by multiple scattering theory. Their performance as a function of number of layers and orientation of the ellipses will be reported and compared with similar structures based on cylinders of circular section. A comparison with available experiments will also be presented. Finally, a genetic algorithm is applied to optimize the attenuation in a wide range of frequencies. [Work supported by MEC of Spain.]

Session 4aPAa

Physical Acoustics, Biomedical Ultrasound/Bioresponse to Vibration and ASA Committee on Standards: Cavitation and Other Mechanical Effects in Biomedical Ultrasound: A Special Session to Honor the Work of Wesley Nyborg I

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

Junru Wu, Cochair
Dept. of Physics, Univ. of Vermont, Burlington, VT 05405

Chair's Introduction—7:55

Invited Papers

8:00

4aPAa1. Wes Nyborg: Scientist, role model, and friend. Marvin C. Ziskin (Ctr. for Biomed. Phys., Temple Univ. Med. School, 3420 N. Broad St., Philadelphia, PA 19140)

Wesley L. Nyborg obtained his Ph.D. degree in physics at Pennsylvania State College in 1947. He has held faculty positions at Penn State, Brown University, and since 1960 has been at the University of Vermont where he is presently Emeritus Professor. Dr. Nyborgs entire career has been devoted to biophysical acoustics: low frequencies at first, but since the early 50s ultrasonic frequencies of biomedical interest. He has made significant contributions in many areas of ultrasound biophysics, especially in cavitation and other non-thermal mechanisms relevant to biological effects. He has also derived a number of important relationships that are widely used today in determining temperature elevation in clinical ultrasound examinations. He has been the role model and friend to many colleagues, and has received many awards.

8:20

4aPAa2. Sound sources and propagation. Mahlon Burkhard (157 Stephen Ln., Charles Town, WV 25414)

From July 1945 through May 1950, the Acoustics Laboratory of the Department of Physics at Pennsylvania State [College] University conducted ''investigations concerning the production and propagation of sound, both sonic and ultrasonic, in the lower atomosphere, through the ground, and in other media.'' Wesley L. Nyborg was a key member of the group of 53 researchers employed in the program at some-time during the period. A sound source used extensively in the study of ultrasonic propagation in the atmosphere was a small whistle. ''Edge tones'' generated by a jet of air impinging on a narrow object are phenomena critical to the operation of these whistles as well as organ pipes and many sounds associated with fluid flow. Because the sound can be controlled and levels can be quite high, one proposed application was as a source for radiating small quantities of liquids with ultrasound. In addition to developing a physical understanding of this sound generation mechanism, Nyborg participated in studies and analysis of the influence of micrometeorology on propagation of high frequency sound in the atmosphere.

8:40

4aPAa3. Infrasonic tonal resonances of tropical hurricanes. Samuel A. Elder (Phys. Dept., U.S. Naval Acad., Annapolis, MD 21402)

In news pictures saturating the media last summer there is contained subtle but compelling evidence that tropical hurricanes emit intense and extremely low-frequency tones, originating in the eye. The excellent symmetry of a well-formed eye predisposes the central cavity of a hurricane to resonate in the normal modes of a low-Q nearly cylindrical chamber, due to the slight mismatch in sound speed at the eye wall. Second order perturbation analysis of interaction between eye-modes and steady wind corkscrewing up through the eye has confirmed that positive feedback is generated, the condition for self-excited oscillation. Dimensions and speed of typical hurricanes require that the tones have frequencies on the order of 1/100 of 1 Hz, which is doubtless the reason why they have escaped previous detection. Visual evidence appears in multiple alternating rain and dry bands that originate around the inner circumference of the eye wall, observable in aerial photos by NOAA. These are most easily explained as due to intense acoustically-driven temperature oscillations about the dew point, inside the eye. For a given hurricane, the actual tonal frequencies can be deduced from the spacing of these startups.

9:00

4aPAa4. Some medical applications of acoustic streaming. Lawrence A. Crum (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105)

Classical acoustic streaming results when an acoustic field is absorbed in a liquid, and the resulting momentum transfer causes the liquid to be translated. Acoustic microstreaming results when a bubble or other compressible entity is caused to oscillate by an acoustic field and hydrodynamic flow is induced in the vicinity of the oscillating surface. Much of the pioneering work on this general topic was performed by Wesley Nyborg and his students at the University of Vermont. There are a number of conditions for which acoustic streaming can be useful in medical ultrasound; e.g., one can enhance diffusion rates of drugs by inducing streaming at a specific site in tissue; together with Doppler imaging, one can determine the consistency of a particular sample of fluid, such as a blood clot. These and other examples of the use of streaming in medical ultrasound will be presented. [Work supported in part by the NIH and NSBRI.]

9:20

4aPAa5. The mechanical index and cavitation in tissue. Charles C. Church and Xinmai Yang (Natl. Ctr. for Physical Acoust., Univ. of Mississippi, University, MS 38677)

The mechanical index (MI) quantifies the likelihood that exposure to diagnostic ultrasound will produce an adverse biological effect by a nonthermal mechanism. The current formulation of the MI is based on inertial cavitation thresholds in two liquids, water and blood, as calculated by Apfel and Holland [Ultrasound Med. Biol. **17**, 179–185 (1991)]. Although tissue contains a high proportion of water, it is not a liquid but a viscoelastic solid. The importance of this difference was studied by deriving a Keller-Miksis-like equation assuming a gas bubble in a linear Voigt solid and performing numerical computations similar to the analytical work underlying the MI. Thresholds for inertial cavitation were determined for pulse lengths of 1–14 acoustic periods, equilibrium bubble radii (R_o) of 0.1–10.0 mm, a frequency range of 0.5–15.0 MHz, 2 threshold (P_t) criteria: $R_{max}=2R_o$, and $T_{max}=5000$ K, and tissue elasticities and viscosities of $G=0$, 0.5, 1.0 or 1.5 MPa and $\mu=0.005$ or 0.015 Pa·s; which span the range of values for soft tissue. It is found that thresholds in tissue are up to 10 times those in liquid, and that P_t increases nearly linearly with frequency. The relevance of these results to ultrasound safety will be discussed.

9:40

4aPAa6. Wesley Nyborg and bioacoustics at the University of Vermont. Junru Wu (Dept of Phys., Univ. of Vermont, Burlington, VT 05405)

Wes Nyborg came to UVM in 1960. Burlington, Vermont has become his home. He did his most pioneering research in microstreaming, acoustic radiation pressure and bioeffects of ultrasound at UVM. His research was continuously supported over 20 years by NIH. In 1987, after receiving Ph.D. in Physics and working with Isadore Rudnick for two years as a postdoc in the field of nonlinear physics, I joined the faculty of the University of Vermont as an assistant professor. Dr. Nyborg just retired from teaching in 1986. He was the person who convinced me to start research in the field of biomedical ultrasound. This presentation will review the development and achievement of bioacoustics at UVM and will also introduce him as a mentor, friend, colleague and collaborator.

Contributed Papers

10:00–10:30 Break

10:30

4aPAa7. A technique for monitoring and controlling cavitation activity during high intensity focused ultrasound application. Charles R. Thomas, Caleb H. Farny, Ronald A. Roy, and R. Glynn Holt (Dept. of Aerosp. and Mech. Eng., Boston Univ., 110 Cummington Streeet, Boston, MA 02215)

It has been reported that cavitation can lead to enhanced heating in the focal region of a HIFU source. In order to exploit this heating for *in vivo* use, it is essential that the cavitation only occur in the focal region. Thus, the onset and evolution of inertial cavitation activity must be monitored and controlled during HIFU therapy. One candidate sensor is a confocally-aligned passive cavitation detector; however this would add complexity to a clinical HIFU applicator. Instead we propose that the HIFU source itself can serve as a monitoring device. The combination of broadband acoustic emissions (inertial cavitation) and backscatter (stable cavities) emanating from the HIFU focus manifests itself as fluctuations of the otherwise constant driving voltage amplitude, providing a convenient means for sensing cavitation activity. We will present results of experiments assessing the feasibility of using the variance in the amplitude of the HIFU drive voltage as a feedback control signal. Success is determined by the shape of the lesion created using supplemental control compared to the shape obtained otherwise. Two independent control parameters were used: the amplitude and the duty cycle of the HIFU. [Work supported by the Department of the Army award DAMD17-02-2-0014.]

10:45

4aPAa8. Spherical bubble pulsation between parallel plates. Jianying Cui, Mark F. Hamilton (Dept. of Mech. Eng., Univ. of Texas, 1 University Station C2200, Austin, TX 78712-0292), Preston S. Wilson, and Evgenia A. Zabolotskaya (Univ. of Texas, Austin, TX 78713-8029)

A simple model is proposed for investigating bubble pulsation in a constrained medium. In free space, compressibility of the surrounding medium introduces primarily radiation losses that can often be neglected. In a constrained medium, compressibility plays an essential role. The example we investigate here is linear pulsation of an acoustically driven spherical bubble between two parallel rigid plates. The method of images is used to obtain a bubble dynamic equation that accounts for time delays associated with reflections of pressure waves radiated by the bubble. When the bubble is midway between the plates, the series of terms for the reflections can be expressed in closed form. For narrow plate separations, less than about ten bubble diameters, the bubble resonance frequency is reduced from its value in a free field due to increased effective inertia of the liquid, and radiation damping is increased. For larger plate separations, but less than the acoustic wavelength, the quality factor associated with radiation damping is proportional to plate separation. If effects of compressibility are ignored, the effective inertia becomes infinite, and bubble pulsation is prohibited by the equations of fluid mechanics. [Work supported by NIH Grant EB004047 and ARL IR&D funds.]

11:00

4aPAa9. Evaluation of backscattered intensity to quantify the destruction rate of echogenic liposomes. Tyrone M. Porter, Sampada S. Vaidya, Christy K. Holland (Dept. of Biomed. Eng., Univ. of Cincinnati, Cincinnati, OH 45267), Shao-Ling Huang, Robert C. MacDonald (Northwestern Univ., Evanston, IL), and David D. McPherson (Northwestern Univ., Chicago, IL)

Echogenic liposomes (ELIP) are vesicles with a phospholipid bilayer shell that can serve as ultrasound contrast agents (UCA) for diagnostic purposes and for targeted drug delivery. The efficacy of ELIP as an UCA depends upon its stability in an acoustic field, whereas the use of ELIP for drug delivery will require releasing the encapsulated drug rapidly at the desired treatment site. The objective of this study was to evaluate the rate of destruction of ELIP as a function of pressure and pulse repetition frequency (PRF). Assuming ELIP destruction is directly related to its echogenicity, transducers with center frequencies of 3.5, 7.5, and 10 MHz were used to acquire pulse-echo data from ELIP suspensions ([lipid] $=0.2$ mg/ml). The rate of ELIP destruction was defined by fitting an exponential decay function Ae^{-kt} to the backscattered intensity data. The relationship between ELIP destruction and acoustic pressure and PRF was determined by comparing the decay time constant k across samples. Based on this analysis, the rate of ELIP destruction was found to be directly proportional to pressure and PRF for all frequencies tested.

11:15

4aPAa10. Microbubble oscillations and stability for drug delivery. John S. AllenIII (Dept. of Mech. Eng., Univ. of Hawaii-Manoa, 2540 Dole St., Honolulu, HI 96822)

Ultrasound contrast agents have been developed from micron size bubbles whose gas core is enclosed by a polymer, lipid or protein shell. Furthermore, specific designs have been developed for drug delivery in which the ultrasound contrast agent acts as drug delivery vehicle. A drug may be suspended in the shell of these agents which is released at a particular site as the microbubble undergoes destruction. Localized delivery depends on the destruction of a sufficient number of bubbles within a confined geometry typically a capillary or small vessel. Experimental evidence suggests the composition and thickness of the shell play important role in the break-up. Break-up scenarios of buckling have been suggested for elastic shell agents and shape stability for fluid shell agents. However, these have not been rigorously examined theoretically. The formulations of contrast agent designed for drug delivery are highlighted including a description of a double polymeric layer design. Shape stability equations for fluid shell agents are derived and analyzed in limiting cases. Furthermore, biological issues related to potential endothelial cell transport and interactions are briefly discussed.

11:30

4aPAa11. Cavitation inception on micro-particles: Possible drug carriers? Manish Arora, Bram Borkent, and Claus-Dieter Ohl (Phys. of Fluids, U. Twente, Postbus 217, 7500 AE Enschede, The Netherlands)

Hydrophobic particles are known to act as cavitation nuclei when they are exposed to a sufficient tensile wave. Yet, the dynamics of cavitation inception from particles, the growth of the bubble and the separation from the particle has only recently been observed. [M. Arora, C. D. Ohl, and K. A. Moerch, Phys. Rev. Lett. **92**, 174501 (2004)] In this presentation, high-speed photography of the remarkable dynamics is presented together with a modeling effort using a force balance approach. One of the results is that particles are accelerated to velocities up to 10 m/s and detach from the bubble. It was proposed, that micro-particles might act as drug carriers which can be activated by a tensile wave and accelerate into tissue or cells. In an effort to explore this possibility experiments on the cavitation inception ability on various types of micro-particles have been conducted in a reproducible way. Cavitation inception is initiated with a single cycle tensile wave and recorded with a camera. The number of cavitation bubbles decreases from shot-to-shot, which can be explained with one-time trigger-able nucleation sites on the particles.

11:45–12:00
Open Comments

THURSDAY MORNING, 19 MAY 2005 PLAZA A, 8:00 TO 10:15 A.M.

4a THU. AM

Session 4aPAb

Physical Acoustics: Wind Noise and Atmospheric Sound Propagation

Richard Raspet, Chair
Univ. of Mississipi, NCPA, Coliseum Dr., University, MS 38677

Contributed Papers

8:00

4aPAb1. Framework for windnoise studies. Richard Raspet, Jeremy Webster, and Kevin Dillion (Dept. of Phys. and Astron., Univ. of Mississippi, P.O. Box 1848, University, MS 38677)

Research in wind noise reduction in outdoor measurement microphones has been limited largely to comparisons between bare and screened microphones. Morgan and Raspet [J. Acoust. Soc. Am. **92**, 1180–1183 (1992)] used simultaneous wind velocity and noise measurements to show that the source of wind noise is incident wind fluctuations. In this paper, two methods for predicting the upper limits of wind noise pressure spectra from velocity spectra in the inertial range are developed. A lower limit on wind noise is estimated from two theories of the intrinsic turbulent pressure fluctuations. Empirical results for the self-noise windscreens in substantially non-turbulent flows are also presented. Measurements of the wind velocity spectra and wind noise spectra from a variety of windscreens are described and compared to the theoretical predictions. All of the wind noise data lies between the upper and lower limits. The theoretical framework allows windscreens to be evaluated in terms of the best and worst-case scenarios and establishes practical lower limits on wind noise reduction for varying wind conditions. [Work supported by the Collaborative Technology Alliance sponsored by the US Army Research Laboratory.]

8:15

4aPAb2. Nighttime traffic noise in an urban environment. Kenneth E. Gilbert, Roger Waxler, Carrick L. Talmadge, and James P. Chambers (Natl. Ctr. for Physical Acoust., Univ. of Mississippi, University, MS 38677, kgilbert@olemiss.edu)

Traffic noise ducted near the ground at night and in the early morning can be represented simply in terms of normal modes. The effects of surface loss and atmospheric absorption, as well as meteorology, are contained in the mode attenuation coefficients. For a long (5–10 km) road section, the average noise levels are a function of the product of the mode attenuation coefficients and the perpendicular distance from the roadway. This paper considers a two-dimensional model for computing the effective mode attenuation coefficients in an urban environment. Modal attenuation is estimated by propagating modes over an irregular surface that approximates typical urban structures. The mode attenuation due to scattering from the urban structures is compared to the attenuation from atmospheric absorption and from a finite ground impedance. Predicted noise levels are compared to available data and the implications for nighttime urban traffic noise are discussed.

8:30

4aPAb3. The stability of the nocturnal quiet height. Roger Waxler, Kenneth E. Gilbert, (NCPA, U. of Mississippi, University, MS 38677, rwax@olemiss.edu), and Carrick Talmadge (NCPA, University, MS 38677)

It has recently been demonstrated that for narrow band signals propagating in the nocturnal boundary layer there is a height, a few meters off the ground, at which the sound pressure is significantly reduced. In this presentation the stability of the quiet height against fluctuations of the sound speed will be discussed. It is demonstrated theoretically, and verified by experiment, that, in the first few meters of the atmosphere, the dependence of the sound pressure level on altitude, relative to the sound pressure level at any fixed altitude, is insensitive to sound speed fluctuations.

8:45

4aPAb4. Wind noise at a flush microphone in a flat plate. Kevin Dillion, Richard Raspet, and Jeremy Webster (Dept. of Phys. and Astro., Univ. of Mississippi, P.O. Box 1848, University, MS 38677)

Elliot [Boundary Layer Meteorology **2**, 476–495 (1972)] states that a flush-mounted microphone can measure the turbulent pressure fluctuations in outdoor flows without self-noise problems. This suggests that flush mounted microphones can be used to minimize wind noise. Wind velocity spectra from a hot wire anemometer and wind noise spectra from a flush-mounted microphone in a flat plate at ground level have been measured outdoors. The velocity power spectral densities and average velocities as a function of height are determined from the data. The measured pressure power spectral densities are compared to theoretical values calculated from fluid dynamic and meteorological turbulence theory. In particular we investigate the contributions of predicted turbulence-turbulence and turbulence-mean shear contributions to the pressure spectrum measured at the surface of the plate. [Prepared in part through collaborative participation in the Collaborative Technology Alliance for Advanced Sensors sponsored by the US Army research Laboratory under Cooperative Agreement DAAD19-01-0008.]

9:00

4aPAb5. Spatial correlations in fluctuations induced by long-range propagation of sound in the atmosphere. Carrick L. Talmadge, Shantharam Dravida, Kenneth E. Gilbert, and Roger Waxler (Univ. of Mississippi, NCPA, Oxford, MS 38677, clt@olemiss.edu)

Amplitude and phase fluctuations associated with sound propagating over long distances are well known to increase rapidly with distance [e.g., Wilson, Noble, and Coleman, J. Atmos. Sci. **60**, 2473 (2003)]. This increase in the fluctuation level is usually given as an argument that the sound source becomes temporally incoherent after propagating large distances, making signal processing techniques that rely on the temporal coherence of the signal impracticable for single microphone measurements. Results are presented from a series of experiments designed to measure the spatial coherence of sound in both day- and nighttime conditions. In these experiments, 24-element horizontal and 8-element vertical arrays were used to measure the coherence of sound sources over the frequency range 30–200 Hz (horizontal array) and 100–500 Hz (vertical array). In many of these experiments, surprisingly large spatial coherences in the source strength were observed. These results suggest that array measurements may allow us to compensate for the poor temporal coherence of the signals for long-range propagated sound. We will also discuss the different sources of the observed fluctuations in the sound level and phases for daytime versus nighttime conditions. These sources include atmospheric turbulence, changes in the nocturnal boundary layer height and nocturnal gravity waves.

9:15

4aPAb6. Numerical simulation of acoustic tomography of the atmosphere. Vladimir E. Ostashev, Sergey N. Vecherin, George H. Goedecke (Phys. Dept., New Mexico State Univ., Las Cruces, NM 88003, vostashe@nmsu.edu), D. Keith Wilson (U.S. Army Engineer Res. and Development Ctr., Hanover, NH 03755), Alexander G. Voronovich (NOAA/ETL, Boulder, CO 80305), and Edward G. Patton (Natl. Ctr. for Atmospheric Res., Boulder, CO 80307)

A state-of-the-art array for acoustic travel time tomography of the atmosphere is under construction by several organizations in the U.S. The array will allow the estimation of the temperature and wind velocity fields within a tomographic volume located a few meters above the ground with a horizontal size of about 80 m. This paper is devoted to numerical simulation of acoustic travel time tomography of the atmosphere. The temperature and velocity fields within the tomographic volume are modeled with the use of quasi-wavelets and Large Eddy Simulation. Then, the travel times of sound propagation between different pairs of sources and receivers of the tomography array are calculated. Given these travel times, three algorithms for reconstruction of the temperature and velocity fields are employed: the cell approach, basis function approach, and stochastic approach. It is shown that the first of these approaches allows us to reliably estimate the mean temperature and velocity vector within the tomographic volume, while the other two give a good reconstruction of fluctuations in the temperature and velocity with respect to their mean values. [Work supported by ARO, Grants DAAD19-03-1-0104 and DAAD19-03-1-0341.]

9:30

4aPAb7. Modeling pulse propagation in the nocturnal boundary layer. Roger Waxler, Kenneth E. Gilbert (NCPA, Univ. of Mississippi, University, MS 38677, rwax@olemiss.edu), Sergey Kulichkov (Russian Academy of Sciences), and Carrick Talmadge (NCPA, University, MS 38677)

The propagation of broad band (0 to 1000 Hz) pulses in downward refracting atmospheres over lossy ground is modeled in the time domain by Fourier synthesis using a modal model for sound propagation in the frequency domain. Of particular interest is the dispersion of the pulse as it propagates. A conjecture of Chunchuzov, Bush, and Kulichkov [J. Acoust. Soc. Am. **88**, 455–461 (1990)] that at long ranges such pulses have a universal form, is shown to be true. In general, at long ranges from the source, the pulse develops a narrow band tail, centered around 20 or 30 Hz. This tail is formed by the superposition of surface modes.

9:45

4aPAb8. Prediction of aerodynamically generated sound via 3D particle velocity measurements. Sean Wu, Aditya Kumar, Zhi Ni (Dept. of Mech. Eng., Wayne State Univ., 5050 Anthony Wayne Dr., Detroit, MI 48202), and Hans-Elias de Bree (Microflown Technologies, AH Zevenaar, The Netherlands)

It has been shown [Wu and Hu, J. Acoust. Soc. Am. **103**, 1763–1774 (1998)] that acoustic radiation from a vibrating object can be calculated directly by an Alternate Integral-formulation Method (AIM), once the particle velocity distribution over a hypothetical surface enclosing this object is specified. This concept is extended here to prediction of aerodynamically generated sound. The particle velocity distribution on a hypothetical enclosure is measured by a fully integrated sound pressure-velocity (p-u) probe. This novel p-u sensor combines a MEMS particle velocity sensor and miniature pressure transducer, which allows for measurements of the acoustic pressure and particle velocity at a single point simultaneously. It is simple and easy to use and is suitable for both narrow and broadband sound measurements. This p-u probe is employed to measure the particle velocity field generated by a typical hairdryer. Measurements thus obtained are taken as input to AIM to predict the radiated acoustic pressure field. Experiments are conducted inside an anechoic chamber at the Acoustics, Vibration, and Noise Control Laboratory at Wayne State University. The predicted acoustic pressures are then compared with those measured at the same locations by microphones. Satisfactory agreements are obtained at all frequencies.

10:00

4aPAb9. A comparison of conventional and metal foam windscreens. Gunnar R. Becker, Guenther H. Hermstruewer, and Rainer Knoetsch (Rheinmetall Defence Electron. GmbH, Brueggeweg 54, 28309 Bremen, Germany)

In outdoor applications wind induced noise reduces the signal quality significantly. Conventional windscreens made of polyurethane (PU) foam are widely used to reduce this noise. However under harsh environmental conditions such as high temperatures, exposition to UV radiation, etc., PU may suffer. An alternative may be the usage of metal foams instead of PU foams. Spherical windscreens made of open-pored metal foam with dimensions close to commercial ones and different pore distributions have been manufactured by m-pore and were tested under different wind conditions. Comparative measurements between both conventional PU- and metal foam windscreens indicated that the reduction of wind induced noise is almost the same. A reduction of the receiver level due to the windscreens was not observed. Measurements and test results will be presented. [Work supported by m-pore.]

THURSDAY MORNING, 19 MAY 2005 PLAZA A, 10:45 A.M. TO 12:00 NOON

Session 4aPAc

Physical Acoustics: Thermoacoustics and Particle Agglomeration

Matthew E. Poese, Chair
Applied Research Lab., Pennsylvania State Univ., State College, PA 16804

Contributed Papers

10:45

4aPAc1. Specific acoustic impedance measurements of a thermoacoustic stack. Heui-Seol Roh, Richard Raspet, and Henry E. Bass (Dept. of Phys. and Natl. Ctr. for Physical Acoust., Univ. of Mississippi, University, MS 38677)

Simmons [Ph.D. dissertation, University of Mississippi, 2003] attempted to use a two-microphone specific acoustic impedance (SAI) measurement of a thermoacoustic stack to determine the thermoviscous functions and the coefficient of the thermal gradient in the acoustic pressure differential equation for arbitrary geometry stacks. The proposed method was tested in part by comparing the measured and predicted SAI of a well-characterized parallel pore stack. Significant disagreement between theory and measurement occurred. In this paper the results of an improved four microphone SAI measurement on the well-characterized stack are reported. This measurement will reduce uncertainties in the transfer matrix method used by Simmons and determine if the discrepancies in Simmons work were experimental or if the thermoacoustic theory is incomplete.

11:00

4aPAc2. High frequency operation of thermoacoustic coolers and prime movers. Husam El-Gendy, Young Sang Kwon, and Orest G. Symko (Dept. of Phys., Univ. of Utah, 115 S. 1400 E., Rm. 201, Salt Lake City, UT 84112-0830)

By operating thermoacoustic engines at high frequencies, 4 kHz and higher, the devices have characteristics which are important for many applications. Since they are resonant systems, the power density increases with frequency. Reduction of device size provides quick thermal response time in both the cooler and the prime mover. Moreover, small device size makes it practical to incorporate them into arrays, which can handle large powers. Most important is the fact that small devices make it simple for operation at high pressures in working gas without exceeding strength of materials limitations. This leads to high power densities. Results will be presented to illustrate how the above features affect device performance for the frequency range of 4 kHz to 21 kHz. Measurements using Particle Image Velocimetry of streaming, instabilities, and resonator mode interactions will be discussed for this high frequency range. Ultimately as the operating frequency is raised, device efficiency is limited by heat conduction along the stack and working fluid. [Work supported by the Office of Naval Research and the State of Utah.]

11:15

4aPAc3. Initial investigations into thermoacoustic coal agglomerators. Gordon Smith (Dept. of Phys. and Astonomy, Western Kentucky Univ., 1 Big Red Way, Bowling Green, KY 42101)

Each year, Kentucky coalmines produce about 160 million tons of coal, mainly used to generate electricity at power plants in Kentucky and throughout the midwestern and southeastern United States. Coal processing generates an excessive amount of pollution in the form of heavy exhaust, which creates an environmental concern. In response to this growing concern for a balance between environmental impact and energy production, many processing plants utilize electrostatic precipitation (ESP) systems to agglomerate and filter out particulate matter (PM) from exhaust streams before they enter the general environment. However, ESP efficiency is limited by the size of the particulate exhaust. Particle agglomeration also results via time-averaged forces inherent in an acoustic standing wave. Thermoacoustic technology, which utilizes waste heat sources for operation, embodies an ideal source for providing the requisite acous-

tic conditions. This technology could also be installed to supplement existing technology as a preconditioner (increasing the particle size to one more suitable for current ESP systems), and increase the overall PM removal efficiency. This talk will present initial investigations into applying thermoacoustic technology to this problem.

11:30

4aPAc4. Experimental study of aerosol concentration in flow-through, low-frequency resonators. Douglas Meegan, Justin Smith, and Wayne Wright (Appl. Res. Labs., Univ. of Texas, Austin, TX 78713-8029)

An experimental method to study the concentration of micron-sized aerosols in a low-frequency sound field was developed. The physical mechanism for acoustic aerosol concentration in a low-frequency sound field is the asymmetric Stokes drag which has been predicted to exceed the effects of radiation pressure for typical aerosol sizes [Meegan and Ilinskii, J. Acoust. Soc. Am. **114**, 2387 (2003)]. Related experiments were conducted in which 5 micron aerosol was drawn through a small duct that was driven in a transverse resonant mode with peak sound pressure levels of greater than 150 dB re 20 microPascals and frequencies in the range 1 to 10 kHz. The aerosol stream was illuminated by a laser sheet through the transparent walls of the duct in order to visualize (by video) the effects of the sound field. The experiments confirm the basic feature of the asymmetric Stokes drag model—specifically, the aerosol is observed to concentrate along acoustic velocity nodes. Ongoing experiments to quantify the level of concentration will be described. [Work supported by RDECOM.]

11:45

4aPAc5. The effects of orthokinetic collision and the acoustic wake effect on acoustic agglomeration of polydisperse aerosols. Shaozeng Dong (Dept. of Mech. Eng., Virginia Commonwealth Univ., 601 West Main St., Richmond, VA 23284), Bart Lipkens (Western New England College, Springfield, MA 01119), and Timothy Cameron (Kettering Univ., Flint, MI 48504)

A new concept of effective agglomeration length, which measures the maximum particle separation distance for effective collisions, is proposed for acoustic agglomeration of polydisperse aerosols with respect to the separate and combined effects of orthokinetic collision and acoustic wake in a horizontal acoustic wave. Particle gravity is found to be significant for the acoustic wake effect while the particle collision efficiency is important for the orthokinetic collision. Results indicate that orthokinetic collision dominates at low frequencies for intermediate size ratios while the acoustic wake effect is more significant at higher frequencies for all particles. The optimum frequency for orthokinetic collision is confirmed but shifts downward with the increase of sound power. For the acoustic wake effect, the agglomeration increases monotonically with sound frequency. Results also show that the orthokinetic collision is not effective for agglomeration of sub-micron particles because of low particle collision efficiency.

THURSDAY MORNING, 19 MAY 2005 REGENCY C, 8:00 A.M. TO 12:00 NOON

Session 4aPP

Psychological and Physiological Acoustics: Temporal Factors, Masking and Pitch (Poster Session)

Magdalena Wojtczak, Chair

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Contributed Papers

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

4aPP1. The temporal effect for signal frequencies around a notched cochlear hearing loss. Elizabeth Strickland and Lata Krishnan (SLHS Dept., Purdue Univ., W. Lafayette, IN 47907)

The temporal effect (or overshoot) refers to the change in signal-to-masker ratio at threshold for a short-duration tone as it is delayed from the onset of a longer-duration masker. This study is one in a continuing series of studies on the relationship between the temporal effect and the cochlear active process. In a previous study, the temporal effect was measured as a function of signal level in a broadband masker, for listeners with cochlear hearing loss. Results showed that the pattern of the temporal effect with level depended not only on the degree of hearing loss at the signal frequency, but also on the degree of hearing loss above the signal frequency. In the present study, this was examined further by measuring the temporal effect as a function of signal level in listeners with notched cochlear hearing loss, when the signal frequency was below, at or above the frequency of the greatest hearing loss (the notch). Results will be analyzed in terms of changes in the amplification due to the cochlear active process at and above the signal frequency. [Work supported by the Kinley Trust.]

4aPP2. Effect of frequency on the relationship between intensity discrimination and the detection of amplitude modulation. Rebecca E. Millman and Sid P. Bacon (Dept. of Speech and Hearing Sci. Arizona State Univ., P.O. Box 870102, Tempe, AZ 85287-0102, rebecca.millman@asu.edu)

Wojtczak and Viemeister [J. Acoust. Soc. Am. **106**, 1917–1924 (1999)] found a clear relationship between the detection of intensity increments and the detection of amplitude modulation (AM) for continuous 1-kHz sinusoids. In the present study AM detection thresholds and intensity difference limens (DLs) were measured for 1-s gated sinusoids with a frequency of 125 Hz or 1 kHz. The sinusoids were presented at levels between 10 dB SL and 90 dB SPL. Modulation detection thresholds [$20 \log(m)$] improved as the carrier level was increased; for a comparable range of levels (45 dB), the improvement was similar at the two frequencies. Intensity DLs [$10 \log(\Delta I/I)$] generally improved as the level of the reference sinusoid was increased, although the improvement (“near miss” to Weber’s Law) was smaller at 125 Hz than at 1 kHz over the same 45-dB range of levels. For the 1-kHz sinusoids, the relationship between intensity DLs and modulation detection thresholds was consistent with that shown by Wojtczak and Viemeister (1999). This same relationship apparently does not hold at 125 Hz. Because the relationship between intensity dis-

crimination and modulation detection depends upon frequency, the mechanisms underlying the two tasks may be somewhat different. [Work supported by NIDCD.]

4aPP3. Level effects in amplitude modulation tuning. Magdalena Wojtczak and Neal F. Viemeister (Univ. of Minnesota, 75 East River Rd., Minneapolis, MN 55455, wojtc001@umn.edu)

Psychophysical experiments on amplitude-modulation (AM) masking suggest the existence of channels selectively sensitive to different ranges of modulation rates. This study investigates whether tuning depends on carrier level. Masking of a 40-Hz signal AM was measured as a function of the masker-AM rate selected from the range between 4 and 256 Hz, for Gaussian-noise and 5.5-kHz carriers. For the noise carrier, similar patterns were observed at two different spectrum levels (25 and 40 dB SPL), leading to similar estimates of the modulation-filter bandwidths. For the tonal carrier, however, tuning in AM masking varied considerably across levels. At the lowest level tested (40 dB SPL), the patterns were very broad and highly asymmetric. The patterns became more similar across carrier levels when a highpass noise was presented with the carrier. The noise limited the use of spread of excitation and thus, raised thresholds for the unmasked signal detection. The data suggest that the AM selectivity is strongly affected by the detection threshold for the unmasked AM. [Work supported by NIH grants DC00683 and DC006804.]

4aPP4. Comparison of intensity discrimination, increment detection, and comodulation masking release in the envelope and audio-frequency domains. Paul C. Nelson (Dept. of Biomed. & Chemical Eng. and Inst. for Sensory Res., Syracuse Univ., 621 Skytop Rd., Syracuse, NY 13210, pcnelson@syr.edu), Stephan D. Ewert (Tech. Univ. of Denmark, 2800 Kgs. Lyngby, Denmark), Laurel H. Carney (Syracuse Univ., Syracuse, NY 13210), and Torsten Dau (Tech. Univ. of Denmark, 2800 Kgs. Lyngby, Denmark)

In the audio-frequency domain, the envelope apparently plays an important role in detection of intensity increments and in comodulation masking release (CMR). The current study addressed the question whether the second-order envelope (''venelope'') contributes similarly for comparable experiments in the envelope-frequency domain. One set of experiments examined the relationship between gated intensity discrimination and continuous-carrier increment detection. In contrast to the asymmetry observed in the audio-frequency domain (listeners are more sensitive to increments), AM-depth discrimination thresholds were found to be the same in conditions with a continuous (modulated) carrier and with traditional gated stimuli for AM frequencies ranging from 4–64 Hz. The second set of experiments compared the amount of CMR in a tone-in-noise detection task when slow, regular fluctuations were imposed on the masking waveform in both domains. A significant release from masking of a 32-Hz signal in the modulation frequency domain was obtained only when the venelope fluctuations were slower than 1–2 Hz. Both experiments suggest a relatively weak contribution of venelope cues in the AM domain when compared to those provided by envelope cues in the spectral domain. [Work supported by NIH-NIDCD R01001641 (PCN, LHC) and the Danish Research Council (SDE, TD).]

4aPP5. Temporal integration functions of amplitude modulation detection and amplitude modulation depth discrimination. Jungmee Lee and Glenis Long (Speech and Hearing Sci., The Grad. School and Univ. Ctr., The City Univ. of New York, 365 5th Ave., New York, NY 10016)

Previous studies (Lee and Green, 1994; Lee and Bacon, 1997) suggested that both AM rate and AM depth discrimination were influenced by the number of AM cycles, instead of the duration of stimuli. AM detection and AM depth discrimination (standard depth=0.1) were measured as a function of the number of AM cycles for modulation rates of 10, 20, 40, 80, 125, 160, and 320 Hz. Different numbers of modulation cycles were used for each modulation rate: 2, 4, or 8 for 10 Hz; 2, 4, 8, or 16 for 20 Hz; 2, 4, 8, 16, or 32 for 40 Hz; 2, 4, 8, 16, 32, or 64 for 80, 125, 160, and 320 Hz. The carrier was a broadband-noise (10 kHz lowpass), and the carrier was either gated with the modulator or presented 250 ms earlier and 250 ms later than the modulator. The overall level of each presentation was randomized within 6-dB range from 65 dB SPL. The results suggest that there might be different temporal integration processes for AM detection and AM depth discrimination. The pattern is different for lower and higher modulation rates. [Work was supported by NIDCD Grant No. R03 DC06605-01.]

4aPP6. Coherent modulation enhancement: Improving performance in noise for hearing aids and cochlear implants. Pamela Souza (Dept. of Speech and Hearing Sci., Univ. of Washington, 1417 NE 42nd St., Seattle, WA 98105), Les Atlas, Steven Schimmel (Univ. of Washington, Seattle, WA 98105), Jay Rubinstein, Ward Drennan, and Jong Ho Won (Univ. of Washington, Seattle, WA 98105)

Difficulty hearing in noise is a pervasive problem for hearing-impaired listeners. One approach is to digitally reduce noise using a modulation filter, which can selectively modify the time envelope of selected spectral bands. Previous approaches to modulation filtering, such as those based upon a Hilbert transform magnitude, increased signal distortion. A new coherent approach was used to determine a fixed and an optimal adaptive modulation filter. The optimal adaptive filter used information from the target speech to design the modulation filter. Threshold signal-to-noise ratio was measured adaptively using a spondee-in-noise task [Turner *et al.*, J. Acoust. Soc. Am. **115**, 1729 (2004)]. Subjects selected the spondee heard in a forced-choice task with a two-talker babble background. The spondee level was fixed at the listeners most comfortable level and noise level adjusted adaptively using a 2 dB step size, with threshold based on 10 reversals. Sensorineural hearing-impaired listeners completed three conditions: unprocessed speech; fixed modulation filter; and optimal adaptive modulation filter. Normal-hearing subjects heard a parallel set of conditions processed to simulate a 6-channel cochlear implant. There was significant improvement in threshold signal-to-noise ratio with the optimal adaptive filter for both groups. There was no improvement for the fixed modulation filter relative to unprocessed speech.

4aPP7. Factors affecting gap duration discrimination performance. John H. Grose, Joseph W. HallIII, and Emily Buss (Dept. Otolaryngol.-HNS, Univ. of North Carolina at Chapel Hill, 1115 Bioinformatics CB#7070, Chapel Hill, NC 27599-7070, jhg@med.unc.edu)

This investigation tests the hypothesis that reduced acuity in temporal tasks is evident relatively early on in the aging process for tasks involving brief stimuli or intervals, across-frequency processing, and/or significant processing loads. In an earlier study of gap duration discrimination (GDD) using fixed 20-ms markers, it was found that young listeners performed significantly better than middle-aged listeners for both within-channel and across-channel configurations. However, this age effect did not emerge in a subsequent study that employed random-duration markers. The purpose of the present study was to clarify stimulus conditions under which early aging effects are evident in GDD tasks. Normal-hearing young (20–25 years) and middle-aged (40–55 years) listeners were tested using both fixed- (20-ms) and random- (40 ms$\pm$50%) duration markers in configurations that were either within-channel or across-channel. In addition, task complexity was manipulated by incorporating a rhythm discrimination feature into the GDD task. Results to date indicate that performance declines when marker duration varies randomly, both for within- and across-channel conditions, as well as when task complexity is increased. These findings will be discussed in the context of identifying early aging effects in temporal processing. [Work supported by NIDCD DC001507.]

4aPP8. Fundamental frequency discrimination and harmonic resolvability in sensorineural hearing impariment. Joshua G. W. Bernstein and Andrew J. Oxenham (MIT Res. Lab. of Electrons. and Harvard-MIT Speech & Hearing Bioscience & Technol. Prog., 77 Massachusetts Ave., Cambridge, MA 02139, jgbern@mit.edu)

Sensorineural hearing loss (SNHL) often results in impaired fundamental frequency ($F0$) processing. Three experiments tested the hypothesis that this deficit is related to a loss of frequency selectivity, which may result in fewer peripherally resolved harmonics. $F0$ difference limens ($F0$DLs) were measured as a function of $F0$ for bandpass filtered sine- and random-phase harmonic complexes in listeners with mild-to-moderate SNHL. All listeners showed a transition between small (good) $F0$DLs at high $F0$s and large (poor) $F0$DLs at low $F0$s, although the transition point varied across subjects. Two measures that are thought to reflect frequency selectivity generally corresponded to the $F0$DL transition point: the maximum $F0$ for which $F0$DLs were phase-dependent, and the minimum modulation frequency required to discriminate amplitude modulation from quasi-frequency modulation. These results provide some support for the idea that the $F0$-processing deficit associated with SNHL is related to poor frequency selectivity. However, contrary to our hypothesis, poor correspondence was observed between the $F0$DL transition point and auditory filter bandwidths estimated using the notched-noise method in the same subjects. This suggests a possible discrepancy between traditional spectral-masking estimates of frequency selectivity and estimates based on temporal interactions between unresolved components. [Work supported by NIH grants R01DC05216 and 5T32DC00038.]

4aPP9. Modeling comodulation masking release using an equalization cancellation mechanism. Tobias Piechowiak, Stephan D. Ewert, and Torsten Dau (Ctr. for Appl. Hearing Res., Oersted-DTU, Tech. Univ. of Denmark, Bldg. 352, 2800 Kgs. Lyngby, Denmark, tda@oersted.dtu.dk)

Across-filter comparisons of temporal envelopes are a general feature of auditory pattern analysis which plays an important role in extracting signals from noise backgrounds. One example where a comparison of the temporal envelope in different frequency regions can lead to a substantial facilitation of detecting a signal in noise is comodulation masking release (CMR). It has been suggested that in (monaural) across-channel processing, like in (binaural) across-ear processing, an equalization-cancellation (EC) based circuit might be an effective strategy of noise reduction. The first part of the study investigates the relation between CMR and envelope-based binaural masking level differences (BMLD), using narrowband noise maskers and classical across-channel configurations (like N0Spi, N0Sm). In the second part, a model is presented that explicitly simulates CMR whereby the EC mechanism is assumed to take place at the output of a modulation filterbank. In the case of several flanker bands, the equalization is realized effectively by averaging the flanker bands, and cancellation by subtracting the averaged representation from the signal band. A generalized version of the model is presented that considers both across-channel and within-channel contributions to signal detection in CMR. [Work supported by the Danish Research Council.]

4aPP10. A unified view of the temporal-window and the adaptation-loop model in conditions of forward- and simultaneous masking. Ole Hau, Stephan D. Ewert, and Torsten Dau (Ctr. for Appl. Hearing Res., Oersted DTU, Tech. Univ. of Denmark, Bldg. 352, 2400 Kgs. Lyngby, Denmark, se@oersted.dtu.dk)

Two different mechanisms have been discussed in the literature as a possible explanation for forward masking: persistence (or temporal integration) and adaptation. In this study, two well established models of temporal processing in the auditory system are compared in a unified modeling framework. The temporal-window model representing a temporal-integration mechanism and the adaptation-loop model as the representative for the adaptation mechanism. The unified modeling framework shared a compressive, non-linear auditory filter stage and a template-based (optimal detector) decision stage. Predictions from both models were obtained in conditions of forward masking at 1 and 4 kHz for masker-to-signal gaps of 0 to 160 ms, and in conditions of simultaneous masking, where the signal was temporally overlapping with the masker. The signal was a 10- or 12-ms raised-cosine ramped pure tone and the masker a 200-ms broadband noise. It was found that when both models are adjusted to behave similarly, the two mechanisms can be considered as being essentially equivalent. In fact, the combination of integration and the signal-to-noise-ratio based detection criterion in the temporal-window model can be interpreted as adaptation. Suggestions for a unified model of auditory processing are presented. [Work supported by the Danish Research Council.]

4aPP11. An examination of temporal signal uncertainty for sequences of noise or random-frequency maskers. Lori Leibold, Donna Neff, and Walt Jesteadt (Boys Town Natl. Res. Hospital, 555 North 30th St., Omaha, NE 68131, leiboldl@boystown.org)

The effect of temporal uncertainty was examined on detection of a 1000-Hz tone presented simultaneously with one of a sequence of five masker bursts, using a 2-AFC, adaptive procedure. The signal and each masker burst were 100-ms (10-ms rise/fall), with no temporal overlap between masker bursts. Each masker burst was 60 dB SPL. Across conditions, maskers were broadband-noise (300–3000 Hz) or random-frequency multi-tonal maskers with 1, 2, or 10 components. Components for the random-frequency maskers were drawn from 300 to 3000 Hz, excluding a 160-Hz band around the signal. Thresholds for conditions with no temporal uncertainty (signal presented with the first, third, or fifth masker burst on all trials) were compared to performance with maximal temporal uncertainty (signal position varied on every trial). There was no effect of temporal uncertainty for the noise masker. For random-frequency maskers, performance was uniformly poor for both fixed- and random-position signals, except for some release from masking for signals presented with the last masker burst. For three of four listeners, even 1-component maskers showed large amounts of masking in all conditions. These results suggest that the effects of temporal uncertainty are minimal for these stimuli. [Work supported by NIDCD.]

4aPP12. Spectral integration and mutliple looks. Robert Lutfi (Dept. of Communicative Disord. and Waisman Ctr., Univ. of Wisconsin, Madison, WI 53706)

A fundamental property of hearing is that signals become more detectable as their bandwidth is increased. The rate at which detection improves is reasonably well predicted by models that assume integration of energy at the output of a single auditory filter roughly matched in bandwidth to the signal. This paper tests an alternative account in which detection is mediated by not one but several independent auditory filters whose outputs are combined (multiple-looks model). Listeners detected an increment in the level of a multitone complex whose bandwidth was increased by adding tones at successively higher and lower frequencies. In different conditions the frequencies were fixed (F) or were perturbed at random from trial to trial with the perturbation being the same (PS) or different (PD) for each frequency. The multiple-looks model predicts performance should improve with bandwidth at a faster rate for PD and at a slower rate for PS compared to F; this due to differences in the shared variance in the output of separate auditory filters. The single filter model predicts little effect of frequency perturbation in either case. Individual differences in the results provided partial support for both models. [Work supported by NIDCD.]

4aPP13. Response growth with and without a low-frequency suppressor. Jeffrey J. DiGiovanni (School of Hearing, Speech and Lang. Sci., Ohio Univ., Athens, OH 45701) and Magdalena Wojtczak (Univ. of Minnesota, Minneapolis, MN 55455)

In a psychophysical study, Wojtczak and Viemeister (2005) demonstrated that the response to a tone suppressed by a fixed-level, higher-frequency suppressor grows faster than the response to the same but unsuppressed tone. This finding is consistent with a linearization of the response under suppression. This study extends their experiment to suppressors with frequencies below that of the suppressee. Detection of a 10-ms, 4-kHz probe was measured under two forward-masking conditions: one with a 200-ms, 4-kHz masker presented alone, and the other with the same masker/probe paired with a 2.4-kHz fixed-level suppressor. A range of probe levels was used to measure growth of masking with and without the suppressor. The 4-kHz masker level was varied adaptively to find the masked threshold. Suppression was revealed by the difference between the masker levels needed for the masked threshold in the presence and absence of the suppressor. Initial comparison of the rates of masking growth between the two conditions suggests a linearization of the response to the 4-kHz masker in the presence of a lower-frequency suppressor. A few levels of the suppressor were used to determine the rate of suppression growth and its effect on the growth of response at the suppressee-frequency place.

4aPP14. Stochastic resonance in Gaussian and uniform noise. Dennis Ries (School of Hearing, Speech, and Lang. Sci., Ohio Univ., Grover Ctr. W221, Athens, OH 45701, ries@ohio.edu)

The improvement in threshold of a 2.0 kHz tone in the presence of low levels of uniform and Gaussian noise is compared to tonal threshold measured in quiet. Uniform noise was created using a rectangular amplitude distribution. The noises were low-pass filtered at 12.0 kHz and were presented continuously throughout a run at spectrum levels ranging from 0 to -30 dB/Hz. Pure-tone signals were 400 ms in duration including 10 ms raised cosine onset and offset ramps. Thresholds were obtained using a 3-interval, forced-choice procedure with correct answer feedback in conjunction with a two down, one up adaptive tracking paradigm that targets 70.7% correct on the psychometric function. Subjects were instructed to select the interval that was different from the others. Preliminary results for the Gaussian noise conditions are similar to earlier results [Zeng *et al.*, Brain Res. **869**, 251–255 (2000)]. Initial results for the uniform noise indicate that it produced equivalent or slightly lower thresholds than those measured in the presence of Gaussian noise.

4aPP15. Spatial release from informational masking along the front-back dimension. Neil L. Aaronson (Dept. of Phys. and Astron., Michigan State Univ., 4230 BPS Bldg., East Lansing, MI 48824), Brad Rakerd (Michigan State Univ., East Lansing, MI 48824), and William M. Hartmann (Michigan State Univ., East Lansing, MI 48824)

When two independent speech samples are presented together from a single location in front of the listener, one will mask the other. The amount of masking can be reduced by presenting a repeated masker from a different location off to the side and shifting it slightly forward (+) or backward (−) in time compared to the masker in front. New experiments, using the coordinate response measure technique with a two-female-talker masker and a female target, show that masking release can also be obtained when the target and masker are in front and the repeated masker is directly in back. Release is seen for both forward and backward time shifts, ranging from -32 to 32 ms. The amount of release is somewhat more than half that obtained when the repeated masker is off to the side. Release from masking can also be seen when the repeated masker comes from a location directly above the target and masker in front, but only for a single value of time shift, namely ± 2 ms. It is concluded that both spatial and spectral cues mediate release from informational masking in the front-back dimension. [Work supported by the NIDCD grant DC 00181.]

4aPP16. The advantage of knowing where to listen. Gerald Kidd, Jr., Tanya L. Arbogast, Christine R. Mason, and Frederick J. Gallun (Commun. Disord. and Hearing Res. Ctr., Boston Univ., 635 Commonwealth Ave., Boston, MA 02215)

This study examined the importance of *a priori* knowledge about the location of a target in a multitalker environment. On each trial, three equal-level sentences from the Coordinate Response Measure test were presented from different loudspeakers separated by 60° azimuth. The sentences—target and 2 maskers—had the structure "Ready [callsign] go to [color] [number] now." The task was to repeat the color and number associated with a specific callsign. The target location was chosen among the 3 loudspeakers on every trial. The main experimental variables were the probability of occurrence of the target at each speaker location and whether the listener was provided with the callsign of the target before or after stimulus presentation. Performance ranged from near perfect when the target location was completely certain to about 33% correct when the location was random and the callsign was not known in advance. When target location was random, performance improved from about 33% to 70% correct when the callsign was known in advance. Overall, these results support the view that knowing where to focus attention provides a great advantage in speech recognition in complex and uncertain environments. [Supported by NIH/NIDCD.]

4aPP17. Effect of signal frequency uncertainty for random multi-burst maskers. Rong Huang and Virgina M. Richards (Dept. of Psych., Univ. of Pennsylvania, 3401 Walnut St., Ste. 302C, Philadelphia, PA 19104)

The detectability of a sequence of equal-frequency tone pips masked by random multi-burst complexes may depend on the perceptual segregation of the signal stream from the random masker. If so, detection thresholds may be independent of whether the signal frequency is known versus uncertain. In this experiment observers detected a signal stream of 8 sequential equal-frequency 30 ms tone pips embedded in a random masker composed of 8 sequential bursts. A yes/no procedure was used, and the independent variable p (probability a tone was played at a particular time-by-frequency location) governed the number of masker tones in each burst. The dependent variable was d. Threshold values of p were obtained for signal streams at 5 different frequencies. Sensitivity was superior for the mid-frequency signal, and decreased as the signal frequencies approached the edge of the frequencies the masker tones occupied (200–5000 Hz). When the frequency of the signal stream was randomly varied from trial to trial, sensitivity was poorer than for any of the fixed-frequency signals. Thus, the detectability of a sequence of tone pips is reduced when the signal frequency is uncertain compared to certain. Additionally, sensitivity increased when the signal stream was delayed relative to the masker bursts.

4aPP18. Training listeners or preserving phase information improves the effect of perceived spatial separation on releasing spectrally degraded Chinese speech from information masking. Jing Chen, Chenfei Ma, Chun Wang, Hongwei Qu (Natl. Key Lab. on Machine Percept. Speech and Hearing Res. Ctr., Dept. of Psych., Peking Univ., Beijing 100871, China), Xihong Wu, Liang Li (Peking Univ., Beijing 100871, China), and Bruce Schneider (Univ. of Toronto at Mississauga, Canada L5L 1C6)

Physical or perceived spatial signal/masker separation unmasks speech more when maskers are informational than when energetic. However, it is unclear how beneficial the separations are to cochlear-implant listeners, because signal transductions applied in cochlear implant degrade signals

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spectrally, and spectrally degraded speech is more vulnerable to maskers. Here, spectrums of both target speech (nonsense sentence) and masker (steady speech-spectrum noise, speech modulated speech C-spectrum noise, or speech) were filtered into 15 frequency bands. For both target and masking speech, the center-frequency pure tone of each band was modulated by the extracted envelope from the band. The target speech was composed by the sum of the 8 odd-band tones, and the masker was either same-band (with the 8 odd-band tones) or different-band (with the 7 even-band tones). The results show that physical but not perceived spatial separation unmasked target speech in naive normal-hearing listeners. However, following pre-presentations of both degraded and normal correspondent speech to listeners for a period of time or the introduction of phase information into modulated tones, perceived spatial separation reduced the influence of different-band speech masking but not that of same-band speech masking. These results are useful for improving cochlear-implant programs at both behavioral and technical levels.

4aPP19. Development of backward masking in elementary school children. Cynthia M. Zettler, Marsha G. Clarkson, Rose A. Sevcik, and Robin D. Morris (Dept. of Psych., Georgia State Univ., Atlanta, GA 30303)

Previous research on backward masking (BM) suggests that the ability develops slowly across childhood. When a spectral notch is placed in a noise masker, thresholds improve due to a decrease in masker energy at the frequency of the signal. To further clarify the developmental course of BM thresholds for BM and backward notched-noise (BM-N), conditions were measured in 81 children ranging in age from seven to ten years. BM stimuli consisted of a 20-ms, 1000-Hz tone presented 20 ms prior to a 300-ms, 600–1400 Hz bandpass noise. BM-N stimuli consisted of a 1000-Hz tone presented 20 ms prior to the onset of a 300-ms, 400–1600 Hz bandpass noise with a spectral notch between 800 and 1200 Hz. Multivariate analysis of variance indicated a significant difference among age groups for the BM condition ($F(3,77)=3.63$, $p<.05$), but not for the BM-N condition. Thresholds generally decreased with increasing age in the BM condition (age seven 87.86 dB; eight 81.36 dB; nine 75.31 dB, and ten 77.26 dB). Increased BM thresholds in younger children may indicate poorer processing efficiency, whereas the notched results suggest that frequency selectivity is relatively mature by age seven.

4aPP20. Tone-in-noise detection using narrowband reproducible maskers with restricted energy and envelope cues. Sean A. Davidson (Dept. of Biomed. and Chem. Eng., Inst. for Sensory Res., 621 Skytop Rd., Syracuse Univ., Syracuse, NY 13244, sadavids@syr.edu), Robert H. Gilkey (Wright State Univ., Dayton, OH), and Laurel H. Carney (Syracuse Univ., Syracuse, NY 13244)

Both energy- and envelope-based models have been successfully used to predict narrowband tone-in-noise detection thresholds. To distinguish between these models, hit and false-alarm rates for 25 reproducible maskers were measured in the N_0S_0 and N_0S_π interaural configurations and 4 cue conditions: (1) normal energy and envelope cues, (2) restricted envelope cues, (3) restricted energy cues, or (4) restricted energy and envelope cues. Preliminary results show that hit and false-alarm rates were strongly correlated between the conditions with and without restricted energy cues, indicating that the reduction of energy cues *did not* substantially affect the detection process. However, hit and false-alarm rates were more weakly correlated between the conditions with and without restricted envelope cues, suggesting that the reduction of envelope cues *did* affect the detection process. Restricting envelope cues also had a larger (but still modest) effect on d' under the N_0S_π configuration, but restricting energy cues had a larger effect for the N_0S_0 configuration. Overall, these results are more consistent with envelope-based models, but indicate that additional cues must play a role. [Work supported by NIDCD RO1-DC-001641 (LHC, SAD) and the Ohio Board of Regents (RHG).]

4aPP21. The spiral model of pitch: Interrelations with musical and psychoacoustic scales and with cochlear parameters. James D. Miller (Dept. of Speech and Hearing Sci., Indiana Univ., 200 S. Jordan Ave., Bloomington, IN 47405, jamdmill@Indiana.edu)

A spiral model of pitch can represent pitch chroma and pitch height. Its arclength provides the frequency-position map of the cochlea, and, when scaled by the number of auditory-ganglion cells, provides the Mel scale. Pitch-like scales fall into three groups. Group I is typified by the Cent scale, which can be expressed as a pitch spiral. Other Group-I scales are musical notation, logarithmic scales of frequency, Fechners Law for complex tones, and the pitch helix. Group II is typified by Greenwoods cochlear map, which is simply related to the arclength of the pitch spiral. Other Group-II scales are the sums of the number of (a) equivalent-rectangular bandwidths (NERBs); (b) difference limens for frequency (NDLFs); and (c) inner-hair cells (NIHCs) and are consistent with critical ratios. Group III is typified by the Mel Scale, which can be derived from the pitch spiral and the distribution of auditory-ganglion cells. Other Group-III scales are the sums of the number of (a) auditory ganglion cells (NSGCs), (b) frequency-modulation-difference limens (NFMDLs), and (c) Barks (NBARKs). The pitch spiral seems a basic concept from which other pitch-like scales can be derived.

4aPP22. Repetition "pitch" in chinchillas. William P. Shofner and William M. Whitmer (Parmly Hearing Inst., Loyola Univ. Chicago, 6525 N. Sheridan Rd., Chicago, IL 60626, wshofne@luc.edu)

Repetition pitches are evoked in human listeners by rippled noises. Infinitely iterated rippled noise (IIRN) is generated when wideband noise is delayed, attenuated, and added to the original wideband noise through positive (+) or negative (−) feedback. In human listeners, the pitch of IIRN(+) is matched to the reciprocal of the delay, whereas the pitch of IIRN(−) is an octave lower. A stimulus generalization paradigm was used to characterize IIRN repetition pitch in chinchillas. Chinchillas were trained to discriminate IIRN(+) with a 4-ms delay from IIRN(+) with a 2-ms delay. In the generalization task, chinchillas were tested with IIRN(+) having delays between 2–4 ms. The pitches evoked by these IIRNs ranged from 500–250 Hz. The delayed noise attenuation for all IIRNs was fixed at −1 dB. A systematic gradient in behavioral response occurred along the dimension of delay for each animal tested, suggesting that a perceptual dimension of pitch exists. Responses to IIRN(−) evoking pitches between 500–250 Hz (delays of 1–2 ms) were also measured. Responses to IIRN(−) were more variable among animals, suggesting that other perceptual cues such as timbre differences may be stronger than the pitch cues. [Work supported by NIDCD R01 DC005596.]

4aPP23. Influence of beats on mistuning detection in consonant and dissonant musical intervals. Craig E. Lewiston and Andrew J. Oxenham (MIT Res. Lab. of Electron. and Harvard-MIT Speech & Hearing Bioscience & Technol. Prog., 77 Massachusetts Ave., Cambridge, MA 02139, lewiston@mit.edu)

This study investigated the mechanisms underlying the detection of mistuning in musical intervals. Musical consonance is thought to be defined in part by an absence of cochlea-generated beats. Dissonant intervals, and mistuned consonant intervals, produce a percept of beats. Thus, mistuning of consonant intervals might be detected through the presence of beats, whereas mistuning of dissonant intervals might be detected by a change in the pattern of beats. Alternatively, mistuning might be detected by a direct comparison of the two pitches in each interval. In our experiment the presence of beats was controlled by presenting the two tones to both ears or to separate ears, by using either pure tones or complex tones, and by presenting the tones sequentially or concurrently. Results using highly trained musicians showed large threshold elevations (poorer performance) in the absence of cochlea-generated beats, suggesting that the mistuning of concurrent sounds in normal circumstances is mediated by beat detection. However, even in the absence of beats, mistuning detection

thresholds for dissonant intervals, such as the tritone or minor second, were poorer than for consonant intervals, such as the octave or fifth. [Work supported by NIH grants T32DC00038 and R01DC05216.]

4aPP24. Further explorations of the contribution of a nonsimultaneous mistuned harmonic to residue pitch. Hedwig E. Gockel, Robert P. Carlyon (MRC Cognition and Brain Sci. Unit, 15 Chaucer Rd., Cambridge CB2 2EF, UK, hedwig.gockel@mrc-cbu.cam.ac.uk), and Christopher J. Plack (Univ. of Essex, Wivenhoe Park, Colchester CO4 3SQ, UK)

Ciocca and Darwin [V. Ciocca and C.J. Darwin, J. Acoust. Soc. Am. **105**, 2421–2430 (1999)] reported a surprising finding: The shift in residue pitch caused by mistuning a harmonic was the same when the mistuned harmonic was presented after the remainder of the complex as when it was simultaneous. The present study tried to replicate this result, and investigated the role of the presence of the nominally mistuned harmonic in the matching sound. Subjects adjusted a matching sound so that its pitch equaled that of a subsequent 90-ms complex tone (12 harmonics of a 155-Hz $F0$), whose mistuned ($\pm 3\%$) 3rd harmonic was presented either simultaneously with or after the remainder. In experiment 1, the matching sound was a harmonic complex whose 3rd harmonic was either present or absent. In experiment 2, it was a sinusoid. In experiment 3, the target and matching sound contained non-overlapping harmonics. In all experiments, a non-simultaneous mistuned component produced significantly smaller pitch shifts than a simultaneous one. In the absence of the nominally mistuned harmonic in the matching sound, the pitch shift with simultaneous presentation was about five times larger than that with non-simultaneous presentation of the mistuned component. [Work supported by EPSRC Grant GR/R65794/01.]

THURSDAY MORNING, 19 MAY 2005 PLAZA B, 9:00 TO 11:45 A.M.

Session 4aSA

Structural Acoustics and Vibration and Musical Acoustics: Vibration of and Acoustic Radiation from Musical Instruments I

Courtney B. Burroughs, Cochair
Applied Research Lab., Pennsylvania State Univ., P.O. Box 30, State College, PA 16804-0030

Thomas D. Rossing, Cochair
Physics Dept., Northern Illinois Univ., De Kalb, IL 60115

Invited Papers

9:00

4aSA1. The banjo in the time-domain. Joseph Dickey (The Whiting School of Eng., Johns Hopkins Univ., 810 Wyman Park Dr. Ste. G010, Baltimore, MD 21211)

The banjo is under-represented in the musical acoustics literature. This is the case even though it is basically a vibrating and radiating membrane driven by a plucked string, and thereby lends itself naturally to modeling and analysis. A recent frequency domain model, [Dickey, J. Acoust. Soc. Am. **114**, 2958–2966 (2003)] will be reviewed. The work reported here is a time domain analysis. First, the response of an isolated, plucked, lossy and dispersive string is presented. The string is then connected through a bridge to the center of a circular membrane. The string and head are treated as two fully connected dynamic systems and the response of the head is calculated. The isolated head is then driven off-center with an harmonic drive and the temporal response is determined by ray-tracing and summing amplitudes over a grid. Resonance frequencies and modal patterns agree well with experiment thereby validating the technique. Finally, the resonant string is considered as an external, transient, off-center and spatially distributed drive and the temporal response of the head is derived using the ray tracing technique. These modal patterns evolve in time and eventually stabilize to patterns that are in qualitative agreement with experiment.

9:30

4aSA2. Structural vibrations and sound radiation fields of classical guitars. Bernard E. Richardson (School of Phys. and Astron., Cardiff Univ., 5 The Parade, Cardiff CF24 3YB, UK, RichardsonBE@cardiff.ac.uk)

In its few lines, Federico Garcia Lorca's poem ''Las Seis Cuerdas'' (the six strings) invokes a beautiful image of sound production on the guitar. It is the player's interaction with the six strings which makes the music, but were it not for the body of the guitar, the instrument would have no voice. Most studies of the acoustical function of guitars have concentrated on measuring the structural vibrations or the sound radiation fields in isolation of the strings. These studies fail to encapsulate the important aspects of string-body coupling, which has a marked influence on the decay rates of string vibrations, and the radiativity of individual body modes. By measuring both input admittance at the bridge and sound-field topology for individual modes of the guitar, it is possible, through psychoacoustical experiments, to investigate the relative importance of the various acoustical parameters which define the response of the instrument. This paper will discuss experimental techniques and present data on ten classical guitars which go some way to identifying key components in the low- to mid-frequency mechanical and acoustical actions of the guitar. [Work originally supported by the Leverhulme Trust.]

10:00–10:15 Break

10:15

4aSA3. Development of a composite material concert harp soundboard to match the structural acoustic performance of a wooden soundboard. Thomas J. Royston, Melinda J. Carney, Curt Preissner, and John Roxworthy (Acoust. & Vib. Lab., Univ. of Illinois at Chicago, 842 W. Taylor St., MC 251, Chicago, IL 60607, troyston@uic.edu)

The replacement of a Sitka spruce grand concert harp soundboard with a carbon fiber-reinforced plastic soundboard could provide improved durability and long-term stability, helping to mitigate stresses relating to fluctuating temperature and humidity that are inherent in wooden instruments. This presentation outlines the development of a composite soundboard by a combined experimental and computational approach. Experimental modal analysis data was compared with the results from a computational finite-element model to find the effective material properties of the multi-layered laminate wooden soundboard, treating the veneer, soundboard and varnish as one material. With the effective wood material properties determined, a composite test section was designed and fabricated based on specific matching criteria, duplicating the behavior of the wood with the composite laminate. The experimental natural frequencies and mode order of the composite test section closely matched the experimental results of the wood section within a 10% difference. Given these results, the method was then applied to design and manufacture a full composite soundboard. Various finite element models were created to develop the most practical design while sufficiently matching the wood soundboard properties. Issues regarding the implementation of the composite soundboard were also investigated.

10:45

4aSA4. Modes of vibration and sound radiation from percussion instruments. Thomas D. Rossing (Phys. Dept., Northern Illinois Univ., DeKalb, IL 60115, rossing@physics.niu.edu)

When a membrane or bar or plate is struck, it vibrates in a complex manner, which can be described in terms of normal modes of vibration. We describe the modes of vibration of percussion instruments, such as drums and bells, some ways in which the normal modes are observed, and the way in which they determine sound radiation from the instruments.

11:15

4aSA5. Interferometric studies of a piano soundboard. Thomas R. Moore (Dept. of Phys., Rollins College, Winter Park, FL 32789, tmoore@rollins.edu)

Ongoing efforts to understand and model the dynamics of the modern piano are hampered by a lack of understanding of the deflection shapes of the soundboard, which are extremely complicated due to the complex construction. Often deflection shapes of harmonically vibrating objects can be determined using holographic or speckle pattern interferometry, but in practice the stability necessary to implement these methods is difficult to achieve for a large wooden structure such as a piano. We show theoretically and experimentally that the deflection shapes of large objects that are typically too unstable for interferometry can be determined by modifying the common form of the electronic speckle pattern interferometer. Furthermore, using this modified interferometer the decorrelation of the speckle due to ambient vibrations actually enhances the precision of the interferogram. We discuss some interesting observations of the deflection shapes of the soundboard of a fully assembled piano, compare the deflection shapes of the lowest resonances with simple models, and demonstrate how the interferograms can be used to determine the driving point impedance.

Session 4aSC

Speech Communication: Advanced Methods in Speech Research and Speech Technology (Poster Session)

Terrance M. Nearey, Chair
Dept. of Linguistics, Univ. of Alberta, 4-32 Assiniboia Hall, Edmonton, AB T6G 2E7, Canada

Contributed Papers

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

4aSC1. Development of an anthropomorphic talking robot and the mimicking speech control. Kotaro Fukui, Kazufumi Nishikawa, Toshiharu Kuwae, Atsuo Takanishi (Dept. of Mech. Eng., Waseda Univ, 3-4-1 Ookubo, Shinjuku-ku, Tokyo, Japan), Hideaki Takanobu (Kogakuin Univ., Tokyo, Japan), Takemi Mochida (Commun. Sci. Labs., NTT, Kanagawa, Japan), and Masaaki Honda (Waseda Univ., Saitama, Japan)

We developed an anthropomorphic talking robot WT-4 (Waseda Talker No. 4) to produce human speech. WT-4 consists of 1-DOF lungs, 4-DOF vocal cords and articulators (the 7-DOF tongue, 5-DOF lips, 1-DOF teeth, nasal cavity and 1-DOF soft palate); the total DOF is 19. The lips and the tongue are made of elastic material to allow large deformation and to prevent from the air and sound leaks, and are controlled by a looped wire mechanism to form their various configurations. The talking robot enables to produce vowel and consonant sounds by mimicking the vocal cords vibration and the fricative and plosive source generation by the air flow as well as dynamically controlled vocal tract acoustic resonance in human speech production. Articulatory control of the talking robot is designed to track the acoustic goals (pitch, sound power, two formant frequencies, and voice-unvoiced timing) of the speech. The robot parameters are determined by minimizing the weighted mean squared error of these acoustic parameters between the human and robot speech sounds. It is shown that this mimicking speech control is effective in producing fluent continuous speech by the talking robot.

4aSC2. The tube resonance model speech synthesizer. Leonard C. Manzara (Dept. of Comput. Sci., Univ. of Calgary, 2500 Univ. Dr. NW, Calgary, AB, Canada T2N 1N4, manzara@cpsc.ucalgary.ca)

The Tube Resonance Model (TRM) synthesizer is an articulatory speech synthesizer implemented in software. It directly emulates the resonant behavior of the oropharyngeal and nasal tracts using digital waveguides. The oropharyngeal cavity is subdivided into 8 regions of unequal length, where particular regions correspond to the human articulators of tongue, teeth, and mouth. The radius (cross-sectional area) of each region can be varied independently over time. The differences in radii between regions gives rise to differences in acoustic impedance, which is modeled using two-way scattering junctions. The nasal cavity is composed of 5 equal-length sections, and is connected to the vocal tract via another section (the velum) using a three-way scattering junction. The total length of the tube can be varied over a continuous range, allowing one to synthesize male, female, and juvenile voices.

4aSC3. Study of effect of speaker variability and driving conditions on the performance of an automatic speech recognition engine inside a vehicle. Shubha Kadambe (HRL Labs., LLC, 3011 Malibu Canyon Rd., Malibu, CA 90265)

Spoken dialogue based information retrieval systems are being used inside vehicles. The user satisfaction of using such a system depends on how an ASR engine performs. However, the performance of an ASR is affected by speaker variability, driving conditions, etc. Here, we report the study that we performed to analyze these effects of speaker variability, different driving conditions and the effect of driving task on the ASR performance. This study consists of experimental design, data collection and systematically testing an ASR engine using this data. From the obtained results, it can be observed that (I) the ASR performance exhibits (a) significant speaker variability since the stress of driving task varies from speaker to speaker, (b) significant performance degradation across driving conditions since the noise type and level varies and (c) significant effect of driving task on recognition performance, and (II) the effect of live noise on recognition performance is not same as adding car noise to the prerecorded speech data. The former observation is important since by just training an ASR engine on lots of speech data will not help and it is essential to include stress factors and cognition load in ASR engines to improve its performance.

4aSC4. A noise-reduction strategy for speech based on phase-opponency detectors. Om Deshmukh (Dept. of Elec. and Comp. Engr. and Inst. for Systems Res., Univ. of Maryland, College Park, MD 20742), Michael C. Anzalone (Syracuse Univ., Syracuse, NY 13224), Carol Y. Espy-Wilson (Univ. of Maryland, College Park, MD 20742), and Laurel H. Carney (Syracuse Univ., Syracuse, NY 13224)

A noise-reduction algorithm was developed based on a neural model for detection that is robust in fluctuating noise. The phase-opponency (PO) neural model correlates the outputs of two different auditory filters that differ in phase by 180 degrees at the target frequency. The PO detector used here consists of a pair of overlapping bandpass filters with phase responses that differ by 180 degrees near the center frequency (CF). The correlation between the filter responses is reduced when a narrowband signal near CF is present in a noisy background. A bank of PO detectors was used to process speech corrupted by additive Gaussian noise. The time-varying outputs of the detectors can be post-processed to retain information-rich regions, such as formants and frication onsets, while greatly reducing noise between formants. The final detector output controlled the gains in a separate analysis/synthesis filterbank. Spectrograms of speech sounds before and after noise reduction illustrate the ability of the system to detect major features in speech down to low signal-to-noise ratios. The quality of the processed signal will be demonstrated. This system is intended as a front-end for speech recognition systems or in

hearing-aid noise-reduction algorithms. [Work supported by NSF BCS0236707 (OD,CYE) and NIH DC-001641 (MCA,LHC).]

4aSC5. A new method of extracting the filter characteristics of the nasal cavity using homorganic nasal-stop sequences. Hansang Park (Dept. of English Education, Hongik Univ., 72-1 Sangsu-dong, Mapo-gu, Seoul, Korea)

This study attempts to derive the filter characteristics of the nasal cavity of individual speakers. Since the only difference between a nasal and a homorganic voiced stop, such as [mb] and [nd], is whether the passage to the nasal cavity is open or not, the subtraction of the LPC spectrum of the voiced stop from that of the preceding nasal leads to the filter characteristics of the nasal cavity of an individual speaker regardless of place of articulation. The results showed that the spectral differences between samples of 20 ms taken from the steady states of the nasal and the following voiced stop were close to constant regardless of place of articulation, representing characteristic poles and zeroes, and that the spectral differences varied with speakers. This study is significant in that it provides a new method of extracting the filter characteristics of the nasal cavity, and that the spectral difference between a nasal and a homorganic voiced stop can be used as a parameter of the filter characteristics of the nasal cavity of individual speakers.

4aSC6. Synthesizing speech acoustics from head and face motion. Adriano V. Barbosa, Hani C. Yehia (CEFALA/PPGEE, Universidade Federal de Minas Gerais, Av. Antonio Carlos, 6627, Belo Horizonte, MG, 31270-010, Brazil, adriano.vilela@gmx.net), Andreas Daffertshofer (Vrija Universitat, Amsterdam, The Netherlands), and Eric Vatikiotis-Bateson (Univ. of British Columbia, Vancouver, BC, Canada V6T 1Z1)

This work outlines a quantitative analysis of the relation between speech acoustics and the face and head motions that occur simultaneously [A. V. Barbosa, Ph.D. thesis, Universidade Federal de Minas Gerais, Belo Horizonte, Brazil, 2004]. 2-D motion data is obtained by means of a video camera. An algorithm has been developed for tracking markers on the speaker's face from the acquired video sequence [A. V. Barbosa, E. Vatikiotis-Bateson, and A. Daffertshofer, in Proceedings of the 8th ICSLP Interspeech 2004, Korea, 2004]. The motion domain is represented by the 2-D marker trajectories, whereas line spectrum pairs (LSP) coefficients and the fundamental frequency $F0$ are used to represent the speech acoustics domain. Mathematical models are trained to estimate the acoustic parameters (LSPs + $F0$) from the motion parameters (2-D marker positions). The estimated acoustic parameters are then used to synthesize the acoustic speech signal. Cross-domain analysis for undecomposed (i.e., full head + face) and decomposed (i.e., separated head and face) normalized 2-D motions is performed. Syntheses from each method using intelligibility tests and qualitative comparison of the original and synthesized utterances are being evaluated.

4aSC7. ArtiSynth designing a modular 3D articulatory speech synthesizer. Florian Vogt, Oliver Guenther, Allan Hannam, Kees van den Doel (Univ. of British Columbia, 2356 Main Mall, Vancouver, BC, Canada V6T 1Z4, fvogt@ece.ubc.ca), John Lloyd, Leah Vilhan, Rahul Chander, Justin Lam, Charles Wilson, Kalev Tait, Donald Derrick, Ian Wilson, Carol Jaeger, Bryan Gick, Eric Vatikiotis-Bateson, and Sidney Fels (Univ. of British Columbia, Vancouver, BC, Canada V6T 1Z4)

ArtiSynth is a modular, component-based system for performing dynamic 3D simulations of the human vocal tract and face. It provides a test bed for research in areas such as speech synthesis, linguistics, medicine, and dentistry. ArtiSynths framework enables researchers to construct, refine, and exchange models of all parts of the vocal tract and surrounding structures. ArtiSynth introduces a probe concept to unify input and output data flow, which allows control of and access to models with time varying data series. ArtiSynth supports interconnected heterogeneous models, such as rigid body, mass-spring, and parametric, using a point-set connection method, called markers, for constraint satisfaction. Using ArtiSynth, we created a muscle-driven rigid body jaw model, a parametric principle component tongue model from MRI images, a parametric lip model, and mass-spring face tissue model. We combined them in various ways. Data from medical imaging (MRI, CT, and ultrasound) and other technologies such as optical tracking can be used to drive ArtiSynth models. We are currently developing an acoustical rendering framework supporting source-filter models and other advanced methods. The system incorporates a powerful scripting interface as well as an easy-to-use graphical interface. [Work supported by NSERC Canada and ATR Japan.]

4aSC8. Design of a 6 degree of freedom anthropomorphic robotic jaw. Edgar Flores and Sidney Fels (Dept. of Elec. & Comput. Eng., UBC, 2356 Main Mall, Vancouver, BC, Canada V6T 1Z4)

We have created a 6 DOF robotic jaw capable of producing, in real-time, the complex set of motions described by the human jaw during speech or mastication. The jaw is designed to fit within a larger robotic human figure such as the head, neck and torso of the 25 DOF Infanoid. [Kozima, Hideki: Infanoid: A Babybot that Explores the Social Environment, K. Dautenhahn *et al.* (eds.), Socially Intelligent Agents: Creating Relationships with Computers and Robots, Kluwer Academic Publishers, pp. 157–164, 2002]. The produced mechanical prototype has been designed to accommodate a prosthesis mandible with dentures. The mechanism could fit within the skull of the average man; where it would occupy less than 1/3 of the skull cavity. Two TMJs (temporomandibular joints) support the prosthesis, where each is driven by a 3 DOF parallel manipulator. In order to combine the motion of both manipulators each TMJ is capable of 3 DOF. The system is controlled via a USB port using software that models the human skull including collision detection mechanisms. The jaw allows for linear control, zero-backlash, and up to three times exaggerated mobility ranges making it also suitable for speech research, facial gesture affect research and dentistry applications.

4aSC9. Effects of subglottal acoustics on phonation onset. Juergen Neubauer, Zhaoyan Zhang, and David Berry (UCLA School of Medicine, 31-24 Rehabilitation Ctr., 1000 Veteran Ave., Los Angeles, CA 90095, zyzhang@ucla.edu)

The effect of subglottal acoustic loading on the vocal fold vibration was investigated using a self-oscillating mechanical model of the folds. Although the influence of the supraglottal tract on vocal fold vibration has received more attention than the subglottal system, the influence of the subglottal system on vocal fold vibration is also potentially significant, and merits investigation. In this study, the subglottal system consisted of a uniform tube connected to an expansion chamber on the flow supply end (e.g., a pseudo-lung). The length of the subglottal tube was varied systematically over a relatively large range in order to investigate the influence of subglottal acoustics on vocal fold vibration. Phonation onset and offset pressures were measured in the subglottal tube as a function of tube length. Over the range of investigation, the fundamental frequency of phonation was found to be negatively correlated with the subglottal tube length. However, both phonation onset and offset pressure were positively correlated with subglottal tube length, with the onset pressure increasing faster than the offset pressure. This hysteresis effect vanished and the two pressures merged at a small value of the subglottal tube length, indicating a change in the onset behavior from a subcritical Hopf bifurcation to a supercritical Hopf bifurcation (a codimension-2 bifurcation point). In addition, phonation did not exist below a critical subglottal tube length.

4aSC10. Classification of initial segments of sounds using onset data structures. Dagmar S. Fraser and Leslie S. Smith (Dept. of Computing Sci. and Mathematics, Univ. of Stirling, Stirling FK9 4LA, Scotland, dsf@cs.stir.ac.uk)

Biologically motivated techniques are used to record onset data structures [L. S. Smith and D. S. Fraser, IEEE TNNS **15**, 1125–1134 (2004)]. These structures record sets of events detailing both the spectral and intensity structure of sound onsets. They are inspired by the output spikes from onset cells in the cochlear nucleus. We suggest that the spectrotemporal characteristic of these structures provides useful information for classifying initial segments of sounds. The original onset data structure is of high dimensionality, and is difficult to interpret directly. A self-organizing feature map (SOFM or Kohonen network) is therefore used to provide a lower dimensional coding. The SOFM requires fixed-length data vectors, preferably not of too high a dimensionality. A number of different reformatting techniques are used to provide this. For the SOFM, identified activated map areas are associated with particular sounds. Using the TIMIT database these areas can be labelled, providing a testable classification scheme. This is compared with other work classsifying TIMIT phonemes. We believe that the SOFM can be extended to classify the onsets of other (non-speech) sounds. [Work supported by UK EPSRC.]

4aSC11. Computational fluid dynamics simulation of a sibilant. Kazunori Nozaki, Toyokazu Akiyama, Shinji Shimojo (Cybermedia Ctr., Osaka Univ., 5-1 Mihogaoka, Ibaraki, Osaka 567-0047, Japan, nozaki@cmc.osaka-u.ac.jp), Shingo Maeda, Takeshi Kaishima (Osaka Univ.), Hiroo Tamagawa, and Yoshinobu Maeda (Osaka Univ.)

It is well known that a sibilant is generated around the anterior teeth and the frontal part of palate. However, there is no study that the sibilant sound source is detected and visualized. Dentists often need to fabricate the prostheses including the anterior teeth and the palate. It would be useful if the critical part of the prostheses on the generation of sibilant could be detected before the fabrication and treatment. In this present study, Computational Fluid Dynamics simulation was performed in order to examine the oral air flow on pronouncing sibilant. Moreover, Powell sound source around the anterior teeth was detected and visualized by combining Computational Fluid Dynamics simulation and Computational Acoustical Analysis. As a result of this study, dentists can obtain the clinical criteria of the prosthetic treatment with considering sibilant.

4aSC12. Deriving speaking rate effects on tonal realization without varying the speech rate. Alan C. L. Yu (Univ. of Chicago, 1010 E 59th St., Chicago, IL 60637, aclyu@uchicago.edu)

Previous research investigates the effect of durational variation on tonal realization by artificially inducing a change in speaking rate. Through an investigation of the $f0$ pattern of Cantonese mid-rising tone, this paper introduces a novel, indirect, method of investigating durational effect on tonal realization by looking at the variation in tonal realization in different syllable types. Cantonese has three syllable types: CV, CV(V)O, CV(V)N. CV(V)N syllables are found to be significantly longer than CV syllables, which in turn are significantly longer than CV(V)O syllables. In this study, five native Cantonese speakers were asked to recite a list of Cantonese words with a mid-rising tone in a carrier phrase. The $f0$ of the Turning Point (the lowest point of a rise), the $f0$ peak of the rise, the durations of the Turning Point and the $f0$ peak relative to the onset of voicing, and the duration of the syllable rime were measured. The results show that the $f0$ peak frequency remains constant regardless of the duration of the syllable. The $f0$ of the Turning Point gets higher when the duration of the syllable becomes shorter. Finally, the slope of the $f0$ rise remains constant regardless of the duration of the syllable.

4aSC13. Mechanical vocal cord model mimicking human biological structure. Eiji Shintaku, Kotaro Fukui, Kazufumi Nishikawa, Shunsuke Ikeo (Dept. of Mech. Eng., Waseda Univ., 3-4-1 Ookubo, Shinjuku-ku, Tokyo 169-8555, Japan), Kentaro Takada, Atsuo Takanishi (Waseda Univ., Saitama, Japan), Hideaki Takanobu (Kogakuin Univ., Tokyo, Japan), and Masaaki Honda (Waseda Univ., Saitama, Japan)

We present a mechanical vocal cord model aiming for a talking robot, WT-5 (Waseda Talker No. 5). Unlike a musical reed which has been used in conventional mechanical speech synthesizer, the vocal cord model is formed to mimic the human's vocal cord in the shape and the biological structure. It is made of a thermoplastic rubber, Septonh (Kuraray Co. Ltd.) of which the elasticity like a human's, and has 3-DOF mechanisms which is similar to the human structure. 1-DOF link mechanism could change the pitch by stretching the length of the vocal cords. The 2-DOF arm mechanism is used to mimic the abduction and adduction of a human arytenoid cartilage. The vocal cord model was excited by air flow exhausted from a mechanical lung model. The vibration pattern was observed by a high-speed camera, and the glottal volume velocity and the sound pressure were recorded by a mask-type wire screen pneumotachograph and a microphone. It was shown that the lower and upper edges of the vocal cords could vibrate in a different phase and the sound spectrum was similar to the source spectrum of human speech.

4aSC14. An educational articulatory synthesizer, EASY. Richard S. McGowan (CReSS LLC, 1 Seaborn Pl., Lexington, MA 02420) and Reiner Wilhelms-Tricarico (CReSS LLC and Haskins Labs., New Haven, CT 06511)

An articulatory synthesizer has been written for general educational and research use. It has been named Educational Articulatory Synthesizer, EASY, and written in the MATLAB programming language. In its current instantiation, the synthesizer performs calculations in the frequency domain and includes the effects of wall-vibration, viscous and thermal boundary layer loss, and radiation loss. Sources anywhere from the glottis to the mouth can be specified. Further, the code allows for side branches other than the nasal tract so that sounds such as laterals can be specified. A low-frequency aerodynamic module has also been included. Midsagittal shape is controlled by a hierarchy of flesh points. Points high in the hierarchy specify the overall shape by determining reference positions for points lower in the hierarchy, which, in turn, determine the shape locally. [Work supported by grant NIDCD-001247 to CReSS LLC.]

4a THU. AM

4aSC15. The relation between learning Mandarin Pinyin or Zhuyin and $L2$ (English) production. Yan Helen Yu and Fredericka Bell-Berti (Dept. of Speech, Commun. Sci., & Theatre, St. John's Univ., 8000 Utopia Pkwy, Jamaica, NY 11439)

This study investigates the relation between early written first language learning experience and the perception and production of speech in a person's second langauge ($L2$). It compares the English pronunciation of native Mandarin speakers who have had different sound-annotating learning experiences (Pinyin versus Zhuyin). We predict that native Mandarin speakers who used the Pinyin system will produce some English sounds as their Pinyin counterparts if the same Roman letter is used as a pronunciation symbol both in English and in Pinyin system. However, because Pinyin and Zhuyin are not the primary written form for Mandarin, and are extensively used only in the first years of school, and because $L2$ learning involves exposure to speech as well as reading, we expect only moderate differences between the two groups' perception and production abilities. We will examine the interaction for Pinyin and Zhuyin learners between (Mandarin) $L1$ learning experience, age effects, $L2$ experiences, and the establishment of English phonetic categories for the English monophthongal vowels and consonants /r/, /z/, and /l/.

signals were extracted from the waveform and analyzed and compared using customized Matlab routines. Submitted For (Speech Communication) Young Presenter Award and (Speech Communication) Best Student Paper Award.

4aSC16. Characterization of voice pathologies from acoustic signals using wavelet analysis. Sarah A. Bentil and Yuling Yan (Dept. of Mech. Eng., 302 Holmes Hall 302, Univ. of Hawaii-Manoa, Honolulu, HI 96822, bentil@hawaii.edu)

We present a method to quantitatively characterize voice abnormalities using wavelet analysis. The proposed method uses wavelets to decompose acoustic signals, acquired clinically from patients with normal and pathological voices, into their designated frequency/time components. These components contain valuable information on the unique dynamic properties of the vocal system and correlate with specific voice conditions. A comparative analysis of these vocal signals using spectrogram is also presented. These combined analyses provide comprehensive representation and quantitative characteristics of the vocal dynamics, which may provide key indication of vocal abnormalities. Further wavelet analysis of acoustic data evaluates variations in the characteristics of the vocal signal from one glottal cycle to the next. This is valuable since vocal signals representing most pathological voice productions exhibit inter-cycle variations in intensity or/and frequency. Our results of analysis show that the wavelet analysis can be used to characterize voice pathologies and provide information regarding the type and severity of the disorder. [Work supported by NSF awarded to Yan.]

4aSC17. Synthesizer software for modeling voice quality. Norma Antonanzas-Barroso, Bruce R. Gerratt, and Jody Kreiman (Div. of Head/Neck Surgery, UCLA School of Medicine, 31-24 Rehab Ctr., Los Angeles, CA 90095-1794, jkreiman@ucla.edu)

This poster presents a formant synthesizer that is designed especially for detailed modeling of voice quality. Users may interactively manipulate source pulse shapes and/or spectral characteristics of the voice source, the noise spectrum and noise-to-signal ratio, jitter and shimmer, vocal tremor rate and extent, and the vocal tract transfer function (formants and bandwidths), all in near-real time. Pitch contours can be modeled directly or imported. The synthesizer may be utilized in method-of-adjustment tasks, or sequences of stimuli may be created for use in other experimental paradigms. An interactive audio-visual display makes this a useful tool for teaching voice acoustics as well. Both an executable version of the program and the underlying code will be available during the conference. Copies of software that supports the voice modeling effort, including an interactive inverse filter, will also be distributed. [Work supported by NIDCD grant DC01797.]

4aSC18. Simulation and analysis of tremor in speech production. Kimberly A. Farinella and Brad H. Story (Dept. of Speech, Lang., and Hearing Sci., Univ. of Arizona, P.O. Box 210071, Tucson, AZ 85721-0071)

Tremor of the muscles used for voice and speech production creates frequency and amplitude modulations in the acoustic speech signal. Frequency modulation results primarily from changes in muscle activations that alter the mass and stiffness of the vocal folds, consequently altering the fundamental frequency ($F0$). The primary source of amplitude modulation comes from time-varying activation of the respiratory muscles (which alter alveolar pressure), from laryngeal muscles that modify the maximum glottal area, or from interaction of voice source harmonics with the vocal tract filter. The purpose of this preliminary investigation was to analyze acoustic signals generated with an articulatory speech synthesizer in which tremor was imposed separately at each of the three anatomical sites: the respiratory system, larynx, and upper airway (pharynx). The synthesizer consisted of a voice source model of the time-varying glottal area coupled to a wave propagation model of the airways upstream and downstream of the vocal folds. This allowed for complete control of $F0$, alveolar pressure, glottal area, vocal tract shape, and modulation frequencies of each. Frequency and amplitude contours of the resulting acoustic

4aSC19. Temporal features in TV news and weather forecasts. Tatiana I. Shevchenko and Natalia Uglova (Moscow State Linguistic Univ., 38 Ostozhenka, Moscow 119992, Russia)

The paper is aimed at investigating the specific features of temporal component as manifested in authentic TV speech, news and weather forecasts of the three channels: NBC, Texas News, Philadelphia Fox. In our 10 min corpus (9 speakers, 5 men and 4 women) durational patterns and pauses between them as well as their ratio were measured. The data showed that on average a durational pattern lasts 2655 ms, while an average pause duration is 193 ms. Thus pausing does not take a major amount of time, while phonation is 14 times as long. Furthermore, the analysis reveals the abundance of short and extra short pauses which also stands for a rapid tempo. The temporal characteristics were correlated with regional affiliation, genre and gender distinctions. The speakers demonstrated uniformity in the basic temporal features determined by time limit constraints. The results of our investigation basically agree with previous research based on mass media interviews, reading and spontaneous monologues. However, our findings show that a salient feature of newsreaders speech is their ability to deliver an enormous volume of information at a very quick tempo, which is a real challenge to viewers.

4aSC20. Stabilization techniques for ultrasound imaging of speech articulations. Lisa Davidson and Paul De Decker (Dept. of Linguist., New York Univ., 719 Broadway 4th Fl, New York, NY 10003)

One challenge for the ultrasound imaging of the tongue during connected speech is stabilization of both speakers heads and the ultrasound transducer. To accurately analyze tongue shapes, researchers must ensure that differences among images result from changes in tongue shape, not head or transducer movement. In this validation study, we present an inexpensive, space-saving technique for stabilizing the head and transducer during the collection of ultrasound images in the laboratory. Four speakers were video-recorded producing 5 blocks of 19 sentences each. The speakers' heads were immobilized with a moldable head stabilizer (Comfort Company, model #HSM) affixed to a wall while the transducer was held by a microphone stand. Markers were attached to points on the speakers' faces and the transducer. Custom-written MATLAB software was used to find the center of the markers for every 10th frame of the recordings. Results indicate that the transducer does not move, and head movement beyond the measurement error of 1 mm is confined to the first block, while speakers are adjusting to the equipment. As compared to other stabilization techniques [Stone and Davis, J. Acoust. Soc. Am. **98**, 3107–3112 (1995)], this method provides equivalent immobilization that is significantly less expensive and portable for fieldwork.

4aSC21. Acoustic eigenmodes and formant-cavity affiliations for the time-varying vocal tract. Gordon Ramsay (Haskins Labs., 270 Crown St., New Haven, CT 06511-6204, ramsay@haskins.yale.edu), Philip Rubin (Haskins Labs., New Haven, CT 06511-6204), and Catherine Best (Haskins Labs., New Haven, CT 06511-6204)

Acoustic analyses of speech production are often based on assumptions about formant-cavity affiliations. Typically, these are derived from approximations based on coupled resonators, or from small-perturbation analyses for open vowel configurations. Few studies have accurately calculated formant-cavity affiliations for tightly-constricted vocal tract shapes, or for vocal tract shapes that change in time. In this paper, we show how to determine the acoustic eigenmodes of the vocal tract from a physical simulation, and apply this to examine time-varying formant-cavity affiliations during sequences of consonants and vowels. A state-

space model of quasi-one-dimensional acoustic wave propagation in a time-varying elastic tube is derived, using the finite volume method. The eigenvalues and eigenvectors of the resulting implicit matrix recursion are shown to determine the formant frequencies and bandwidths, and the spatial distribution of potential and kinetic energy for each formant, at each point in time. Illustrations are given for examples of VCV sequences constructed using an articulatory model, including stops, fricatives, and approximants in different vocalic contexts. Formant-cavity affiliations, as defined by the localization of acoustic energy along the vocal tract, do not necessarily correspond to actual cavities. [Work supported by NIH.]

THURSDAY MORNING, 19 MAY 2005 GEORGIA A, 8:30 TO 11:45 A.M.

Session 4aSP

Signal Processing in Acoustics: Design and Applications of Arrays

Paul J. Gendron, Chair
Naval Research Lab., Acoustics Div., 4555 Overlook Ave., SW, Washington, DC 20375-0002

Contributed Papers

8:30

4aSP1. Broadband nearfield beamformer with compact size by using Nash genetic algorithm. Soonkwon Paik (Elect. & Comput. Eng., Univ. of Texas, 1 Univ. Station Stop C0803, Austin, TX 78712, soonkwon@mail.utexas.edu) and Elmer L. Hixson (Univ. of Texas, Austin, TX 78712)

Though the acoustical array is effective in the reverberant field, its usage has been limited at low frequencies since the array with the length of more than 1 meter is difficult to install inside a vehicle. Furthermore, it is impractical to use circular coordinates especially inside vehicle since the sound source, a human mouth, stays at a fixed distance from and moves along the parallel line to the surface of the headliner. For the delay-and-sum beamformer in near-field, it is difficult to get the complex coefficients, amplitude weight and time delay, by analytical methods. In this paper, one numerical optimization method for a fixed microphone array is investigated which uses the Nash Genetic Algorithm (Nash GA) which was originally used in electromagnetism problems. By using Nash GA, first the optimal geometry of array element alignment inside a vehicle is found, then the complex coefficients are obtained. The results measured in an anechoic chamber correspond to that of computer simulation and satisfy the requirement of hands-free mobile telephony where the microphone array is installed in a vehicle's headliner and the data is also measured in a reverberant field to investigate the signal to noise ratio.

8:45

4aSP2. Capacity of the oceanic waveguide. W. J. Higley, Philippe Roux, and W. A. Kuperman (Marine Physical Lab., Scripps Inst. of Oceanogr., 9500 Gilman Dr., La Jolla, CA 92093-0238)

Capacity between two multiple-input/multiple-output (MIMO) arrays can be computed by decomposing the Green's function matrix into orthogonal functions, which can be thought of as independent subchannels. In a waveguide with sufficient array sampling, the modes are the independent subchannels, and capacity can easily be calculated for given waveguide parameters. Capacity is calculated using the optimal power allocation strategy of "water-filling." Water-filling is simultaneously done across modes and frequency. Theoretically computed capacities for an oceanic waveguide between two vertical arrays for various waveguide parameters, along with physical explanation, will be presented.

9:00

4aSP3. Detection enhancement using multiple time-reversed guide sources in shallow water: Analysis of TREX04 data. David C. Calvo, Charles F. Gaumond, and David M. Fromm (Acoust. Div. Naval Res. Lab., Washington, DC 20375-5320)

Detection in a monostatic, broadband, active sonar system in shallow water is degraded by propagation-induced spreading. In the TREX04 experiment, performed south of the Hudson Canyon off the coast of New Jersey, 0.25 sec LFMs were transmitted with 500 Hz bandwidths chosen over a 0.5–3.5 kHz range using the NRL 64 element source-receiver array. The transmissions were then echo-repeated by a distant ship at a range varying between 1–5 km. The echo-repeating ship also transmitted one-way, 1 s, LFMs which were used as guide-source signals. These guide-source signals, which contain environmental information, form the basis of a technique for improving detection without having explicit environmental knowledge. Using an empirical-orthogonal-function representation of the set of monostatic guide-source signals, echoes were convolved with the time-reversed orthogonal functions as part of a filter bank. The result is improved probability of detection of noisy echoes using multiple guide-source signals in comparison with a baseline probability of detection using matched-filtering. In this talk, ROC curve improvement is analyzed as a function of the many parameters, e.g., target depth, source aperture, bandwidth, number of guide sources and their spatial and temporal separation. Comparison with numerical simulations is also made. [Work supported by ONR.]

9:15

4aSP4. Backpropagation image analysis of broadband decomposition of the time reversal operator (DORT) data from TREX-04. David M. Fromm, Charles F. Gaumond, Richard Menis, David C. Calvo (Naval Res. Lab., 4555 Overlook Ave. SW, Washington, DC 20375-5320), Joseph F. Lingevitch, Geoff F. Edelmann, and Elisabeth Kim (Naval Res. Lab., Washington, DC 20375-5320)

Single-frequency DORT is a method of isolating scatterers with a multiple-source/multiple-receiver system [C. Prada *et al.*, J. Acoust. Soc. Am. **99**, 2067–2076 (1996)]. Broadband DORT overcomes problems of frequency fading and dispersion by exploiting the response of singular values across the entire signal bandwidth. In spring 2004, the Time Reversal Experiment, TREX-04, collected data for the evaluation of the broadband DORT technique. An 80 m vertical source/receiver array with 64 hydrophones was deployed in 100 m deep water in an area south of Hudson Canyon off the New Jersey coast. Sets of four, five, or six beams were transmitted at small angles from horizontal and the response from an echo repeater was recorded on 64 channels. Data was collected in 500 Hz bands between 500 and 3500 Hz with the echo repeater at ranges from 0.5 to 5.0 km and at mid-water column and near-bottom depths. Applying the

broadband DORT algorithm, the isolation of the echo repeater signal in the singular vectors and the robustness of broadband backpropagation images will be presented as a function of frequency band, signal coherence, echo repeater range and depth, noise levels, and receiver aperture. [Work supported by the ONR.]

9:30

4aSP5. Ultrasonic tomography for density gradient determination and defect analysis. Xiang Zhao and Joseph L. Rose (Dept. of Eng. Sci. and Mech., Penn State Univ., University Park, PA 16802)

Ultrasonic tomography is used to study density gradients and structural defects. In practice, it is not possible to measure an almost infinite number of projections, necessary for the transform-based techniques to produce highly accurate results. To overcome this difficulty, interpolations with respect to sample angle and projection angle based on fewer measurements are used to generate the required projection data. The number of sampled grid values necessary for displaying a well-balanced reconstructed image to reduce the aliasing distortions caused by insufficiency of the input data is then possible. For slowness reconstruction, as an example, one should constrain the time of flight at a source point to zero when using interpolation, other than using extrapolation to generate projection data to guarantee that the interpolated data around the source converges to zero. Hence no time is taken when a source and a detector overlap. Algebraic reconstruction algorithms, however, lack the accuracy when using a smaller number of measurements, and computational efficiency when using an abundant data set generated by interpolating. Results show that this approach can improve the overall accuracy and cost more effectively, thus making it a potentially, powerful tool for studying in-situ process and health monitoring. [Work supported by NSF.]

9:45

4aSP6. Implementation of a nearfield acoustic holography system based on pressure and velocity measurements. Michael C. Harris, Jonathan D. Blotter (Dept. of Mech. Eng., Brigham Young Univ., 435 CTB, Provo, UT 84602), and Scott D. Sommerfeldt (Brigham Young Univ., Provo, UT 84602)

Analytical models have indicated that a pressure and velocity-based nearfield acoustic holography (NAH) method will provide the same reconstruction resolution of current NAH methods with significantly fewer measurement locations. This would lead to a considerable savings in data acquisition time for scanning array systems and reduce the inefficiency of NAH for high spatial frequencies. The pressure and velocity-based NAH method will be introduced. Experimental results for a simply supported plate and cylinder will be presented. These results will be compared to results obtained using conventional NAH methods.

10:00–10:15 Break

10:15

4aSP7. Energetically optimal regularization in nearfield acoustical holography. Xiang Zhao (Dept. of Eng. Sci. and Mech., Penn State Univ., University Park, PA 16802)

Regularization techniques, such as singular value discarding or Tikhonov regularization, are commonly used to improve estimate of source field to be reconstructed from measured acoustic pressures at many points in nearfield acoustical holography. Theoretically, however, the regularized solution always underestimates the sound power of the real source. This paper presents an energetically optimal regularization method to solve this problem in nearfield acoustical holography by introducing a compensation factor in the generalized cross-validation function used to determine the optimal regularization parameter based on the Tikhonov regularization technique. Numerical examples show that the resulting regularization parameter will not only ensure a good fit of the predicted acoustic pressures to all measured data, but also ensure that the sound power of the equivalent source is always in agreement with the measured sound power, thus making this regularization method more ideal and much better than others in finding a compromise between the fidelity to input data and the fidelity to source field, and yielding a robust solution to the source field. [Work supported by NSF.]

10:30

4aSP8. Multi-variate error analysis of beam-forming acoustic measurements in a wind-tunnel. Ramani Ramakrishnan, Greg Kawall, Pushpinder Bhullar (Ryerson Univ., 350 Victoria St., Toronto, ON, Canada M5B 2K3, rramakri@ryerson.ca), and Norman Ball (Inst. for Areospace Res., NRCC, Ottawa, Canada K1A 0R6)

Source localization has been in the forefront in acoustics, since quantification of the source power and its exact location is de rigueur for successful noise control. Starting from single microphone measurements, many different methods, progressively more successful and complex, have been attempted for source localization. Recently, beam-forming methods, due to their success in underwater acoustics and architectural acoustics, have been used in wind tunnel tests. These tests have been quite extensively utilized both in automotive and aircraft tests. Beamforming techniques use an array of microphones, arranged in different patterns, to detect the source location and source power. In beamforming, the array in effect beams, not in the physical sense, towards the source to determine its position. Various parameters determine the success of the beamforming techniques. Some of these parameters are: number of microphones, microphone spacing, array pattern, source frequency and signal analysis procedures. A series of microphone array tests were conducted at the national Research Council of Canada's wind tunnel, with and without flow. The source location was known *a priori*. A multi-variate error analysis was performed to determine the importance of the different parameters. The results of the analysis will be presented in this paper.

10:45

4aSP9. Acoustical spectral analysis of a wake vortex cross-section using microphone-arrays. Hadi S. Wassaf (John A. Volpe Natl. Transportation Systems Ctr., Cambridge, MA 02142), Oliver C. Ibe (Univ. of Massachusetts, Lowell, MA 01854), and Robert P. Dougherty (OptiNav Inc., Bellevue, WA 98004)

The ability to unambiguously identify the acoustic energy generated by airplane wake vortices using phased microphone-array intensity distribution maps is hampered by the lack of knowledge of the associated nonstationary spectrum throughout the wake's duration. This ambiguity is especially pronounced when the low broadband signal to noise ratio renders the wake signal difficult to isolate from contaminating background noise. In this paper, a technique to isolate the acoustic spectral content of a cross sectional wake track (CSWT) is presented. First, image processing is used to generate a CSWT from vertical intensity distribution maps. In a second beamforming stage, the array's main-lobe traces the track to amplify the acoustic signal along the CSWT and estimate its time-varying power spectral density $P_{CSWT}(f)$ using Short Time Fourier Transform. Simultaneously, the background noise frequency-dependent upper confidence limit $CL(f)$ is estimated at each time increment by steering the array away from the wake source. The time varying $CL(f)$ is then used as an adaptive threshold function on $P_{CSWT}(f)$ in order to dynamically segment out the wake frequency bands from those of the contaminating noise sources.

11:00

4aSP10. Performance analysis of ambiguity function based broadband chirp direction of arrival estimation. Ning Ma and Joo Thiam Goh (DSO Natl. Labs., Singapore)

When estimating the direction of arrival (DOA) estimation of broadband sources with an array, the source is usually modeled as white noise. However, when the source signal has certain time-frequency structure, studies have shown that the DOA estimation performance can be improved

by using the information about its structure. In our previous work [Ning Ma and Joo Thiam Goh, ''DOA Estimation for Broadband Chirp Signals,'' Proc. of ICASSP 2004, Vol. II, pp. 261–264], we have proposed two new methods for broadband chirp signal DOA estimation based on the ambiguity functions, namely Incoherent Broadband Chirp DOA estimation (CBD-I) and Coherent Broadband Chirp DOA estimation (BCD-C). Besides improving the accuracy of the DOA estimates, these algorithms work with signal frequencies that are higher than the array design frequency. The number of sources are also not constrained by the number of sensors. The performance of the proposed DOA estimation methods is analyzed theoretically and numerically in this work. The analysis results show that these methods can improve the output SNR and therefore the DOA estimation performance when input SNR is higher than certain level.

11:15

4aSP11. Experimental reproduction of random pressure fields in laboratory conditions. Teresa M. Bravo and Cedric Maury (Dept. de Genie Mecanique Acoustique, UTC, Ctr. de Recherche Royallieu, BP20529, 60205 Compiegne France, teresa.bravo-maria@utc.fr)

The design and the physical implementation of an experimental set-up are discussed for the off-line reproduction of random wall-pressure fluctuations with given spatial correlation characteristics. This approach could provide a cost-effective laboratory method of both reducing the variability of low frequency sound transmission measurements as well as measuring the boundary layer noise transmitted through aircraft fuselage structures. Three different types of random excitations are considered, namely an acoustic diffuse field, a turbulent boundary layer excitation and an acoustic progressive wave at grazing incidence angle. The corresponding excitations are generated in a semi-anechoic chamber using a near-field array of 4×4 loudspeakers optimally driven and located above a set of 13×16 microphones. The microphones are positioned a short distance apart from an aluminum test panel. The optimal driving signals are determined from acoustic transfer functions measurements between the grid of microphones and the array of loudspeakers. Given this number of loudspeakers, a reasonable reproduction of the boundary-layer excitation is achieved up to about 200 Hz and a good approximation of both the acoustic diffuse field and the progressive wave is obtained over the frequency range of interest, i.e. up to about 700 Hz. [Work supported by ANVAR.]

11:30

4aSP12. Acoustic to seismic ground excitation using time reversal. Brad Libbey and Douglas J. Fenneman (Night Vision and Electron. Sensors Directorate, 10221 Burbeck Rd., Fort Belvoir, VA 22060)

Time reversal is a promising method of controlling the arrival of acoustic signals at receiver locations underwater and in biomedical tissues. This work applies the techniques in soil for landmine detection. The measurement of seismic vibrations between 50 and 300 Hz is problematic due to difficulties in transmitting energy into and through the soil from loudspeakers. The experimental setup consists of an array of loudspeakers capable of playing independent signals and a soil vibration sensor. The system response between the loudspeakers and the sensor are approximated, time reversed, and rebroadcast such that arrivals at the sensor are coincident. If the airborne and seismic paths are sufficiently complex, time reversal may help to control the energy distribution of a signal reaching the target region. However, the focusing ability is limited by an autocorrelation of the system impulse response that is inherent with time reversal processing. Simulations and experiments will be presented that demonstrate the advantages and limitations of using time reversal for controlling energy at a receiver. Results will compare vibration amplitude with and without the time reversal processing for target soil locations.

THURSDAY MORNING, 19 MAY 2005 GEORGIA B, 8:30 A.M. TO 12:00 NOON

Session 4aUW

Underwater Acoustics: Propagation: Modeling, and Experimental Results I

Kathleen E. Wage, Chair
Dept. of ECE, George Mason Univ., 4400 University Dr., Fairfax, VA 22030

Contributed Papers

8:30

4aUW1. Spectral features of sound field fluctuations in shallow water with internal solitons. Mohsen Badiey (Univ. of Delaware, College of Marine Studies, Newark, DE 19716), Valery Grigorev (Voronezh State Univ., Voronezh 394006, Russia), Boris Katsnelson (Voronezh State Univ., Voronezh 394006, Russia), and James Lynch (Woods Hole Oceanogr. Inst., Woods Hole, MA 02543)

Mode coupling of acoustic wave propagation in shallow water, in the presence of internal soliton is considered. It is shown that quasiperiodical behavior of temporal variability of the received signal (arising from the motion of soliton packets) provides some peaks in frequency spectrum. Positions of these peaks in spectrum are determined both by the space scales of interference beating (ray cycles) and velocity of solitons. Results of theoretical and numerical modeling as well as related experimental setup is discussed. [Work supported by ONR, RFBR and CRDF.]

8:45

4aUW2. Influence of random nonlinear internal wave parameters on resonant acoustic mode coupling. Scott D. Frank (Dept. of Mathematics, Marist College, Poughkeepsie, NY 12601) and William L. Siegmann (Rensselaer Polytechnic Inst., Troy, NY 12180)

Shallow water transmissions sometimes suffer anomalous amplification or loss caused by mode coupling due to the presence of nonlinear internal wave packets. The possibility of coupling between two acoustic modes is specified by an internal wave-acoustic resonance condition, which relates acoustic wavenumber differences to peak locations of the packet spectrum. This mechanism is critical in a recent analysis [S. D. Frank *et al.*, J. Acoust. Soc. Am. **116**, 3404–3422 (2004)] of some broadband intensity variations from the SWARM 95 experiment. That investigation used regularly spaced, evenly-sized nonlinear internal wave packets and it is important to determine how random variations in the wave widths and inter-wave spacings affect propagation results. An analytic expression

for the wavenumber spectrum of an idealized packet is obtained in terms of these parameters. This expression predicts spectral peak locations for parameters from the SWARM 95 model that correspond with observed data. Estimates of peak location statistics arising from random packet parameter variations are analyzed computationally and analytically. Variability of internal wave spectral peak locations is correlated with broadband intensity fluctuations obtained from parabolic equation computations. [Work partially supported by ONR.]

9:00

4aUW3. Influence of internal wave fluctuations on acoustic underwater propagation. Tatiana A. Andreeva, William W. Durgin, and Stefanie E. Wojcik (Mech. Eng. Dept., Worcester Polytechnic Inst., 100 Institute Rd., Worcester, MA 01609)

The work presents a ray acoustic based analysis of the effect of ocean internal waves and its fluctuations on acoustic wave propagation. In the present work, it is shown that consideration of only perfect internal waves simplifies the real problem, disregarding the fluctuations of internal waves. The focus of the paper is to study numerically the influence of internal wave fluctuations and the role of an initial ray angle onto underwater acoustic propagation. In the present formulation the eikonal equation is considered in the form of a second order, nonlinear ordinary differential equation with harmonic excitation due to internal wave. The harmonic excitation is taken imperfect, i.e., with a random phase modulation due to Gaussian white noise. The amplitude and wavelength of the acoustic waves are used as the principle signal characteristics in bifurcation analysis. The regions of instability are identified using the bifurcation and phase diagrams. The effect of internal waves phase modulation is demonstrated by means of comparison of the numerical data obtained in the present work with data, obtained perfect internal wave. The preliminary analysis shows very strong dependence of acoustic propagation on intensity of fluctuations.

9:15

4aUW4. Influence of internal waves on vertical coherence of sound propagation in the East China Sea. Jie Yang and Peter H. Rogers (School of Mech. Eng., Georgia Inst. of Technol., Atlanta, GA 30332-0405)

Literature of the last two decades has shown that internal waves are a major factor determining sound field fluctuations [Zhou *et al.*, J. Acoust. Soc. Am. **90**, 2042–2054 (1991); Lynch *et al.*, *ibid.* **99**, 803–821 (1996)]. Due to its time-dependent and anisotropic nature, internal waves can cause both temporal and spatial variations in the sound speed profile and hence, the acoustic field. In this paper, a model is built to study the range and azimuthal dependence of wave propagation in the presence of internal wave activity. The results are based on an *in situ* experiment ASIAEX in the East China Sea where intense tidal driven internal waves were recorded. The model explains the observed decorrelation after long range propagation (20 km) and in certain directions relative to the internal wave wavefront.

9:30

4aUW5. Intensity and time-angle fluctuations of high frequency signal propagation in shallow water. Mohsen Badiey, Jing Luo, and Aijun Song (Univ. of Delaware, College of Marine Studies, Newark, DE 19716)

Several high frequency experiments in recent years have been directed towards assessing signal variability as a function of channel environmental parameters. In this paper, results from one of these experiments in a shallow water region are shown. Acoustic communication coded signals were transmitted between a source and two receiver arrays separated 1 and 2 km respectively while detailed measurement of the channel environmental parameters were recorded. Channel impulse response function and the correlation time of the channel are obtained. Ray theory is used to analyze and interpret the experimental data. Arrival time-angle fluctuations were found to be directly correlated with the environmental variability due to ocean dynamics in the experiment region.

9:45

4aUW6. Mean and variance of the 3-D field forward propagated through random internal waves in continuously stratified deep and shallow water ocean waveguides. Tianrun Chen, Purnima Ratilal, and Nicholas C. Makris (MIT, 77 Massachusetts Ave., Cambridge, MA 02139)

A general modal solution for the mean and covariance of an acoustic field propagating through 3-D random inhomogeneities in an ocean waveguide [Ratilal and Makris, J. Acoust. Soc. Am. **114**, 2428 (2003)] is applied to propagation through a general 3-D internal wave field. Our previous two-layer internal wave parameterization is extended to continuous stratification and applied in both continental shelf and deep ocean waveguides. Physical and statistical properties of the internal waves are modeled with the Garret-Munk and other theoretical and empirical formulations. The mean and variance of the forward field propagating through internal waves is expressed in terms of the stochastic moments of the medium's scatter function density. This is formulated with the Rayleigh-Born approximation to Green's theorem in terms of the medium's compressibility and density fluctuations. In this way, the effect of both internal-wave sound speed and density variations on the forward propagated field are quantified. Density variations are shown to greatly affect high order modal propagation in some cases. Two-dimensional models of propagation through random internal wave fields are shown to become inaccurate after exceeding threshold ranges where 3-D scattering becomes important.

10:00

4aUW7. A mean field transport theory for long-range ocean acoustics. Michael Wolfson and Frank Henyey (Univ. of Washington, APL, 1013 NE 40th St, Seattle, WA 98105-6698)

Voronovich has found from an ocean model that the mean acoustic wavefield at low frequency persists after multi-megameter propagation in the deep ocean. The common belief has been that only the incoherent field survives long propagation distances. His results, if true in the real ocean, have important consequences to extracting signals from long-range propagation. The results of an alternative ocean model, for which the mean field is evaluated using transport theory, are presented, and compared to his results. Our model consists of a single background sound speed field superposed with sound speed fluctuations due to internal waves. We present results for sources used in recent experiments such as LOAPEX (75 Hz with about 30 Hz bandwidth).

10:15–10:30 Break

10:30

4aUW8. Calculation of amplitudes of acoustic normal modes from the reciprocity principle. Oleg A. Godin (CIRES, Univ. of Colorado and NOAA/Environ. Technol. Lab., Mail Code R/ET-0, 325 Broadway, Boulder, CO 80305)

Recently, J. D. Achenbach [J. Acoust. Soc. Am. **116**, 1481–1487 (2004)] put forward, on heuristic grounds, an elegant technique to calculate amplitudes of guided waves generated by mechanical loading in a range-independent elastic medium. The technique is based on application of the reciprocity principle and allows one to solve the problem without use of integral transforms. In this paper, the technique is applied to acoustic waveguides in a layered fluid. By taking into account continuous spectrum of the field, which was disregarded by Achenbach, a mathematical justification of the technique is obtained. The technique is shown to be exact. It is further extended to enable calculation of excitation coefficients of modes of both discrete and continuous spectra by a given sound source. The results are shown to be identical to those derived with the traditional, more cumbersome methods that rely on integral transforms. [Work supported by ONR.]

10:45

4aUW9. Moving ship tomography a first look at BASSEX 04. Kevin D. Heaney (OASIS Inc., Falls Church, VA 22044) and Arthur B. Baggeroer (MIT)

During the Basin Acoustic Seamount Scattering Experiment (BASSEX04) a line array was towed between two source moorings in the North Pacific separated by 500 km. The moorings were part of the SPICE04 test led by Scripps Institution of Oceanography. Each mooring had two broadband sources (center frequency 250 Hz) at depths of nominally 800 m and 3000 m. Thirty transmissions were recorded as the ship passed south along the line bisecting the two source moorings. The sources transmitted simultaneously (for each source). In order to break the conical ambiguity of the arrivals from each source, the array was turned 20 degrees for each reception. One set of sources used coded m-sequences with different laws, permitting separation of the signals by signal processing. Comparisons of the received signals will be compared with initial propagation model results using the measured sound speed field from the ship. An approach to $Nx2-\mathrm{D}$ tomography (where $N=30$) will be investigated. [Work supported by ONR Ocean Acoustics.]

11:00

4aUW10. Basin Acoustic Seamount Scattering Experiment (BASSEX 04) II. Downslope propagation. Kevin D. Heaney (OASIS Inc., Falls Church, VA 22044), Arthur B. Baggeroer (MIT), Kyle M. Becker (ARL-Penn State Univ.), Eddie Scheer, and Keith Vonderheydt (Woods Hole Oceanogr. Inst.)

During the Basin Acoustic Seamount Scattering EXperiment (BASSEX04), the Five-Octave Research Array (FORA) was deployed near the North Pacific Acoustics Laboratory (NPAL) bottom mounted tomography source just off the coast of Kauai. Receptions from the Kauai source were taken at ranges from 2 km out to 500 km. The data was taken to examine the affects of seafloor interaction on downslope propagation (propagation down both steep and shallow slopes), shallow water range-dependent 3-D propagation and conversion from shallow water propagation to deep water propagation. Conventional and adaptive beamforming results will be presented as well as comparisons between measured responses and PE modeled responses. Geo-acoustic inversions will be performed with the broadband source in end-fire and broadside orientations of the array. [Work supported by ONR Ocean Acoustics.]

11:15

4aUW11. Forward scattering from the Kermit-Roosvelt Seamount complex during the SPICEX-LOAPEX-BASSEX experiments. Arthur B. Baggeroer, Joseph SikoraIII (Massachusetts Inst. of Technol., Rm. 5-206, Cambridge, MA 02139), Kevin Heaney (Oasis Corp., Lexington, MA 02173), Edward K. Scheer, Keith von der Heydt (Woods Hole Oceanogr. Inst, Woods Hole, MA 02543), and Kyle Becker (Pennsylvania State Univ., State College, PA)

The SPICEX-LOAPEX-BASSEX expermiments were executed in the Northeast Pacific to examine several long range, low frequency propagation phenomena. Low frequency sources centered at 68 and 75 Hz with nominal bandwidth of 30 Hz and a source at 250 Hz with 100 Hz bandwidth were deployed transmitting orthogonal M seqeunces and FM chirps. A 64 element towed array cut for 250 Hz from Penn State Univ. was the primary multichannel receiving system. This presentation concerns the BASSEX study of low frequency scattering around the Kermit-Roosevelt Seamount complex which shoals to 900 m near 39 N and 145 W. We examine (i) the forward scattering shadow, (ii) any patterns of horizontal refraction as a function of source and receiver range from the seamounts, (iii) any backscattering from the seamounts, and (iv) the modal content of the signals by travel time methods. In addition, directional spectra of ambient noise were measured.

11:30

4aUW12. Three different directions of sound propagation in the China South Sea. Ma Li (Inst. of Acoust., Chinese Acad. of Sci., Beijing 100080, China)

In June 2004 three directions of sound propagation experiments were done in the China South Sea. One direction is equally sea depth. The second direction is slope. The third direction is largest gradient in sea depth direction. In the mean time three temperature chains have been applied for measuring the internal wave. During each temperature chain their distances are about 1000 m. The experiment results show that the packet of solitary waves has been found and each direction of sound propagation has different transmission loss. In the equally sea depth direction the sound propagation has the smallest transmission loss. And in the largest sea depth direction the sound propagation has the largest transmission loss. By three temperature chains data the packet of solitary waves propagation velocity and direction may be measured. And through the packet of solitary waves propagation direction the packet of solitary waves may be induced by the West Sand Island. In the end the couple normal mode program has been used to simulate the sound propagation which is affected by the packet of solitary waves. The simulating results are compared to experimental data.

11:45

4aUW13. Long-range acoustic transmission in a deterministic range-dependent ocean: Contribution from interference of near-axial waves. Natalie S. Grigorieva and Gregory M. Fridman (Dept. of Appl. Mathem. and Mathem. Modeling, St. Petersburg State Marine Tech. Univ., 3 Lotsmanskaya Str., St. Petersburg, 190008, Russia, nsgrig@natalie.spb.su)

The propagation of energy along the waveguide axis cannot be described in terms of geometrical acoustics because of the presence of cusped caustics repeatedly along the axis. In neighborhoods of these cusped caustics a very complicated interference pattern is observed. Neighborhoods of interference grow with range and at long ranges they overlap. For an arbitrary range-independent ocean, it was shown in [N. S. Grigorieva and G. M. Fridman, J. Comp. Acoust. **12**, 127–147 (2004)] that the interference of wave fields corresponding to near-axial rays results in a diffractive (as opposed to geometrical acoustics) component of the field which can be viewed as distinct, ''axial'' wave and effectively calculated using its integral representation. In this paper the integral representations of the axial wave in the frequency and time domains are obtained for a range-dependent ocean. In the frequency domain this representation has the form of a linear superposition of the solutions of the Helmholtz equation that are concentrated in a neighborhood of the sound-channel axis. The weight function is selected in such a way that the localization principle holds. The axial wave is simulated for a deterministic model of a range-dependent ocean corresponding to AET experiment. [Work supported by ONR Global.]

Session 4pAA

Architectural Acoustics and Noise: Soundscapes from an Architectural Viewpoint

Brigitte Schulte-Fortkamp, Chair
Technical Univ. Berlin, Inst. of Technical Acoustics, Secr TA 7, Einsteinufer 25, Berlin 10587, Germany

Chair's Introduction—2:15

Invited Papers

2:20

4pAA1. The soundscape as a tool for urban planners and town designers. Catherine Semidor (GRECO, EAPBx, Domaine de Raba F-33400 Talence, France, catherine.semidor@bordeaux.archi.fr)

Urban planners and other town designers need information to improve the environmental quality of cities, for instance the components of the urban soundscape. To use the latter as a decision-making tool in a planning process it is necessary first to link objective and subjective acoustic criteria with architectural parameters and then to transmit the observed characteristic relations in a way which designers will understand. One of these ways is to make use of spatial references (streets' shape, facades' material, etc.) to explain the impact of the urban morphology, for example, on the sound environment. That is why, for many years, surveys were conducted in a lot of different areas. This paper deals with some comparative results obtained in different European towns.

2:45

4pAA2. Phase characterization of soundscapes. Vincent Gibiat, Abril Padilla (PHASE, Univ. Paul Sabatier, 118 route de Narbonne, 31062 Toulouse, France, gibiat@cict.fr), Valentin Emiya (ENST, 75014, Paris, France), and Lionel Cros (ENSIETA, 29806 Brest Cedex 09, France)

Signal processing of soundscapes remains far from the auditive perception. It is particularly true when sounds recorded inside railway stations, are compared with the most accurate spectrographic representations. Such architectural realizations provide a soundscape between closed and opened spaces. As microphones used to discriminate sound sources as well as low level sounds use diffractions properties of radiation, as time discrimination between near sounds is efficient through time domain perception, as filtering often introduces phase distorsions, it has been decided to explore a new kind of signal representation based on the phase of the Short Time Fourier Transform. After a basic presentation of the numerical tool named SAFIR for Spectrography in Amplitude and Frequency, Instantaneous and Reallocated, examples will be given that first show the differences between closed and opened spaces, then the differences between the soundscape inside a train and inside a station. As this work has been realized with sounds recorded for a later musical realization that deals to reconstruct both a new realistic or imaginary coherent soundscape, it has been essential to extract the pertinent, coherent, details as magnification of time or space opposed with the incoherent urban noise. Finally SAFIR will be compared with the composer work.

3:10

4pAA3. How does hearing get into architectural design? Juergen Bauer (Architect, Mierendorffstr. 11, 10589 Berlin, Germany, jbauer.berlin@tiscali.de) and Brigitte Schulte-Fortkamp (Tech. Univ. Berlin, D-10587 Berlin, Germany)

Serious soundscape research needs an interdisciplinary approach to architecture. The design and the perception of architecture are dominated by the visual senses. Architects developing a complex building may be advised by a specified acoustic engineer. However the support provided by the acoustic engineer focuses less on creating acoustic qualities than on minimizing acoustic problems and noise effects. How about the sense of hearing of the architect? Designing architects have learned to shape 3-D. They have a precise vision of space and proportion. They know about light and shadow, about fabric and color. This profound world of vision is the basis of their inspiration, and they would be grateful to get more consciously in touch with the world of hearing not in terms of creating extraordinary sound-effects but in terms of imaginative listening. The visual phenomena architects are dealing with offer an approach to the world of hearing or rather analogies from watching towards listening. Light, shadow, fabric and color are no more matters only of sight but subject to audiovisual awareness. What acoustic processes might be most relevant to designing architects? What acoustic events and impacts should be specified in the education of design?

3:35

4pAA4. Soundscape perception of an urban park under flight paths of Naples airport. Giovanni Brambilla (CNR-Inst. of Acoust., via del Fosso del Cavaliere 100, 00133 Rome, Italy, giovanni.brambilla@idac.rm.cnr.it), Leda De Gregorio, and Luigi Maffei (Second Univ. of Naples, 81031 Aversa, Italy)

The Capodimonte historical urban park is the largest green area in Naples, located in the North and under the flight-paths of the city airport. Noise measurements, taken in the morning on two weekdays and one week-end in different sites inside the park, have shown LAeq levels, excluding aircraft noise, lower than 50 dB(A) and up to 55 dB(A) on Sunday morning near the museum. During the noise measurements one aircraft flyover occurred approximately every five minutes, producing an increase of LAeq in the range

5-15 dB(A) and LAmax levels up to 90 dB(A). Interviews have been carried out on a sample of subjects present in the park while the noise measurements were taken. By means of a questionnaire, they were asked to rate different factors describing the environment and influencing their experience to enjoy the park. The analysis of the subjective data collected is in progress and the preliminary outcomes show that the reported annoyance due to aircraft noise is much lower than expected by the noise levels, most likely due to synergetic effects of the other environmental factors.

4:00

4pAA5. Community design with soundscape in mind. Bennett M. Brooks (Brooks Acoust. Corp., 27 Hartford Turnpike, Vernon, CT 06066, bbrooks@brooksacoustics.com)

Community standards for sound are usually defined by limits on noise emitters. These limits may be absolute in level, or relative to a background level. Most often a limit is given as an overall A-weighted level. Occasionally, limits are placed on octave band spectrum levels or on tonal content. The time function of noise is generally addressed only broadly, defined by terms such as impulse, intermittent, or continuous. The character of sound in the community is addressed in policy or law by vague definitions which prohibit sounds which are obnoxious or that cause a nuisance. The positive aspects of sound in communities are usually left to chance. Recently, one summer resort community initiated a policy which promotes outdoor entertainment in one district, while requiring quiet in another. This unique policy balances lifestyle and economic interests, and was developed through an inclusive community-based public process. The question of how the community-based policy development process may be used to enhance the soundscape of a targeted area is explored. The implications of such a policy process for the designers of buildings and public spaces are discussed.

Contributed Papers

4:25

4pAA6. Aesthetic transformations of soundscapes as a basis for architectural design. Gary Siebein (Univ. of Florida School of Architecture, P.O. Box 115702, Gainesville, FL 32607)

A graduate design studio in acoustics examines sound and silence as manifest in urban and rural exterior soundscapes as well as interior soundscapes within performance venues as potential theoretical and formal bases for architectural form. The soundscape analyses were composed of a series of quantitative and qualitative studies of real and virtual environments such as the actual project site in downtown Orlando and the interior soundscape of the Stats Opera in Vienna in the 19th century. Transformations of the soundscape analysis served as generative ideas for the basis of architectural interventions in complex sites and programs for a multi-venue performing arts center.

4:40

4pAA7. Bow bells—creating a sound community. Brenda Kiser and David Lubman (David Lubman & Assoc., 14301 Middletown Ln., Westminster, CA 92683)

For centuries, the sound of ringing church bells has given communities a sense of identity. Church bells have long been used to announce significant events to the community, such as births and deaths, to warn of impending danger, to call the faithful to church and to announce the curfew. This paper focuses on one prominent example of a sound community-the church of St. Mary-le-Bow in the City of London, the location of the famous Bow bells. These bells have featured in Londons folklore and history since the 14th century and, to this day, identify those born within the sound of Bow bells as Cockney. This sound neighborhood was extended far beyond its natural borders when a recording of the bells was played by the BBCs World Service during World War II in broadcasts to occupied Europe. Sound examples are given. A short history of European church bells is presented. A cognitive scale is proposed, ranging from belonging to alienated, delineating psychological reactions of those within a sound neighborhood.

THURSDAY AFTERNOON, 19 MAY 2005 — PLAZA C, 1:25 TO 4:55 P.M.

Session 4pAB

Animal Bioacoustics: Tools for Animal Bioacoustics: New Designs and Directions II

Susanna Blackwell, Chair
120 Tamarack Dr., Aptos, CA 95003

Chair's Introduction—1:25

Contributed Papers

1:30

4pAB1. Autonomous remote humpback whale acoustic monitoring devices. Whitlow Au, Alison Stimpert, Marc Lammers (Marine Mammal Res. Program, Hawaii Inst. of Marine Biol., P.O. Box 1106, Kailua, HI 96734), and Niklas Johnson (Lund Univ., Lund, Sweden)

Four acoustic monitoring devices controlled by a CF-1 Persistor Instrument Inc. microcontroller board were developed to monitor the chorusing sounds of humpback whales in their wintering grounds off the island of Maui. The units were programmed to turn on at the same time and sample chorusing sounds at a sampling rate of 3.33 kHz for 3 minutes every hour. The CF-1 was chosen because of its low power requirement of 40 ma while collecting data during the 3-min collection period, 3 ma while sleeping and 96 ma while storing the data onto a 2 Gbyte compact flash card. At this sampling rate, sampling period duration, and interval between sampling period, each unit has sufficient power to last 30 days when powered from a 12 Ahr lead acid battery. The 2 GB memory allows for 34 days of data collection. Four of these units were deployed along the west

shore of Maui at locations separated by 15 to 20 miles. The amplitude of chorusing by singing humpback whales and the temporal characteristics at the different locations were obtained with these sensors during the 2005 humpback whale season.

1:45

4pAB2. Monitoring ecologically important fish sounds: The characteristics and diel trends of sounds produced by three species of Pomacentrids. T. Aran Mooney, Marc O. Lammers (Hawaii Inst. of Marine Biol., Univ. of Hawaii, Kaneohe, HI 96744), Pedro A. Santos (Univ. of the Azores), and Paul E. Nachtigall (Univ. of Hawaii)

Many fish species make sounds. These sounds can be recorded, characterized, identified to species, correlated with behavior, and then monitored for temporal and behavioral trends. We developed a two-part approach to accomplish these tasks. The first is a Portable Underwater Acoustic (PUA) device, composed of a two-hydrophone, handheld, digital recorder. This system is used to identify fish sounds that can then be correlated with specific behaviors. The second is a single-hydrophone, autonomous recording system that is placed on the seafloor to record the ambient sound field at set temporal intervals. This presentation describes both systems and how they are being applied in the field to study three species of fish: *Abudefduf luridus*, in the Azores, Portugal, and *Dascyllus albisella*, and *Abudefduf sordidus*, in Hawaii, USA. *A. luridus* sounds were usually produced in a doublet, 100 ms in duration, a peak frequency of 400–450 Hz, and associated with aggressive displays. Sounds produced by *D. albisella* were typically 2–4 s, with a peak frequency of 300–400 Hz and a bandwidth of up to 6 kHz. *A. sordidus* produced 100 ms broadband clicks, up to 5 kHz bandwidth, in aggressive chases. After identification, sounds were associated with behaviors and ecologically important events, and monitored for diel trends in their production.

2:00

4pAB3. Application of a diver-operated 4-channel acoustic/video recording device to study wild dolphin echolocation and communication. Michiel Schotten, Marc O. Lammers (Marine Mammal Res. Program, Hawaii Inst. of Marine Biol., 46-007 Lilipuna Rd., Kaneohe, HI 96744, micschotten@hotmail.com), Ken Sexton (The Sexton Co. LLC, Salem, OR 97301), and Whitlow W. L. Au (Hawaii Inst. of Marine Biol., Kaneohe, HI 96744)

Despite decades of research on captive dolphins, there still is much to learn about dolphin echolocation and communication in the wild. This is mainly due to the difficulties associated with recording dolphin signals accurately in the wild, such as their broadband (with frequencies over 200 kHz) and directional nature, the difficulty of identifying vocalizing dolphins, and being confined to equipment aboard a vessel. To resolve these difficulties, an underwater portable battery-operated digital recording device was developed that is capable of simultaneously capturing acoustic signals up to 220 kHz on 4 channels as well as video. It consists of a custom-made underwater housing, containing two batteries, a digital camcorder, and an acoustic recording unit with signal conditioning board, 4-channel data acquisition card, PC/104-plus single board computer and power supply, and a notebook hard disk. Attached to the housing is a 4-hydrophone star array. Post-recording, acoustic signals recorded within 15 m are accurately localized in 3-D and attributed to individual dolphins on the video. Thus, acoustic signal features are correlated with different echolocation behaviors and communicative signal exchanges among dolphins are assessed. The system has been successfully field tested with spinner dolphins (*Stenella longirostris*) along the leeward coast of Oahu, Hawaii.

2:15

4pAB4. A new acoustic bat detector. Matthew Heavner (Univ. of Alaska, Southeast, 11120 Glacier Hwy., Juneau, AK 99801, matt.heavner@uas.alaska.edu)

A new small, low-power, long-duration, field-deployable computer for the acoustic monitoring of bats in Southeast Alaska is being developed. The most recent study of bats in the region, reported by Parker *et al*., (1997), found the geographic range of four bat species to have northern distribution limits in Southeast Alaska. (A fifth species, *E. fuscus*, goes much farther north.) The relationship of the bats to forest management practices, the habitat usage of the bats, and population size and trends are all very poorly known for bats in Southeast Alaska. Long duration monitoring of several different types of area (such as old-growth versus recently logged forest) will provide knowledge to improve management practices in regards to bat ecology in Southeast Alaska. With the motivations just described, the hardware, development methods, and analysis software designed to develop an improved detector are presented. [Work supported by Alaska Department of Fish & Game.]

2:30

4pAB5. Automatic detection of microchiroptera echolocation calls from field recordings using machine learning algorithms. Mark D. Skowronski and John G. Harris (Computational Neuro-Eng. Lab, Univ. of Florida, Gainesville, FL 32611, markskow@cnel.ufl.edu)

The authors have recently presented experimental results of applying machine learning algorithms, used extensively in human automatic speech recognition research (ASR), to automatic species identification of echolocating bats [Skowronski and Harris, J. Acoust. Soc. Am. **116**, 2639 (2004)]. The results of those experiments demonstrated that *frame-based* classification, preferred in ASR, out-performs *holistic* classification typically employed in automatic echolocating species identification. The authors have extended the paradigm of machine learning algorithms to the related problem of bat call detection. A robust automatic bat call detection algorithm, to replace hand labeling, is required for two reasons: (1) for real-time species identification in the field, and (2) because hand labeling is subjective, tedious, slow, and error-prone. The current experiments compare various frame-based features (log energy, pitch estimates, pitch slopes) with several models of detection (matched filters, Gaussian mixtures, decision trees). Detector sensitivity and specificity are quantified for comparison using hand-labeled calls, with considerations of classification requirements for detected calls. That is, a detector is not penalized for including a short segment of background signal before and after a hand-labeled call. The results demonstrate the superior performance of the frame-based features and machine learning detection algorithms compared to conventional features and detection algorithms.

2:45

4pAB6. Detection of FM signals in the presence of non-Gaussian noise. Ildar Urazghildiiev, Kathryn Cortopassi, and Christopher Clark (Bioacoustics Res. Program, Cornell Lab. of Ornithology, 159 Sapsucker Woods Rd., Ithaca, NY 14850)

In bioacoustics, the problem of detecting frequency-modulated signals in the presence of non-Gaussian noise is of great interest. Matched filters (MF) are often chosen over frequency-specific energy (FSE) detectors because of their improvement in both signal gain and target specificity. Under non-Gaussian conditions however, MF detectors do not ensure acceptable trade-off between false alarm and missed detection rate. To decrease false alarm rate, we propose a two-stage detection technique. First, a MF is applied, with threshold prescribed by the acceptable false alarm rate, to generate candidate detections. Second, a signal recognition (SR) algorithm is applied to the candidates. The SR algorithm estimates modulation parameters from the signal spectrogram, and a detection decision is made based on how well parameters match a criterion set for the signal of interest. We applied this technique to right whale contact calls recorded in Cape Cod Bay. Results demonstrate that the MF-SR technique decreases false alarm rate by 3–5 times, while not increasing missed detection rate

from that of a MF only. These results were compared to those of a FSE detector followed by the same signal recognition and decision stage (FSE-SR), to determine if comparable performance could be achieved without the MF first stage.

3:00–3:10 Break

3:10

4pAB7. A computational model of echolocation: Restoration of an acoustic image from a single-emission echo. Ikuo Matsuo and Masafumi Yano (Res. Inst. of Elec. Commun., Tohoku Univ., Katahira 2-1-1, Aoba-ku, Sendai 980-8577, Japan, matsuo@riec.tohoku.ac.jp)

Bats can form a fine acoustic image of an object using frequency-modulated echolocation sound. The acoustic image is an impulse response, known as a reflected-intensity distribution, which is composed of amplitude and phase spectra over a range of frequencies. However, bats detect only the amplitude spectrum due to the low-time resolution of their peripheral auditory system, and the frequency range of emission is restricted. The amplitude spectrum varies with the changes in the configuration of the reflected-intensity distribution, while the phase spectrum varies with the changes in its configuration and location. Here, by introducing some reasonable constraints, we propose a method for restoring an acoustic image from the echo produced by a single emission. The configuration is extrapolated from the amplitude spectrum of the restricted frequency range by using the continuity condition of the amplitude spectrum at the minimum frequency of the emission and the minimum phase condition. The determination of the location requires extracting the temporal changes of amplitude spectra. For this purpose, the Gaussian chirplets with a carrier frequency compatible with bat emission sweep rates was used. The location is estimated from the temporal changes of the amplitude spectra. This method can determine the acoustic images of objects.

3:25

4pAB8. Broken line 3-dimensional sperm whale diving trajectory reconstruction using passive acoustics on a single hydrophone. Christophe Laplanche, Olivier Adam, Maciej Lopatka, and Jean-François Motsch (Université Paris XII, 61 av. du Gal de Gaulle, 94010 Créteil, France)

Sperm whales make deep dives to hunt. A dive, lasting 45 minutes on average, is composed of a vertical descent to the prey layer depth, the properly so called hunt (at a quasi constant depth) inside the prey layer, and a vertical ascent back to the sea surface. Sperm whales make series of echolocation signals (*clicks*) during the two first stages. The sea surface/bottom click echo detection and delay measurements then make possible the sperm whale range/depth estimation during these stages, by passively using a single hydrophone. The vertical, rectilinear sperm whale trajectory during the first stage is unambiguously estimated from the echo delays. The sperm whale trajectory can also be reconstructed during the second stage, from the sperm whale range variations only, even when not detecting sea bottom click echoes. These range variations strongly suggest the sperm whale trajectory to be a broken line (e.g., a 2-piece line, 600 m straight ahead, 85 degree bend, 1000 m straight ahead). Assuming a vertical silent reascent, the rough hydrophone-relative 3-dimensional sperm whale trajectory can then be reconstructed for the complete dive, by using a single hydrophone.

3:40

4pAB9. Transmitting beam patterns of an echolocating bottlenose dolphin. Thomas G. Muir, Tobias J. Lemerande, Steven R. Baker (Phys. Dept., U.S. Naval Postgrad. School, Monterey, CA 93943, tmuir@olemiss.edu), and Samuel H. Ridgway (U.S. Navy Marine Mammal Program, SPAWARESCEN San Diego, CA 92152)

Measurements on a free-swimming subject, echlocating under a simple go paradigm, were conducted at SSC San Diego, with a linear array of seven simultaneously and individually processed wideband hydrophones (response to 400 kHz), arraigned either vertically or horizontally, and centered on a small underwater video camera, with the video output synchronized with the recording of acoustic data. The measurement apparatus itself served as the test target. Lowering it into the water provided the cue for the blindfolded subject to locate its position in the test pen, swim to the apparatus, and touch it with the tip of its rostrum; whereupon the trainer provided a bridge signal, indicating reward due. During this process, acoustic beampattern measurements were made on echolocation clicks, as a function of frequency, which support previous, non-simultaneous, statistical beampattern measurements. Early in the experiment, the subject emitted echolocation clicks that peaked at 78 kHz, with -3 dB beamwidths of 8 to 10 degrees; but also containing high frequency components (above 135 kHz), with beamwidths that narrowed with increasing frequency, ranging to only a few degrees around 300 kHz, far beyond the subjects hearing range. After several days, the task appeared to become easier and there was much less high frequency content, but the click repetition frequency increased. Wideband noise was then introduced into the test pen, at frequencies below 135 kHz; and as the task became more difficult, the subject resumed transmitting clicks with high frequency components, although the peak frequency remained at 78 kHz, and click rates again increased, to much higher values.

3:55

4pAB10. A juvenile salmon acoustic tracking system (JSATS) for the Columbia River estuary. Lynn McComas (Natl. Marine Fisheries Service, 3305 E. Commerce St., Pasco, WA 99301) and Alex Easton (SAIC, Lynnwood, WA 98037)

The Juvenile Salmon Acoustic Tracking System (JSATS) was developed as a means to estimate the downriver survivability of sub-yearling salmon in the Columbia River. The system consists of a bottom-mounted sonar array which detects signals from acoustic projectors that are surgically implanted in juvenile salmon. JSATS is the result of a development program which began in 2001 when Northwest Fisheries Science Center National Marine Fisheries Service (NMFS) and Science Applications International Corp. (SAIC) developed the top level, biology driven, requirements for the system. SAIC then conducted environmental surveys to characterize the physical and acoustic environments in which JSATS would operate. Battelle PNNL developed the acoustic projector used in JSATS. The signaling characteristics of this projector are driven by the requirement that it not affect the behavior of the 90 mm salmon in which it is implanted, and that it operate continuously for 30 days. SAIC developed the detection array portion of JSATS. The design was driven by the requirements that it be bottom mounted, cabled to shore, and successfully withstand high current and moving sand waves. NMFS and SAIC jointly developed the shore stations and the installation/recovery procedures. Installation of an array spanning the Estuary is planned for April 2005.

4:10

4pAB11. Testing acoustic fish deterrents for use under-ice in arctic lakes. Roberto G. Racca (JASCO Res. Ltd, 2101-4464 Markham St., Victoria, BC, V8Z 7X8 Canada, rob@jasco.com)

Acoustic deterrent technologies can be used in aquatic settings in lieu of physical barriers to keep fish away from potentially harmful industrial operations. The goal of this study was to determine the efficacy of portable, temporary acoustic deterrents as a means of excluding fish from the neighborhood of sub-bottom detonation activities associated with seismic exploration under ice in arctic lakes. In October 2003 trials were conducted in Dolomite Lake near Inuvik, Northwest Territories, Canada on indigenous fish species. Groups of fish were equipped with orally inserted ultrasonic tags, placed in a large experimental net pen and monitored using an acoustic tracking system that produced a detailed three-dimensional swimming pattern for each subject, thereby revealing any behavioral responses. A flex-tensional broadband sound projector driven by digitally synthesized signals was tested as deterrent, and real-time monitoring at two hydrophone sites was used to estimate the local level of insonification throughout the pen volume. Although the study did not identify an overall effective deterrent, sufficient indications of response were observed to support the future testing of a louder projector capable of emitting a tonally modulated sound pattern at frequencies from about 100 Hz to a few kHz. [Work supported by ESRF (esrfunds.org).]

4:25

4pAB12. Design of an oscillating flow test chamber for modeling the fish ear. Charlotte Kotas, Peter Rogers, and Minami Yoda (Mech. Eng. Dept., Georgia Inst. of Technol., Atlanta, GA 30332-0405, gtg227d@mail.gatech.edu)

The acoustically induced oscillating flow fields inside the fish ear, specifically near the otoliths, may provide clues about how fish directionalize underwater sound sources. A new experimental test apparatus has been constructed to study oscillatory and steady streaming flows around model otoliths. The flows are driven by a shaker-piston assembly attached to a 250 mm diameter, 450 mm tall cylindrical test chamber filled with liquid to a depth of 400 mm. This apparatus is capable of generating flow oscillating at frequencies of 1 to 30 Hz with displacements up to a few millimeters within the test chamber. Theoretical predictions of the fluid-borne vibrations in a fluid-filled tube in a vacuum suggest that the flow will have a uniform sinusoidally oscillating velocity field over the central portion of the test section. The flow is visualized in the optically accessible test chamber using small tracer particles; flow velocity fields are measured using particle image velocimetry (PIV). The apparatus is compatible with various working fluids of different viscosities such as water, glycerin and silicon oils to extend the range of attainable Reynolds numbers. Initial results for oscillating flows around basic ''otolith'' geometries such as spheres and spheroids will be shown. [Work supported by ONR.]

4:40

4pAB13. Biomass assessment in commercial catfish ponds. James Chambers (Natl. Ctr. for Physical Acoust., The Univ. of Mississippi, University, MS 38677), C. Douglas Minchew (Mississippi State Univ., Stoneville, MS 38776), and Rachel Beecham (Mississippi Valley State Univ., Itta Bena, MS 38941)

With increasing seafood demand aquaculture is poised to become a major growth industry in the United States in the 21st century. In particular channel catfish represent an approximately $260 million industry in Mississippi with strong growth potential. A major portion of the costs associated with raising channel catfish are related to the cost of feed and aeration which are directly related to the total number of fish being raised in each pond. Crop insurance and bank loans are also contingent upon accurate population estimates. A high frequency, horizontally scanning, active pulse echo sonar system, the Aquascanner, has been developed to estimate pond populations. Commercial catfish ponds are typically 2 to 10 acres and, unlike many offshore fisheries, have fairly shallow depths of 1 to 2 m. The system components and its use will be presented along with results from field tests. [Work supported by US Dept. of Agriculture.]

THURSDAY AFTERNOON, 19 MAY 2005 REGENCY F, 1:30 TO 4:45 P.M.

Session 4pNS

Noise, Architectural Acoustics and Engineering Acoustics: Characterization of Acoustical Materials

Brandon D. Tinianov, Chair
Quiet Solution Inc., 522 Almanor Ave., Sunnyvale, CA 94085

Invited Papers

1:30

4pNS1. On the measurement of the mechanical properties of acoustic porous materials. Noureddine Atalla (Dept. of Mech. Eng., Universit de Sherbrooke, Sherbrooke (QC), Canada J1K 2R1), Franck Sgard, and Luc Jaouen (Ecole Nationale des Travaux Publics de l'Etat, 69518 Vaulx-en-Velin Cedex, France)

This paper discusses the measurements and use of the mechanical properties of porous materials. It starts with a review of the classical methods currently used and concentrates in explaining the assumptions, difficulties and limitations of these methods. Next, it concentrates on two methods: a quasi-static approach accounting for the effect of shape factors on the measured properties and a hybrid inverse method similar to the vibrating beam method used for viscoelastic materials. Through full 3-D numerical simulations, both methods are simulated and their main features, assumptions, and limitations are investigated. In particular, the model uses full poro-elastic modeling and boundary element in order to investigate the effect of acoustic radiation damping and viscous damping. Using various types of materials, comparisons between these two methods and experimental data are also discussed.

1:50

4pNS2. Measurements of the acoustic properties of reticulated vitreous carbon. Ralph T. Muehleisen (Civil and Architectural Eng., Illinois Inst. of Technol., Chicago, IL 60616, muehleisen@iit.edu), C. Walter BeamerIV (Univ. of Colorado, Boulder, CO 80309), and Brandon D. Tinianov (Quiet Solution, Sunnyvale, CA 94085)

Reticulated vitreous carbon (RVC) is an open-cell foam structure composed of amorphous carbon that has a high porosity, strength and rigidity, low bulk thermal conductivity, a high melting point and chemical inertness. These properties make RVC an excellent candidate for use as a material for stacks and regenerators in thermoacoustic devices as well as an acoustic absorber in harsh environments. The four-microphone transfer matrix method was used to measure the characteristic impedance and wave number of 60 to 300 pore-per-inch RVC. From the measurements, a new empirical power law model and acoustic absorption charts were developed. The data are shown to be poorly predicted by the empirical models of Delany and Bazley and only fairly predicted by the Johnson-Allard microstructural model.

2:10

4pNS3. Prediction of the thermal conductivity of fiberglass insulation using propagation constant: A technique overview.
Brandon Tinianov (Quiet Solution, Inc., 522 Almanor Ave., Sunnyvale, CA 94085), Masami Nakagawa, and David Muñoz (Colorado School of Mines, Golden, CO 80401)

This paper describes a novel technique for the prediction of the thermal performance of low density fiber glass insulation and other related fibrous insulation materials using a non-invasive acoustic apparatus. The project motivation is to create an enabling technology for in situ quality control testing of such fiber glass batts during production. Experimental results obtained in the laboratory show excellent correlation between the thermal conductivity and both the real and imaginary components of the propagation constant. Correlation of calculated propagation constant magnitude versus measured thermal conductivity gave an R^2 of 0.94 for the range of typically manufactured fiber glass batt materials. Given the promise of such highly correlated measurements, the acoustic technique could be used to continuously predict the thermal conductivity of the material during its production, replacing current off-line methods. The mechanisms for energy transfer through the materials are distinctly different for the acoustical propagation versus the heat transfer. The nature and behavior of the mechanisms will be reviewed and compared.

2:30

4pNS4. Estimation of the accuracy of measured acoustical parameters of porous materials with an acoustical method.
Xavier Olny (ENTPE–DGCB URA, CNRS 1652, rue Maurice Audin 69518, Vaulx en Velin, France) and Raymond Panneton (Université de Sherbrooke, PQ, Canada J1K 2R1)

Acoustical parameters such as tortuosity, viscous and thermal characteristic lengths, or even static thermal permeability are often required to describe the dynamic behavior of porous materials. Using a middle frequency method, based on the use of a standing waves tube, the direct measurements of the resistivity and porosity, and the accurate measurement of the dynamic density and bulk modulus, these intrinsic parameters can be obtained from analytical inversion of Johnsons and Lafarges models. The interests of the method lie in the simplicity of the apparatus and on the clear separation of viscous and thermal dissipative contributions. This last point specially helps to understand the physics of the medium and check the validity of the assumptions made: the material should not have a strong elastic behavior, and it must fit the models used in the inversion process. In this presentation, the robustness of the method is discussed and estimations of the measurement uncertainties are given. The occurrence of systematic errors coming from the use of semi-phenomenological models for determining intrinsic parameters is tackled. The measurement procedure, and the experimental set-up are detailed, and results obtained on materials with very different properties are presented.

2:50–3:00 Break

Contributed Papers

3:00

4pNS5. Model to simulate the acoustical absorption of laminated fiberglass insulation.
Juan Arvelo, Jr. (Appl. Phys. Lab., Johns Hopkins Univ., 11100 Johns Hopkins Rd., Laurel, MD 20723-6099), Ilene Busch-Vishniac, James West, and Robert Ng (Johns Hopkins Univ., Baltimore, MD 21218-2681)

While fiberglass insulation is known to be a very good acoustical absorber in the audible frequency range, it is not often used for this purpose in rooms because of its appearance and its delicate structural rigidity. Additionally, in some situations, notably medical facilities, the material is inappropriate because of its potential to trap and retain bacteria and dirt. In this work, we report on a study of laminate-coated fiberglass. Although the addition of a laminate coating degrades the acoustical absorption, previous experimental work demonstrated that the composite structure retains a high absorption coefficient. A model of the wall with the installed fiberglass/laminate composite was developed and a wavenumber integration approach was applied to this model to assess the impact of the laminate on the performance of the system. The estimated reflection and transmission losses are computed as functions of auditory frequency and incidence angles with and without the laminate layer to compare the distributions of acoustical energy. This work is performed under a Hafstad fellowship sabbatical appointment.

3:15

4pNS6. Impact of acoustic leakage on the absorption of mono-layer and two-layers porous material.
Franck Castel, Franck Sgard (Laboratoire des sciences de l'Habitat, DGCB URA CNRS 1652, Ecole Nationale des Travaux Publics de l'Etat, 69518 Vaulx-En-Velin Cedex, France), and Noureddine Atalla (Universite de Sherbrooke, Sherbrooke, QC, Canada J1K 2R1)

This paper discusses the effects of small lateral air gaps on the normal incidence absorption coefficient of mono layer and two layers porous materials. Such mounting conditions are responsible for changes in the absorption, leading to dramatic errors in the determination of the acoustics parameter with inverse characterization methods. As this type of mounting conditions is hard to control experimentally, a hybrid finite element-modal method is used to investigate the problem of the porous material inserted in a rectangular wave-guide. At the interface of the material and the wave-guide, coupling between the different domains is accounted for accurately using a modal decomposition. An automatic meshing approach is employed to speed up and guarantee convergence of the method. A large set of materials spanning a wide range of flow resistivities is used for the simulations. The results are presented under the form of charts which makes them an easy to use tool suitable for both inspection and design. Firstly, these charts allow one to identify materials whose normal inci-

dence absorption coefficient is sensitive to lateral air leaks. Secondly, these charts are a helpful tool for designing highly absorptive solutions based on the combination of porous materials and air gaps.

3:30

4pNS7. Sound absorbers with micro-perforated stretched foils and porous materials. Christian Nocke, Catja Hilge (Akustikbuero Oldenburg, Alte Raad 20a, D-26127 Oldenburg, Germany, info@akustikbuero.info), and Jean-Marc Scherrer (Barrisol Normalu SA, F-68680 Kembs, France)

At the spring meeting 2004 in New York first results for the sound absorption of micro-perforated stretched foils have been presented. The classical set-up of a micro-perforated sound absorber consists of a micro-perforated panel in front of an air cavity. The sound absorption coefficient of these set-ups can easily calculated with a high accuracy according to the well-known approximation of D.-Y. Maa if all defining geometrical parameters (diameter of microperforation, distance between orifices, panel thickness and air cavity depth) are known. In this contribution measured sound absorption coefficients of other set-ups with micro-perforated foils as well as combinations with different porous materials will be presented. For these assemblies no closed calculation model exists so far. Comparisons between computer-based calculations of layered absorber set-ups show promising results for some assemblies. Finally different applications in various rooms will be presented. As expected the comparison between theoretical design and the final result show a very high degree of agreement.

3:45

4pNS8. Measured characteristic impedance and propagation constant of some common materials using the two cavity method. Richard Godfrey (Owens Corning, Sci. & Technol., 2790 Columbus Rd., Granville, OH 43023)

The acoustics of porous rigid frame materials is characterized by their characteristic impedance and propagation constant. These properties can be measured using the two cavity method. A sample is placed in an impedance tube with an air space behind the sample and ahead of the rigid termination. Measurements are made in accordance with ASTM E 1050. This procedure is repeated with the same sample, but a different air space. These data are analyzed as outlined by Seybert *et al.* [J. Acoust. Soc. Am. **86**, 637 (1989)] to yield the characteristic impedance and propagation constant. This method was used to characterize fiber glass boards with a range of flow resistivities from 2600 to 56000 mks rayls/m. Repeat measurements were made, and these measurements were compared with the Delaney and Bazley model, [Applied Acoustics (3), 1970]. Measurements were also made for cotton shoddy and cellulose, and power law models were developed. The method produced repeatable results comparable to the Delaney and Bazley model with little effort, and it is a useful method of obtaining property data for modeling tools such as Mechels Acoustics Program System (MAPS).

4:00

4pNS9. Estimation of the absorption coefficients of classroom surfaces by multiple regression. Katrina Scherebnyj and Murray Hodgson (UBC Acoust. & Noise Res. Group, SOEH, 3rd Fl., 2206 East Mall, Vancouver, BC, Canada V6T 1Z3)

The absorption coefficients of room surfaces are important parameters for predicting room sound fields. Of particular interest are classrooms, for which characteristics such as speech intelligibility are essential. Values obtained in laboratory settings may differ significantly from those measured in real-world situations. This study determined surface absorption coefficients in university classrooms and compared them with values given by manufacturers and published literature. Various surface categories were established, and the total area of each was measured in 104 UBC classrooms, along with the early decay times (EDT) in octave bands from 125 to 8000 Hz when the rooms were unoccupied. Average octave-band absorption coefficients for these surfaces were determined from the measured EDTs, using diffuse-field theory. Multiple-regression analysis was then applied, and the data was fit to estimate the average absorption coefficients of each surface category in each octave band. Absorption coefficients were established for seven types of surfaces commonly found in university classrooms; in general they tended to be significantly lower than commonly accepted values.

4:15

4pNS10. Effects of damping materials on heavy-weight impact noise in reinforced concrete floor. Jin Yong Jeon, Young Jeong, and Seung Yup Yoo (School of Architectural Eng., Hanyang Univ., Seoul 133-791, Korea)

Damping materials for reducing heavy-weight floor impact noise in reinforced concrete structures were tested in apartment buildings. The effect of damping materials and an impact isolator were compared in an on-site experiment conducted in a high-rise apartment building. The results showed that the resonance frequency increased and vibration acceleration level decreased when the damping materials were used. Heavy-weight impact sound levels of the structure decreased substantially at 63 Hz, whereas the sound levels of the structure with the impact isolator increased. Subjective evaluation of the perceptual effects of the damping material and the impact isolator were also conducted in the laboratory.

4:30

4pNS11. In situ measurement of absorption of acoustic material with a parametric source in air. Volker Mellert and Roland Kruse (Oldenburg Univ., Inst. Phys., 26111 Oldenburg, Germany)

Measurement of properties of acoustic material in situ is quite desirable in numerous applications. But pulse-echo methods often lack from spurious reflections in a closed-space environment. It is therefore necessary to irradiate the sample material with a highly focussed acoustic beam of defined width. The principle of parametric transducers, which is well known in underwater acoustics, can be used in air as well, if the sound pressure of the ultrasound carrier is high enough. This was recently demonstrated by Hibral and Zakharia (Proc. of the Joint Congr. of CFA/DAGA 04, Strasbourg 2004, p. 541). The new measurement set-up is in principle best suited for in-situ measurements due to the narrow beam-width. But ultrasound intensity, efficiency of generation of audio sound etc. give rise to several limitations. Applications of in-door measurements, in particular at grazing-angle, are presented.

Session 4pPAa

Physical Acoustics, Biomedical Ultrasound/Bioresponse to Vibration and ASA Committee on Standards: Cavitation and Other Mechanical Effects in Biomedical Ultrasound: A Special Session to Honor the Work of Wesley Nyborg II

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

Junru Wu, Cochair
Dept. of Physics, Univ. of Vermont, Burlington, VT 05405

Invited Papers

2:00

4pPAa1. How does ultrasound induce strain related responses in cells? James Greenleaf and Mark Bolander (Mayo Clinic College of Medicine, 200 First St., Rochester MN 55905)

The fact that modulated ultrasound at a pulse intensity of 160 mW/cm^2 can induce fracture healing implies that cells that are sensitive to strain can feel and respond to this signal. A hypothesis that uses Nyborg inspired methods to investigate this effect will be presented along with data from a wide variety of simulations and experimental data. It appears that the response is mediated through the cytoskeleton and may be associated with interaction of the sound with focal adhesion complexes at one end of the skeletal fibers for high frequency responses and with the nuclear membrane at the other end of the fibers for low frequency responses. Although this hypothesis is nascent it is supported with experimental data from the literature.

2:20

4pPAa2. The influence of agent delivery mode on cardiomyocyte injury induced by myocardial contrast echocardiography in rats. Douglas Miller, Chunyan Dou, and William Armstrong (Univ. of Michigan, Ann Arbor, MI 48109)

Bioeffects induced in rat hearts by the interaction of ultrasound with contrast agent gas bodies depends on user controlled parameters. This study examined the influence of contrast agent delivery mode on cardiomyocyte injury. Hairless rats were anesthetized and mounted vertically in a water bath. Evans blue dye was injected as stain for cardiomyocyte injury. The contrast agent (Definity, Amersham Health Inc., Princeton NJ) was diluted in saline and injected IV at 20 or 80 μL/kg either as bolus or continuous infusion. Echocardiography was performed with a GE Vingmed System V in a short axis view with 1:4 or 1:16 ECG triggering at 1.5 MHz for 5 or 20 min. The peak rarefactional pressure amplitude (PRPA) was 2.0 MPa. Fluorescent stained cells were scored on frozen sections of heart samples obtained 24 hr after scanning. Five min infusion led to 3.4 times ($P<0.01$) more cardiomyocyte injury than bolus injection for the same 20 μL/kg dose. Contrast agent dose appeared to be a more important parameter than infusion rate and number of triggers. Contrast agent delivery mode, as well as dose and PRPA, has a significant influence on the bioeffects potential of myocardial contrast echocardiography. [Work supported by NIH grant EB00338.]

2:40

4pPAa3. Nonlinear model for detection of bubble activity from pulse-echo ultrasound. Emad S. Ebbini (200 Union St. SE, Rm. 4-174, Minneapolis, MN 55455)

Cavitation and transient cavitation plays an important role in modern application of biomedical ultrasound, both in therapeutics and diagnostics. Special imaging and signal processing methods have been recently introduced for the detection and localization of cavitation and other nonlinear phenomena for improving image quality and/or characterization of therapeutic outcome of the ultrasound-based treatments. We have recently developed a model-based approach for separating the linear and nonlinear components of the pulse-echo data. In its simplist form, the approach separates the linear and quadratic signal components from the echo data. The latter is related to the sum and difference frequency interactions in the nonlinear medium. When used in conjunction with elongated coded transmit waveforms (e.g., chirp signals), the approach provides significant separation of bubble activity from tissue nonlinear components. Application to ultrasound contrast-agent imaging and detection of HIFU-induced lesions will be discussed.

3:00

4pPAa4. Effect of clinical diagnostic ultrasound pressure amplitude and pulse repetition frequency on echogenic liposome destruction. Denise A. B. Smith, Tyrone M. Porter, Christy K. Holland (Dept. of Biomed. Eng., Univ. of Cincinnati, Medical Sci. Bldg, Rm. 6152, 231 Albert Sabin Way, Cincinnati, OH 45267-0586, smitdn@ email.uc.edu), Shao-Ling Huang, Robert C. MacDonald (Northwestern Univ., Evanston, IL 60208), and David D. McPherson (Northwestern Univ., Chicago, IL 60611)

Liposomes are submicron-sized phospholipid vehicles that contain both gas and fluid. Entrapped microbubbles within the liposome cause them to be echogenic. With antibody conjugation and thrombolytic drug incorporation, such as rt-PA, these liposomes can be used as novel targeted diagnostic ultrasound echo contrast agents to deliver a drug locally. In this study, the echogenicity of

unconjugated liposomes suspended in distilled water was assessed by measuring the decay rate of backscattered intensity using a Philips HDI5000 diagnostic ultrasound scanner with a 12-5 MHz linear array. Pulses of ultrasound with a 7-MHz center frequency and derated peak rarefaction pressure amplitudes of 0.16, 0.35, and 0.61 MPa were used to expose the liposomes. Pulse repetition frequencies in the range of 6.7 kHz to 13.3 kHz were employed at frame rates between 16 and 35 Hz, respectively. The average backscatter intensity was determined from a 20 mm^2 region-of-interest in each image frame. An exponential decay was fit to each data set. Ultrasound pulses with higher acoustic pressures and higher pulse repetition frequencies destroyed the liposomes at a faster rate.

3:20

4pPAa5. Cavitation detection during ultrasound-assisted thrombolysis in porcine blood clots. Saurabh Datta (Dept. of Biomed. Eng., Univ. of Cincinnati, Cincinnati, OH 45267), Louis E. McAdory (Univ. of Cincinnati), Jun Tan, and Christy K. Holland (Univ. of Cincinnati, Cincinnati, OH 45267-0586, Christy.Holland@uc.edu)

Substantial enhancement of recombinant tissue plasminogen activator (rt-PA) mediated thrombolysis can be achieved with exposure of a thrombus to pulsed ultrasound. However, the mechanism of this interaction has not yet been elucidated. In this work cavitation is investigated as a possible mechanism for enhancement in 120-kHz pulsed ultrasound-assisted thrombolysis. Porcine blood clots were immersed in plasma as control and exposed to rt-PA, ultrasound (0.35 MPa, 80% duty cycle, 1667 Hz pulse repetition frequency), or a combination of rt-PA and ultrasound. A confocally aligned active and passive cavitation detection system was employed to detect subharmonic emissions from stable cavitation and broadband superharmonic emissions from inertial cavitation. After exposure, clot mass loss was determined, and clots were subjected to immunohistochemical analysis of fibrin degradation. Spatial investigation of cavitation thresholds inside the clot, on the clot surface, and in the fluid surrounding the clot showed the threshold to be lowest on the clot surface. Stable cavitation was detected in clots exposed to ultrasound alone and a combination of rt-PA and ultrasound. Curiously, inertial cavitation was detected only in samples containing rt-PA. The presence of both stable and inertial cavitation correlated with increased clot mass loss and a distinct pattern of fibrin degradation on histologic evaluation.

3:40–4:00 Break

4:00

4pPAa6. Occlusive thrombosis in the rabbit auricular vein *in vivo* targeted by induction of intralumenal cavitation using HIFU and ultrasound contrast agent. Andrew A. Brayman, Juan Tu, Thomas Matula, Lawrence A. Crum (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105-6698), Joo Ha Hwang, and Michael B. Kimmey (Univ. of Washington, Seattle, WA 98195)

Hypotheses tested: (1) inertial cavitation [IC] could be induced in the venous lumen *in vivo* by combined use of intravascular microbubble contrast agent and transcutaneous application of 1-MHz high intensity focused ultrasound [HIFU] of very low duty factor, and that IC activity could be detected and quantified *in vivo* as in earlier *in vitro* studies via passive cavitation detection methods; (2) robust IC activity would damage the venous endothelium in treated regions; (3) endothelial damage would be proportional to the IC dose developed in the region; (4) severe local endothelial damage alone may be sufficient to induce occlusive thrombosis, or may sensitize the region to low systemic doses of prothrombotic agents, and (5) biologically significant temperature rises and attendant thermal bioeffects in the vessel and perivascular tissues would not occur, even under the highest amplitude acoustic conditions applied. Each hypothesis was supported by the data. The principal result was that under treatment conditions involving very high peak negative acoustic pressures and contrast agent, treated areas thrombosed acutely but non-occlusively. When fibrinogen was administered locally after such treatment, occlusive thrombi formed acutely and only in the treated region, a response observed with none of the other treatments.

4:20

4pPAa7. Controlled nucleation of microcavitation with laser-illuminated nano-particles. Ronald A. Roy, Caleb H. Farny, Tianming Wu, Todd W. Murray (Dept. of Aerosp. and Mech. Eng., Boston Univ., Boston, MA 02215), and R. Glynn Holt (Boston Univ., Boston, MA 02215)

The safe utilization of controlled cavitation in HIFU therapy requires the presence of nucleation sites for bubble formation. Effective cavitation nuclei do not exist in most tissues; nucleation thresholds pressures in excess of 4–5 MPa have been reported. We investigate the efficacy of transient vapor cavity generation from laser-illuminated gold nano-particles as a means for nucleating cavitation with high-intensity focused ultrasound. An acrylamide tissue phantom seeded with 82-nm diameter gold particle was exposed to 20 ns pulses from a 532 nm Nd:Yag laser. Laser firing was precisely synchronized with a pulsed 1.1 MHz HIFU pressure field. Acoustic emissions from inertial cavitation were detected by a 15 MHz focused transducer at a laser energy of 0.10 mJ/pulse and a HIFU peak-negative focal pressure as low as 0.92 MPa. In comparison, a peak-negative focal pressure of 4.50 MPa was required to nucleate detectable cavitation without laser illumination; nano-particles were present in both cases. Since the particles are durable, one can re-activate them as needed, essentially yielding cavitation nuclei on demand. [Work supported by the Dept. of the Army (award No. DAMD17-02-2-0014) and the Center for Subsurface Sensing and Imaging Systems (NSF ERC Award No. EEC-9986821).]

Contributed Papers

4:40

4pPAa8. Sonochemical synthesis and modification of protein microspheres. Kenneth S. Suslick, Farah J.-J. Toublan, and Elizabeth M. Dibbern (School of Chemical Sci., Univ. of Illinois at Urbana-Champaign, 600 S. Mathews Av., Urbana, IL 61801, ksuslick@uiuc.edu)

The need for organ-targeted delivery of drugs and imaging agents creates an interest in biocompatible, biodegradable vesicles. We make protein microspheres using high-intensity ultrasound; these microspheres have a protein shell and a hydrophobic interior making them ideal for delivering hydrophobic materials. We have previously shown that various proteins, e.g., bovine serum albumin (BSA), form a microsphere shell stabilized by inter-protein cross-linking of cysteine residues. In this study, we explore methods to modify the surface of these microspheres by (1) nanoparticle incorporation and (2) layer by layer electrostatic adhesion of macromolecules. We have explored applications of these microspheres for organ targeted imaging using both MRI and optical coherence tomography (OCT).

4:55

4pPAa9. Nonlinear dynamics of gas bubbles in soft tissue. Xinmai Yang and Charles Church (Natl. Ctr. for Physical Acoust., Univ. of Mississippi, University, MS 38677)

Understanding the behavior of cavitation bubbles driven by ultrasonic fields is an important problem in biomedical acoustics. The previous studies are largely limited by availability of experimental data on soft tissue. To approach this problem, we combine the Keller-Miksis equation for nonlinear bubble dynamics with the linear Voigt model for viscoelastic media. Using experimentally determined values of viscoelastic properties of soft tissue as a guide, the effects of elasticity on bubble oscillations are studied. The inertial cavitation thresholds are determined using a criterion of $R_{max}/R_0=2$, and subharmonic emissions from an oscillating bubble are calculated. The results show that the presence of the elasticity increases the threshold pressure for inertial cavitation, and subharmonic signals only can be detected in a certain region of radii and driving pressures at a given frequency. These results should be useful in cavitation detection and bubble-enhanced imaging work. The model also could be used to determine values for the viscoelastic properties of soft tissue.

THURSDAY AFTERNOON, 19 MAY 2005 PLAZA A, 2:00 TO 4:15 P.M.

Session 4pPAb

Physical Acoustics: Computational Methods in Physical Acoustics

T. Douglas Mast, Chair

Univ. of Cincinnati, Biomedical Engineering, 231 Albert Sabin Way, Cincinnati, OH 45267-0586

Contributed Papers

2:00

4pPAb1. Acoustic Casimir pressure for arbitrary media. Raul Esquivel-Sirvent, Luis Reyes, and Jeffrey Barcenas (Instituto de Fisica, UNAM Apdo. Postal 20-364, Mexico DF 01000, Mexico, raul@fisica.unam.mx)

We derive a general expression for the acoustic Casimir pressure between two parallel slabs made of arbitrary materials and whose acoustic reflection coefficients are not equal. The formalism is based on the calculation of the local density of modes using a Green's function approach. The results for the Casimir acoustic pressure are generalized to a sphere/plate configuration using the proximity theorem.

2:15

4pPAb2. Use of complex source points to simplify numerical Gaussian beam synthesis. Stephen Forsythe (Naval Undersea Warfare Ctr., 641 Middle Rd., Portsmouth, RI 02871, stephen.forsythe@verizon.net)

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 Greens 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 $R=[Xr+iXi,Yr+iYi,Zr+iZi]$. When this complex source point is used in the free-space Greens 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 Greens function will be explored.

2:30

4pPAb3. Causality and acoustic group velocity in dispersive media. Joel Mobley (U.S. Army Res. Lab., 2800 Powder Mill Rd., Adelphi, MD 20783, jmobley@arl.army.mil)

This talk examines two specific issues related to causality and the group velocity of acoustic pulses in dispersive media. First, the causal prediction of group velocity from attenuation with finite-bandwidth Kramers-Kronig (K-K) relations is discussed. Without extrapolating beyond the measurement bandwidth, the causal linkages of resonant-type data are established using expressions derived from the acoustic K-K relations to predict group velocity and slope of attenuation (frequency derivative of attenuation). These predictions provide a stricter test of causal consistency than the determination of the phase velocity and attenuation coefficient due to their shape-invariance with respect to subtraction constants. Secondly, conditions under which the group velocity is the velocity of the peak in the envelope of a acoustic pulse are described. The spatial and temporal signatures of acoustic pulses with arbitrarily large (e.g., superluminal) and negative group velocities are also demonstrated. These signatures are shown to be consistent with both simple and relativistic causality.

2:45

4pPAb4. Ultrasonic backscatter in two-dimensional domains. Goutam Ghoshal and Joseph A Turner (Dept. of Eng. Mech., W317.4 Nebraska Hall, Univ. of Nebraska–Lincoln, Lincoln, NE 68588-0526)

The scattering of elastic waves in polycrystalline materials is relevant for ultrasonic materials characterization and nondestructive evaluation (NDE). Ultrasonic backscatter measurements are used widely to extract the microstructural parameters such as grain size and also to detect flaws in materials. Accurate interpretation of experimental data requires robust scattering models. Line transducers are often used for ultrasonic experiments such that an appropriate model for these two-dimensional problems is needed. Here, a theoretical expression for the temporal backscatter is derived for such domains under a single-scattering assumption. The result is given in terms of transducer and microstructural parameters. In addition, the problem is examined in terms of numerical simulations using Voronoi polycrystals that are discretized using finite elements in a plane-strain formulation. The material properties of the individual Voronoi cells are chosen according to appropriate material distributions. Such numerical models also allow scattering theories, including the one discussed here, to be examined for well-controlled microstructures. Example numerical results for materials with varying degrees of scattering that are of common interest are presented. The numerical results are compared with the theory developed with good agreement. These results are anticipated to impact ultrasonic NDE of polycrystalline media. [Work supported by US DOE.]

3:00

4pPAb5. Local dynamical effects in inhomogeneous elastic waaveguides. Victor T. Grinchenko (Inst. of Hydromechanics of NAS of Ukraine, 8/4 Zhelyabov Str., 03680 Kiev, Ukraine)

The properties of normal waves in elastic waveguides are essentially different than properties of normal waves in acoustical waveguides. As the physical reason of the differences one can indicate the specific phenomena of reflection of elastic waves from a free surface. When an elastic wave reflects from the surface, the energy of longitudinal waves transforms to shear waves and vice versa. It is shown that the conversion of the wave motion types sets condition for generation of local disturbances in wave fields when normal waves are reflected from geometrical and physical inhomogeneities. The paper presents quantitative and qualitative analysis of the local effects for different cases. It was shown that such effects occur in a simple elastic waveguide. The specific features and physical meaning of the edge resonance effect in semi-infinite waveguides is presented. It was shown that for the waveguides with geometrical and physical inhomogeneous interesting phenomena of anomalous transparency of discontinuity can occur for any frequency bands. For other frequency bands the same discontinuity can be as an effective reflector of incident normal wave.

3:15

4pPAb6. Simplified series expansions for radiation from a baffled circular piston. T. Douglas Mast (Dept. of Biomed. Eng., Univ. of Cincinnati, Cincinnati, OH 45267-0586, doug.mast@uc.edu) and Feng Yu (Univ. of Cincinnati, Cincinnati, OH 45221-0070)

Computation of acoustic radiation from a baffled circular piston continues to be an active area of investigation, both as a canonical problem and because of numerous practical applications. For time-harmonic radiation, exact series expansions are an attractive approach because they do not require numerical integration or limiting approximations. Here, series expansions due to Hasegawa, Inoue, and Matsuzawa [J. Acoust. Soc. Am. **74**, 1044–1047 (1983); **75**, 1048–1051 (1984)] are shown to reduce to simpler expressions suitable for numerical computations. For the region $r \geqslant a$, where a is the piston radius and r is the distance from the piston center, an exact solution is given by a series of spherical Hankel functions and Legendre polynomials with explicit, closed-form, position-independent coefficients. For the paraxial region $w \leqslant a$, where w is the distance from the piston axis, a second exact series expansion is valid for all axial distances z and reduces to the known analytic solution for $w=0$. These two expansions allow the radiated field to be computed at any point, with rapid convergence except for points near the circle bounding the piston. Example numerical results illustrate application of this method to ultrasonic sources.

3:30

4pPAb7. Fast calculations of time-harmonic near field pressure distributions produced by triangular sources. Duo Chen and Robert McGough (Michigan State Univ., 2120 Eng. Bldg., East Lansing, MI 48824, mcgough@egr.msu.edu)

An analytical expression is derived for the near field pressure response to a sinusoidal excitation for a triangular piston. The analytical expression eliminates a numerical singularity from the impulse response and therefore achieves much more rapid convergence than the impulse response. This results in shorter computation times relative to the impulse response for a given peak error value. These fast-converging expressions are represented by three integrals, where each integral expression corresponds to an edge of the triangle. Pressure fields are evaluated within a grid that includes the face of the triangular source, and computation times are compared with the impulse response for given values of the peak error. For a specified peak error of 1%, the rapidly converging expressions are at least 50% faster than the impulse response for equilateral triangles with sides ranging from one-half wavelength to 8 wavelengths. For a specified peak error of 10%, the rapidly converging expressions are at least 120% faster than the impulse response when applied to the same group of equilateral triangles. Different reductions in computation time are achieved by other piston and grid geometries.

3:45

4pPAb8. Tangible interactive interface using acoustic time reversal process. Ros K. Ing, Nicolas Quieffin (Sensitive Object, Res. and Development Dept., 10 rue Vauquelin 75005 Paris, France), Stefan Catheline, and Mathias Fink (Univ. Paris VII, 75231 Paris cedex 05, France)

Time reversal in acoustic is a very efficient solution to focus sound back to its source in a wide range of material including reverberating media. It expresses the following physical properties: a wave still have the memory of its source location. The concept presented in this paper consists in detecting the acoustic waves in solid objects generated by a simple human touch. In a second step, the information related to the source location are extracted from a virtual time reversal experiment in the computer. Then, an action (turn on the light or a CD player) is associated to each location. Thus, the whole system transforms solid objects into interactive interfaces. Compared to the existing acoustic techniques, it presents the great advantage to be simple and easily applicable on inhomogeneous objects with any shape. The number of possible touch locations at the surface of objects is shown to be directly related to the mean wavelength of the detected acoustic wave.

4:00

4pPAb9. Telecommunication in a disordered environment with iterative time reversal. Gabriel Montaldo, Geoffroy Lerosey, Arnaud Derode, Arnaud Tourin (LOA, ESPCI, CNRS, 10 rue Vauquelin, 75005 Paris, France), Julien de Rosny, and Mathias Fink (CNRS, 75005 Paris, France)

Recent researches in the area of wave propagation in random media applied to telecommunications showed that, contrary to intuition, reverberation or scattering of waves in a disordered medium can actually help to increase the information transfer rate. The key element therein is the ability of a communication system to exploit independent channels of propagation. We present a method to transmit digital information through a highly scattering medium. It is based on iterations of a time-reversal process, and permits to focus short pulses, both spatially and temporally, from a base antenna to different users. This iterative technique is shown to be more efficient (lower inter-symbol interference and lower error rate)

than classical time-reversal communication, while being computationally light and stable. Experiments are presented: digital information is conveyed from 15 transmitters to 15 receivers by ultrasonic waves propagating through a highly scattering slab. From a theoretical point of view, the iterative technique achieves the inverse filter of propagation in the subspace of non null singular values of the time reversal operator. We also investigate the influence of external additive noise, and show that the number of iterations can be optimized to give the lowest error rate.

THURSDAY AFTERNOON, 19 MAY 2005 — REGENCY C, 1:00 TO 5:00 P.M.

Session 4pPP

Psychological and Physiological Acoustics: Localization, Binaural Hearing and Physiology (Poster Session)

Beverly A. Wright, Chair
Northwestern Univ., Dept. Communication Science and Disorders, Evanston, IL 60208

Contributed Papers

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

4pPP1. Auditory and visual performance in an open sound field. Kim F. Fluitt, Tomasz Letowski, and Timothy Mermagen (U.S. Army Res. Lab., HRED-VAP, Bldg 520, Aberdeen Proving Ground, MD 21005)

The ability to estimate a distance to a sound source in an open field is an important element of situational awareness on the battlefield, and is affected by many technical and environmental conditions. A limited body of knowledge regarding auditory perception of sources located over long distances makes it difficult to develop models predicting auditory behavior on the battlefield. Results of previous studies have shown that people greatly underestimate distances. However, it is not known if there is a connection between a visual estimation, verbally reporting an estimate, and auditory estimation. The purpose of the present study was to compare listeners' visual, verbal reporting, and auditory estimates to sound sources 25 to 200 m from the listening position. Data were collected for auditory as well as visual estimations of distance expressed verbally. Twenty-two subjects (men and women, ages 18–25) participated in the study. Six types of sounds were presented from five loud speakers in random order. Test results indicate that auditory and visual estimation errors increased with distance and that at some distances the visual estimation errors exceeded auditory estimations errors, but the differences were not statistically significant. Specific results will be presented.

4pPP2. Effects of different amounts of brief training and rest on the generalization of learning from interaural-level-difference to interaural-time-difference discrimination. Jeanette A. Ortiz and Beverly A. Wright (Dept. of Commun. Sci. and Disord. and Inst. for Neurosci., Northwestern Univ., 2240 Campus Dr., Evanston, IL 60208-3550)

Training-induced improvements on perceptual skills can be enhanced by increasing the amount of training and by resting between training and testing. However, how these two factors affect the generalization of learning from a briefly trained condition to an untrained one is unknown. Here, listeners were trained on an interaural-level-difference (ILD) discrimination condition (4-kHz tones), then were tested for generalization to an interaural-time-difference (ITD) discrimination condition (0.5-kHz tones). The amount of training and the time between training and testing differed across four groups. Listeners tested 10 hours after training had significantly lower ITD discrimination thresholds than naive listeners ($n=94$), regardless of whether training lasted for 20 min ($n=14$) or 2 hs ($n=11$). Thus, when there was a long time between training and testing, learning generalized from ILD to ITD discrimination regardless of the amount of training. In contrast, listeners tested immediately after training on ILD discrimination showed generalization to ITD discrimination with 20 min ($n=14$), but not 2 h ($n=14$), of training, suggesting that without rest between training and testing, longer training disrupted generalization. Hence, increased training interfered with generalization from ILD to ITD discrimination for short, but not long, delays between training and testing. [Work supported by NIH.]

4pPP3. The impact of helmet design on sound detection and localization. Angelique Scharine (U.S. Army Res. Lab.–HRED)

The shape and absorptiveness of a ballistic helmet changes the sound signal reaching the soldier's ear. Although the attenuation of current helmets is minimal, soldiers have reported removing their Personnel Armor System for Ground Troops (PASGT) helmet when it was necessary to determine sound source direction. The Advanced Combat Helmet (ACH) has reduced ear coverage and absorptive internal padding that decreases localization errors. In an effort to provide more integrated headgear, the U.S. Army is testing a new, Scorpion (R4) helmet that has rings that allow the insertion of earmuffs and a modified version without the rings (R4-R). Two studies were conducted to measure the effect of these four designs on sound detection and localization. None of the helmets differed significantly in the degree of attenuation nor did attenuation correlate with localization. However, localization performance varied greatly. The current ACH and the R4-R had the least impact on localization. The older, unpadded PASGT, caused high error rates despite having the lowest attenuation. Despite being otherwise similar to the R4-R, the R4 impaired localization just as much as the PASGT and its rings increased the attenuation by more than a decibel.

4pPP4. Principal components analysis interpolation of head related transfer functions using locally-chosen basis functions. Jacob W. Scarpaci and H. Steven Colburn (Hearing Res. Ctr. and Dept. of Biomed Eng., Boston Univ., 44 Cummington St., Boston, MA 02215)

Spatial interpolation between measured head related transfer functions (HRTFs) may be accomplished by using principal components analysis (Kistler and Wightman, 1992). The data set can be reduced by using only

the most important basis functions (BFs) when reconstructing the magnitude spectrum of the HRTFs. Relatively good performance can be achieved by using a small fraction of the BFs required for an error free representation. In this study a comparison is made between two methods for choosing which basis functions are included in the HRTF reconstruction for a given position. The usual approach is to choose the subset of BFs which have the largest impact (common variance) on the global set of measured HRTFs. This will be compared to a method in which the subset of BFs are chosen to have the largest impact on the measured HRTFs in the area local to the position being reconstructed. Advantages and disadvantages to the two methods are discussed. [Work supported by NIH DC00100.]

4pPP5. Hearing space: Identifying rooms by reflected sound. Lawrence D. Rosenblum and Ryan L. Robart (Univ. of California, Riverside, Riverside, CA 92521)

The acoustic engineering literature has shown that listeners are sensitive to the sound reflecting properties of spaces so as to judge the general dimensions and acoustic quality of the space [e.g., J. S. Bradley, R. D. Reich, and S. G. Norcross, J. Acoust. Soc. Am. **108**, 651–661 (2000)]. There is also speculation on the perceptual processes listeners use to apprehend properties of spaces [e.g., R. K. Clifton, R. L. Freyman, and J. Meo, Percept. Psychophys. **64**, 180–188 (2003)]. Still, there has been little research testing whether listeners can simply recognize different types of rooms based on reflected structure. To examine this issue, five different sound sources were binaurally recorded in four different acoustic spaces. The sources ranged from human speech to white noise bursts. The acoustic spaces included a large indoor gymnasium, a moderately-sized classroom, a public restroom, and a small laboratory room. Untrained listeners heard each sound over headphones as they looked at photographs of the four rooms. They were asked to choose the room in which each sound was recorded. Overall, listeners were quite good at the task, with accuracy dependent on sound source. The results indicate a sensitivity to complex ambient structure not often addressed in the psychoacoustics literature.

4pPP6. Effects of various reverberant conditions on speech intelligibility. Suzanne P. Carr and H. Steven Colburn (Hearing Res. Ctr. and Dept. of Biomed. Eng., Boston Univ., 44 Cummington St., Boston, MA 02215)

Understanding speech in complex environments is a problem for many hearing impaired listeners, yet the specific characteristics of the listening environment which contribute most to listening difficulties are not well understood. Using virtual stimuli, a large range of conditions are tested on a small group of listeners, both normal and hearing impaired, and speech intelligibility thresholds are obtained. Systematic comparisons of the different conditions are made. The conditions that are varied are the amount of reverberation in the virtual room (ranging from pseudo-anechoic to moderately reverberant), the presence or absence of strong, early reflections such as from a wall and/or tabletop, the direction of the target speech relative to the listener, the distance of the target speech from the listener, and the number and location of competing speech maskers. Care is taken to ensure that the conditions are within the bounds of realistic listening. The validity of the use of virtual stimuli is also explored. [Work supported by NIH DC00100.]

4pPP7. The effect of spatial configuration in a divided attention task. Virginia Best, Antje Ihlefeld, and Barbara Shinn-Cunningham (Hearing Res. Ctr., Boston Univ., 677 Beacon St., Boston, MA 02215, ginbest@cns.bu.edu)

The effect of spatial separation on the ability of listeners to report keywords from two simultaneous talkers was examined. The talkers were presented with equal intensity at a clearly audible level, but were processed to minimize their spectral overlap and reduce energetic interference. The two talkers were presented with various angular separations around references of $-45°$, $0°$, or $45°$ azimuth. Overall, performance did not vary dramatically with spatial configuration, but depended on spatial separation and reference direction. With the talkers in front or to the left, performance tended to first increase and then decrease with increasing separation. With both talkers to the right, performance tended to improve monotonically with increasing separation. The relative levels of the two talkers at each ear in each configuration partially accounted for results. For each talker a different ear contained a signal-to-noise ratio advantage, and performance was positively correlated with the mean signal-to-noise ratio across the two "better ears." Thus, when tracking two sources simultaneously, listeners may make use of the information at the two ears independently. Furthermore, the drop in performance for some large talker separations may reflect increased difficulty in following sources that do not fall within a single "spotlight" of spatial attention.

4pPP8. Localization of amplitude and frequency modulated sounds. Mark Ericson (Air Force Res. Lab., AFRL/HECB Bldg. 441, 2610 Seventh St., Wright–Patterson AFB, OH 45433, mark.ericson@wpafb.af.mil)

Everyday sounds, such as speech, music, and environmental noises, vary in level and frequency. The dynamic cues of natural sounds help to identify and segregate multiple sources. Amplitude and frequency modulation of simple pure tone complexes were manipulated in several experiments to measure their effects on auditory localization. The tonal complexes had 500 Hz fundamental frequencies and 26 components, providing 13 kHz in bandwidth. The complexes were either amplitude modulated, frequency modulated, co-modulated in frequency and amplitude, or not modulated. In addition, phase of the carrier and modulation frequencies were randomized to aid in sound source segregtion. Three normal hearing subjects participated in the task of localizing and identifying up to four simultaneous sounds. Amplitude modulated sounds provided the best localization acuity and identification data. Comodulated, frequency modulated and no modulation were found to provide lower amounts of acuity performance in decreasing order.

4pPP9. Different interaural level difference processing with complex sounds and pure tones. Yuxuan Zhang and Beverly A. Wright (Dept. of Commun. Sci. and Disord. and Inst. for Neurosci., 2240 Campus Dr., Northwestern Univ., Evanston, IL 60208-3550)

Interaural level differences (ILDs) are one of the primary cues to sound source position on the horizontal plane. Most studies on human performance with ILDs use simple sounds such as pure tones. However, naturally occurring acoustic stimuli usually have complex waveforms. Here human learning of ILD discrimination was examined with complex waveforms. Sixteen listeners were trained 1 hr/day for 9 days on ILD discrimination with a 4-kHz tone sinusoidally amplitude modulated at 0.3 kHz conveyed through headphones. Before and after training, they were tested, together with sixteen untrained controls, on the trained condition and five related untrained conditions (three with amplitude modulated tones, two with pure tones). The trained listeners improved significantly more than controls on all conditions with amplitude-modulated tones, but not on those with pure tones. The lack of generalization of ILD learning from modulated to pure tones suggests that practice modified ILD processing in a region that encodes complex sounds with a variety of carrier frequencies and modulation rates, but not pure tones. Thus, these findings suggest that waveform complexity is an important factor in ILD processing, and must be considered when evaluating human ILD performance. [Work supported by NIH.]

4pPP10. Sources of variation in masking by competing speech. Richard L. Freyman, Karen S. Helfer, and Uma Balakrishnan (Dept. of Commun. Disord., Univ. of Massachusetts, Rm. 3 Arnold House, Amherst, MA 01003, rlf@comdis.umass.edu)

The literature on speech recognition within a competing speech environment shows great variation with respect to the amount of masking produced, as well as variation in the improvement realized from spatial separation of the target and masking speech. Presumably, the sources of variation include differences in methodology and stimuli used by different researchers. In the current study the target speech stimuli were nonsense sentences spoken by a female talker. The competing speech was created from similar nonsense sentences recorded by 10 other female speakers. Five different maskers were created, each a combination of two of the 10 talkers who had similar fundamental frequencies. The results showed that the variation in masking produced by the different two-talker combinations was substantial only when target and masker were presented from the same loudspeaker. As a consequence, the observed benefit from spatial separation varied widely among the different maskers, and was ordered according to fundamental frequency in an unexpected way. The results are interpreted as showing greater variation in informational masking than in energetic masking. Also tested was the influence of different ways of blocking the trials, which effectively manipulated the uncertainty of the masking stimulus from trial to trial. [Work supported by NIH DC01625.]

4pPP11. Binaural informational masking release in children and adults. Joseph W. HallIII, Emily Buss, and John H. Grose (Univ. of North Carolina at Chapel Hill, Chapel Hill, NC 27599, jwh@med.unc.edu)

Informational masking associated with masker spectral uncertainty can sometimes be reduced by providing cues that promote perceptual segregation of the signal from the masker. Our previous research using the multi-burst-same (MBS) paradigm of Kidd *et al.* [J. Acoust. Soc. Am. **95**, 3475–3480 (1994)] indicated that all adults and most children (5–9 YO) showed reduced informational masking when monaural segregation cues related to gating asynchrony or spectro-temporal coherence were available. The present study examined developmental effects in children aged 6–10 years for release from informational masking in a condition where perceptual segregation was manipulated using a spatial hearing cue. In the informational masking condition, the pure-tone signal and the MBS masker components were presented to the left ear. In the masking release condition, the masker components were also presented to the right ear synchronously (at a level 10-dB higher than in the left ear) so that the signal would be lateralized to the left ear and the masker would be lateralized to the right ear. Results indicated that most adults achieved masking release in this condition, but that none of the children did. These results are consistent with previous findings indicating protracted development for informational masking release related to spatial hearing cues.

4pPP12. Measuring the binaural temporal window. Andrew J. Kolarik and John F. Culling (School of Psych., Cardiff Univ., Tower Bldg., Park Pl. Cardiff, CF10 3AT, UK)

Sensitivity to small ITDs in detection and discrimination tasks was measured. In experiment 1, four listeners performed 3 tasks. The first 2 were 4I- and 2I-2AFC adaptive detection tasks. The third was a 2I-2AFC adaptive discrimination task. All stimuli were 100 ms noise bursts. Reference stimuli were diotic, and target stimuli contained a probe duration of 64, 32, 16, 8, 4, or 2 ms, which carried an interaural time delay. These probes were temporally fringed with diotic noise. Thresholds for the discrimination task were significantly higher than for the detection tasks. In experiment 2, psychometric functions were obtained from 4 participants for the six probe durations using the same 2I-2AFC detection and discrimination tasks. Binaural temporal windows were fitted to the data using a variety of fitting functions. Fits to the detection task data demonstrated narrow tips but unmeasurably long skirts. In the discrimination task neither parameter could be accurately measured, suggesting that the overall stimulus duration was too short to encompass the window. A stimulus length of 500 ms and probe durations of 256, 128, 64, 32 and 16 ms allowed both parameters to be measured. The resulting equivalent rectangular duration was approximately 50 ms.

4pPP13. Individual differences in the masking level difference (MLD) with a narrowband masker at 500 or 2000 Hz. Emily Buss, Joseph W. HallIII, and John H. Grose (Univ. of North Carolina at Chapel Hill, 130 Mason Farm Rd., 1115 Bilinformatics Bldg., CB7070, Chapel Hill, NC 27599, ebuss@med.unc.edu)

The MLD for a narrowband masker is associated with marked individual differences. This study examines factors that might account for these individual differences, including binaural temporal resolution and sensitivity to interaural cues based on time or level. MLD data were collected for 50-Hz wide maskers at 500 and 2000 Hz, gated on for 400-ms. The signal was a pure tone at the center frequency of the masker, of either brief (15-ms) or long (200-ms) duration. Brief signals were coincident with either a local minimum or maximum in the pattern of inherent amplitude modulation. Sensitivity to interaural time and level cues was assessed using similar stimuli, with the exception that the long-duration signal was a 50-Hz band of noise, generated to provide just interaural time cues, just interaural level cues, or both types of cues. Binaural temporal windows were estimated based on pure tone detection thresholds for a brief 500-Hz tone presented at different points in time relative to an abrupt interaural phase transition in a masking noise, bandpass filtered between 100 and 2000 Hz. Relationships between MLD results and those of binaural temporal windows and interaural cues of time and level will be discussed.

4pPP14. The discrimination of interaurally correlated noise bands. Barrie Edmonds and John Culling (School of Psych., Tower Bldg., Cardiff Univ., Cardiff, CF10 3AT, UK)

Models of binaural unmasking suggest that the auditory system detects deviations from unity in the interaural coherence of the waveforms at the two ears. Since anticorrelated noise has high coherence within a frequency channel, such a coherence-detection mechanism should be unable to distinguish a correlation of -1.0 from a high positive correlation. A 3I-2AFC paradigm was used to test whether such a limitation can be observed behaviorally. Stimuli consisted of a 1-ERB-wide sub-band of interaurally correlated noise centered at 500 Hz (i.e. 462–538 Hz) flanked by two spectrally remote bands of noise (0–329 Hz and 717–3000 Hz) with an interaural correlation of 1.0. The flanking bands were intended to reduce cues from the perceived laterality of the stimuli. Participants were presented with three stimuli (reference, target, and distracter) and asked to choose the odd-one-out: the reference and distracter intervals contained sub-bands of anticorrelated noise while the target interval contained a sub-band of noise with interaural correlation between 0.7 and 1.0. Preliminary results indicate that different participants find different values of interaural correlation to be more distinguishable than others.

4pPP15. A biologically inspired binaural approach to monaural modeling. Daniel E. Shub and H. Steven Colburn (Boston Univ. Hearing Res. Ctr., 44 Cummington St., Boston, MA 02215)

The auditory system is often discussed as having monaural and binaural neurological pathways; similarly models are classified as either monaural or binaural. Psychophysical evidence of contra-aural interference (when performance with one ear is better than performance with two ears) suggests that the information used on monaural tasks (e.g., N_0S_0 and N_mS_m detection) may be carried by a binaural pathway. Binaural models often require monaural channels to predict the results of monaural tasks, but these monaural channels prevent the models from predicting contra-aural interference. This modeling work investigates the monaural information carried by a processor which is inherently binaural. The processor design makes the inclusion of monaural channels unnecessary and contra-

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aural interference is predicted under certain conditions. The performance of the model matches results from a variety of traditional psychophysical tasks (including discrimination of differences in overall intensity; discrimination of differences in interaural level, time and coherence; as well as detection under monaural and binaural masking conditions). Results suggest that binaural neurons contain sufficient information to explain performance on both binaural and monaural tasks. [Work supported by NIH grants R01 DC 00100 and 1 F31 DC006769-01.]

4pPP16. Auditory nerve response to broadband noise with high-frequency spectral notches. Ana Alves-Pinto, Enrique A. Lopez-Poveda (Instituto de Neurociencias de Castilla y Leon, Univ. of Salamanca, Avda. Alfonso X El Sabio s/n, 37007 Salamanca, Spain), and Alan R. Palmer (MRC Inst. of Hearing Res., Nottingham NG7 2RD, UK)

The threshold notch depth for discriminating between a flat-spectrum broadband noise and a similar noise with a rectangular spectral notch centered at 8 kHz varies nonmonotonically with stimulus level (Alves-Pinto and Lopez-Poveda, submitted to J. Acoust. Soc. Am.). A possible explanation for this result is that the notch may be encoded in the rate profile of auditory nerve (AN) fibers with high spontaneous rate (HSR) at low levels and in that of low spontaneous rate (LSR) fibers at high levels. To test this hypothesis, the rate profile of guinea pig AN fibers was measured in response to broadband notched noise for different notch depths and widths and for overall levels ranging from 40 to 100 dB SPL. Preliminary results support the hypothesis that HSR fibers can encode the spectral notch only for levels up to around 70 dB SPL. However, they also suggest that, as in the psychophysics, the negative effect of level is less pronounced the wider the notch. Additional data are required to confirm the role of LSR fibers to encode for the notch at higher levels. [Work supported by Spanish FIS PI020343 and G03/203.]

4pPP17. The spatial distribution of sound pressure within the human ear canal. Michael R. Stinson and Gilles A. Daigle (Inst. for Microstructural Sci., Natl. Res. Council, Ottawa, ON, Canada K1A 0R6, mike.stinson@nrc-cnrc.gc.ca)

The sound field inside a human ear canal has been computed using 2 approaches. A simple model, a modified Webster horn equation approach, can accommodate the curvature and varying cross section of the ear canal. Calculations using the horn equation demonstrate the formation of standing waves within the canal. The pressure may be interpreted as either the pressure along the center axis of the canal or the average pressure within a cross-sectional slab. To investigate possible spatial variation through a cross section, the sound field has also been computed using the boundary element method (BEM). Over 2000 triangular mesh elements, 1 mm or less in size, were used to represent the canal geometry. For a plane piston source at the canal entrance and both a rigid and a resistive impedance condition at the eardrum position, the computed sound pressures along the center axis of the ear canal are in good agreement with the horn equation calculations, up to 15 kHz. The BEM approach, though, reveals spatial variations of sound pressure through each canal cross section, increasingly significant as frequency increases. Further, for source configurations that are more realistic than a simple piston, large transverse variations in sound pressure are anticipated.

4pPP18. Distortion product otoacoustic emissions generated by mistuned harmonic stimuli. Glenis R. Long and Jungmee Lee (Speech and Hearing Program, Grad. Ctr. CUNY, 365 Fifth Ave, New York, NY 10016)

Psychoacoustic research in humans [Lee and Green, J. Acoust. Soc. Am. **95**, 718–725 (1994)] suggests that the detection of the mistuning of an harmonic when the harmonic complex and a mistuned component were presented simultaneously to the same ear stems, at least in part, from the resulting envelope interactions on the basilar membrane. Neurophysiological research in chinchillas [Sinex *et al.*, Hear. Res. **168**, 150–162 (2002)], provides evidence that such envelope interactions can be detected at the level of the inferior colliculus. The potential role of cochlear nonlinearites in determining the nature of the signal on the basilar membrane is explored by evaluating the DPOAE generated by multicomponent harmonic and inharmonic complexes. When harmonic complexes are used, the DPOAE all fall at harmonic frequencies. When inharmonic complexes are used, many nonharmonic DPOAE are detected.

4pPP19. The reflectivity function of the cochlear transmission line. Renata Sisto (ISPESL–DIL, Via di Fontana Candida, 1, 00040 Monteporzio Catone (RM), Italy) and Arturo Moleti (Univ. of Roma Tor Vergata, 1, 00133 Roma, Italy)

The generation of evoked otoacoustic emissions is related, for each frequency, to the reflectivity function of the cochlear membrane near the correspondent resonant place. A formal analogy exists between the transmission line equation that is often used to model the propagation of sound along the cochlea and the Schroedinger equation describing the motion of an elementary particle in a one-dimensional potential. In this analogy, the cochlear region near the tonotopic place is equivalent to a negative potential well. Analytical solutions are available in quantum mechanics textbooks, which predict partial reflection of the incoming wave from such a potential well. The reflectivity is expected to be a slow quasi-periodic function of the well width and depth, e.g., of the quality factor of the cochlear resonance. This observation could be useful to predict the large-scale spectral structure of evoked otoacoustic emissions as a function of the cochlear tuning curve.

4pPP20. Multiple auditory steady-state response thresholds to bone-conduction stimuli using three oscillator placements in premature infants. Jennifer Hatton, Susan A. Small, and David R. Stapells (School of Audiol. & Speech Sci., Univ. of British Columbia, 5804 Fairview Ave, Vancouver, BC, Canada V6T 1Z3, stapells@audiospeech.ubc.ca)

Auditory Steady State Responses (ASSRs) are a promising technique for estimating the behavioral audiogram in infants. The current study investigates the effects of bone-oscillator placement on ASSR thresholds in premature infants ($N=15$, mean age: 35 weeks PCA). Using the MASTER research system, ASSR thresholds to multiple bone-conduction tones (0–50 dBHL re:mastoid; 77–101 Hz) presented to the (i) superior-posterior temporal bone, (ii) mastoid, and (iii) forehead were obtained. Results with no response at 50 dBHL were arbitrarily assigned a threshold of 60 dBHL. No differences in threshold were found between the temporal and mastoid placements. Thresholds averaged across these placements were 17, 15, 34, and 30 dBHL for 500, 1000, 2000, and 4000 Hz, respectively. However, on average, thresholds for the forehead placement were significantly higher by at least 14, 11, 18, and 14 dB at 500, 1000, 2000, and 4000 Hz, respectively, compared to the temporal and mastoid placements. These differences may be even greater because absent responses at the maximum intensity were seen more often with forehead placement compared to other placements. In conclusion, one may use either the temporal bone or mastoid for bone-oscillator placement, but the forehead should be avoided. [Work supported by CIHR and NSERC-Canada.]

4pPP21. Multiple auditory steady-state response (ASSR) thresholds to bone-conduction stimuli in young infants with normal hearing. Susan A. Small, Jennifer Hatton, and David R. Stapells (School of Audiol. & Speech Sci., Univ. of BC, 5804 Fairview Ave., Vancouver, BC, Canada V6T 1Z3)

Bone-conduction ASSR thresholds (carrier frequencies: 500–4000 Hz; 77–101-Hz modulation rates; amplitude/frequency modulated; single-polarity stimulus) were obtained in two infant groups ($N=29$ pre-term, tested in NICU; $N=14$ 0-to-8 months of age, tested in soundbooth). Mean (1SD) ASSR thresholds were 16 (11), 16 (10), 37 (10), and 33 (13) dBHL in pre-term infants and 14 (13), 2 (7), 26 (6), and 22 (8) dBHL in older infants at 500, 1000, 2000, and 4000 Hz, respectively. Both infant groups had significantly better thresholds for 500 and 1000 Hz compared to 2000 and 4000 Hz, in contrast to adults who have similar thresholds across frequency (22, 26, 18, and 18 dBHL) [Small and Stapells, J. Am. Acad. Audiol (in press)]. When 500- and 1000-Hz thresholds were pooled, NICU infants and 0–8-month-old infants tended to have better low-frequency thresholds than adults. When 2000- and 4000-Hz thresholds were pooled, NICU and 0–8 month-old infants tended to have poorer thresholds than adults. Overall, these results suggest that low-frequency bone-conduction thresholds worsen and high-frequency bone-conduction thresholds improve with maturation. [Work supported by CIHR and NSERC Canada.]

4pPP22. Human electrophysiological examination of the buildup of the precedence effect (Clifton effect). Andrew Dimitrijevic and David R. Stapells (School of Audiol. & Speech Sci., Univ. of British Columbia, 5804 Fairview Ave., Vancouver, BC, Canada V6T 1Z3, andrew@audiospeech.ubc.ca)

The relationship between behavioral measures of buildup of precedence effect [or Clifton effect (CE)] and electrophysiological responses using an event-related potential (ERP) paradigm was examined in 14 young adults with normal hearing. The CE was elicited using a binaural paired-click (left and right speaker delay) in sound field. This study aimed to determine whether ERP measures were related to the perception of buildup. Subjects participated in two sessions: (i) Session #1 psychoacoustic measures of the degree of buildup as function of click delay (left and right side). (ii) Session #2 recorded cortical ERPs using click delays (varying from 6 to 8 ms) that would maximize the degree of buildup in each subject. Stimuli consisted of repeated trains containing 5 paired-clicks (left speaker leading) with a noise burst separating consecutive trains. Results: Significant $N1$ amplitude hemispheric asymmetries were observed related to click position. $N1$ responses decreased (43%) as a function click position in the train at electrode sites contralateral to the leading stimulus and, conversely, increased (30%) at electrode sites ipsilateral to the lead. These results suggest that differential refractoriness of the $N1$ response may lead to the lateralized perception of buildup to the precedence effect. [Work supported by NSERC-Canada.]

4pPP23. The event-related brain potentials to uncorrelated fragment of noise. Juan Huang, Xihong Wu, Liang Li (Dept. of Psych., Peking Univ., Beijing 100871, China, jacee@pku.edu.cn), James Qi, Bruce Scheneider (Univ. of Toronto at Mississauga, Mississauga, ON, Canada), Yu He, and Claude Alain (Baycrest Ctr. for Geriatric Care, Toronto, ON, Canada)

Two correlated waveforms of broadband noise presented to the left and right ear simultaneously or with a short delay are often perceptually fused into one sound image. However, when an uncorrelated noise fragment (UCNF) is inserted in a long duration sound, listeners report hearing a transient burst of noise. The detection of the UCNF is dependent on the duration of the UCNF and the binaural delays. Here, we recorded event-related brain potentials (ERPs) to 100-ms UCFN and varied the binaural delay (0, 3, 10, and 20 ms) from trial to trial. The likelihood of detecting the UCNF decreased with increasing binaural delay. At 0 delay, the UCNF elicited negative and positive waves peaking at about 100 and 200 ms after UCNF onset (N1-P2 complex), which was present even when the stimuli were ignored. The conscious detection of the UCNF elicited an additional positive wave between 250 and 450 ms at parietal and occipital sites (P3b). The P3b latency was longer and its amplitude larger for binaural delay of 3 than 0 ms. Our results show that the detection of UCF involved both automatic and attention-dependent processes especially when a binaural delay is introduced between two source of correlated noises.

4pPP24. Tinnitus neural map; A positron emission tomography study. Martin L. Lenhardt (Dept. of Biomed. Eng., Otolaryngol., Emergency Medicine, Virginia Commonwealth Univ., Richmond, VA 23298), Abraham Shulman, and Barbara Goldstein (SUNY, Brooklyn, NY)

The neural map of tinnitus involves more than the classical auditory pathways and the limbic system. PET findings in six patients, with severe problem tinnitus, revealed the involvement of the cerebellum, insula and frontal cortices when these patients were imaged before and after high frequency bone conduction therapy. The frontal cortex and cerebellum demonstrated the highest ratios of metabolic change but changes were also noted in the thalamus and the medial temporal lobe system. The PET data supports the view that the frequency specific map of auditory cortex is dynamic and can change with high frequency therapy, presumably due to neural reorganization. The PET data further reflect alterations in multiple areas of brain in all patients to tinnitus and/or high frequency therapy. Encouraging was the observation that patients with the most dramatic change in their global pre/post PETs were also the ones who exhibited the greatest behavioral improvement in regards to tinnitus relief measured by informal report and standard questionnaires. In those patients, post therapy minimal masking levels dropped indicating less acoustic energy was need to mask their problem tinnitus, interpreted as reflecting central changes. Taken as a whole, these data support the value of frequency specific tinnitus therapy.

4pPP25. Real-time implementation of a model auditory-nerve fiber. Satish G. Iyengar (Inst. for Sensory Res. and Dept. of Elec. Eng. and Comput. Sci., Syracuse Univ., Syracuse, NY 13244), Jayant Datta (Syracuse Univ. and Discrete Labs., Inc., Syracuse, NY 13220), and Laurel H. Carney (Syracuse Univ., Syracuse, NY 13244)

A time-varying model auditory-nerve (AN) fiber based on Tan and Carney (2003) was developed on a digital signal processor (DSP) platform to simulate AN responses in real-time, using the Motorola DSP56367, a 150 MIPS audio processor. To understand the schemes employed by the brain for decoding different sound stimulus parameters, it is first essential to get an accurate representation of peripheral (external, middle, and inner ear) responses that in turn excite the higher levels along the auditory pathway. Though present computational models predict essential AN response properties, their use in testing mechanisms hypothesized to occur at higher levels of the auditory pathway is limited due to the large processing times that they require. This is because such nonlinear models require computational implementation in the time domain, with time-step-by-time-step changes in model parameters as the bandwidth and gain of the inner ear change with stimulus fluctuations. The DSP-based model presented here aims to speed up inquiries related to auditory signal processing strategies at higher levels by providing a real-time solution to AN fiber simulations. Details of the implementation and some prospective auditory-research utilities will be discussed. [Work supported by NIDCD R43-DC006591 (JD, SGI) and R01-DC001641 (LHC, SGI).]

4pPP26. Modeling inner hair cell compression. Almudena Eustaquio-Martin and Enrique A. Lopez-Poveda (Instituto de Neurociencias de Castilla y Leon, Universidad de Salamanca, Avda. Alfonso X El Sabio s/n, 37007 Salamanca, Spain)

The voltage-dependent activation of inner hair cell (IHC) basolateral potassium ($K+$) currents results in a compression of voltage responses to injections of large depolarizing currents with values well within the range of IHC transducer currents (Kros and Crawford, J. Physiol. **421**, 263–291 (1990)). *In vivo*, this compression must add to the compressive nonlinearity produced by the gating of transducer channels. The purpose of this work is to present a biophysical model of the IHC that simulates the nonlinear compressive effects of IHC basolateral potassium currents both *in vivo* and *in vitro*. The model incorporates two components for the potassium currents, one fast and one slow, as described by Kros and Crawford. Membrane parameters are provided for the model to reproduce the time course of the receptor potential in response to constant current pulses across the membrane. The model is then used to investigate the degree of compression that the K+ currents contribute to the receptor potential *in vivo*. Also investigated are the effects on IHC compression of a partial blocking of the K+ channels, as occurs in some common forms genetic hearing loss. [Work supported by Spanish FIS PI020343 and G03/203.]

THURSDAY AFTERNOON, 19 MAY 2005 PLAZA B, 2:00 TO 4:45 P.M.

Session 4pSA

Structural Acoustics and Vibration and Musical Acoustics: Vibration of and Acoustic Radiation from Musical Instruments II

Courtney B. Burroughs, Cochair
Applied Research Lab., Pennsylvania State Univ., State College, PA 16804-0030

Thomas D. Rossing, Cochair
Physics Dept., Northern Illinois Univ., De Kalb, IL 60115

Invited Papers

2:00

4pSA1. Piano soundboards: Experimental and modeling studies. Nicholas Giordano (Dept. of Phys., Purdue Univ., West Lafayette, IN 47907, giordano@purdue.edu)

The piano has attracted the attention of many researchers, and most of that work has focused on piano hammers and string vibrations. The vibrational and radiative properties of the soundboard have drawn much less attention, but are essential for an understanding of the sound produced by the instrument. We review experimental studies of the soundboard's vibrational modes and their radiative efficiency. We also describe modeling studies of soundboard vibrations and radiation, which aid in understanding the measurements. Prospects for future work and for applications to other instruments are suggested. [Work supported by NSF grant PHY-9988562.]

2:30

4pSA2. Phase mapping and sound radiation from Caribbean steelpans. Andrew Morrison (Dept. of Phys., Northern Illinois Univ., DeKalb, IL 60115)

The Caribbean steelpan is one of the most interesting acoustic musical instruments invented in the last century. Although simple in design, the acoustic properties of the steelpan are surprisingly complicated. Holographic interferometry was used to determine the resonances of a low tenor steelpan. Placement of a vibrating mirror in the optical path of the reference beam expands the capabilities of the holography system to include phase measurements. Phase maps of several low resonances of notes on a Caribbean steelpan will be shown. Sound intensity measurements were taken to explore the relationship between the resonances and the radiated sound field. The pan was placed in an anechoic chamber, and selected notes were excited electromagnetically with a swept sinusoid signal. A two-microphone probe was used to gather sound intensity measurements. Sound intensity maps of the first three harmonics will be shown for several notes.

3:00

4pSA3. Acoustics of Baltic psaltery; a new outstanding Latvian Kokle. Andres Peekna (Innovative Mech., Inc., 5908 N River Bay Rd., Waterford, WI 53185) and Thomas D. Rossing (Northern Illinois Univ., DeKalb, IL 60115)

We have studied the acoustics of a Latvian Baltic Psaltery (kokle) which was judged by performing and recording musicians to be outstanding. Previous work by the authors pointed out the importance of a high population of body-resonances within the tuning range of the instrument, with good string-to-resonance coupling also playing an important role. This particular kokle shows outstanding coupling of strings to body-resonances, with all strings evidencing some degree of coupling. In some cases, the strings couple to a superposition of more than one body-resonance.

3:30–3:45 Break

3:45

4pSA4. Modal analysis and computer aided design studies of acoustic guitar structural designs. Craig M. Rashkow, Robert D. Collier (Thayer School of Eng., Dartmouth College, Hanover, NH 03755), and Alan Carruth (Luthier, Newport, NH 03773)

This study seeks to systematically identify modern Computer Aided Design (CAD) techniques to optimize acoustic guitar design. The study uses traditional design as a starting point for analysis of variations in materials and bracing structures of the front sound plate. CAD techniques are used to model the guitar and provide a broadband input that allows analysis of its modal response. Design optimization occurs in creating the most favorable modal response of the instrument while still maintaining structural integrity through effective support of the forces created by the strings' tension in the bridge and neck. The measures chosen to help identify the most favorable response are modal density, magnitude of the response at each mode, and the locations of the largest magnitudes. Increased modal density provides a richer timbre while the amplitude of the response determines the relative intensity of the sound projected. Increased modal response in the upper bout of the instrument can improve the timbre of pitches in the upper frequency range of the instrument. After analyzing these measures for multiple variations and iterations of a selected set of structural parameters, an optimal design is suggested. The results show good agreement between experimental modal measurements and computer-aided design modeling.

4:00

4pSA5. Applying vibration and acoustical radiation from musical instruments in a recording studio setting. Pamela J. Harght (Berklee College of Music, 1140 Boylston St., Boston, MA 02115)

Radiation patterns affect the musical characteristics of any instrument. Applying the radiation characteristics of instruments to microphone technique will enable musicians, audio engineers and students to capture the most-desirable sound, and understand how to position microphones efficiently and appropriately. This paper will use the research done on several instruments' acoustical characteristics and apply it to microphone techniques in a professional-level recording studio at Berklee College of Music. Audio examples will be used to demonstrate microphone placement according to the acoustical radiation of musical instruments. [Thank-you to K. Anthony Hoover for his support and encouragment with this paper.]

4:15

4pSA6. Achieving pseudo-degeneracy in handbell modes. John R. Buschert, Sungdo Cha, Daniel A. King, and Daniel B. Horst (Goshen College, 1700 S. Main, Goshen, IN 46526)

Degenerate modes in handbells are split by nonuniformity in the bell. By adding small masses to the bell, the modes can be shifted in position and frequency. Interesting things happen when one attempts to recreate a degeneracy that has been split. Holographic photos of the modes will be shown which can be used to follow the effects of the added mass on individual modes. By moving and varying the mass, one can bring the modes back to a pseudo-degeneracy. Holographic interferometry photos and graphs of the position and frequency of the modes show the stages of this transition.

4:30

4pSA7. Vibration study of Indian Gong hung at one point near the edge. Paresh Shravage, S. Parmeswaran, and Keith deSa (Acoust. Res. Lab, Dept of Phys., N. Wadia College, Pune, India)

Indian Gong is made up of brass and circular in nature. It is Percussion Instrument. It is used in some religious processions and music concerts in India. It is hung by a metal wire through a hole near the edge of circumference. It is vibrated by a metal (iron) hammer after repeated intervals to get sound of same pitch. This paper relates to vibrational analysis of Indian Gong. A study has been done to check the vibrational properties and the modes of vibration of the Gong. The study is done by spectrum analysis (FFT) and Time-average holography as a vibration analysis tool. The study will help to analyze certain vibrational features of brass. The tonal quality of brass is also compared to other metals, so that it will be useful in making musical instruments. The analysis of hologram may yield some interesting properties of the plate. In conclusion the paper is going to deal with modes of vibration of Indian Gong. The vibrational properties will depend on the point of actuations. The paper describes the mode of vibration and its mathematical equation.

Session 4pSC

Speech Communication: Speech Production and Perception I (Poster Session)

Abeer Alwan, Chair
Dept. of Electrical Engineering, UCLA, 405 Hilgard, Los Angeles, CA 90095

Contributed Papers

All posters will be on display from 1:30 p.m. to 5:00 p.m. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 1:30 p.m. to 3:15 p.m. and contributors of even-numbered papers will be at their posters from 3:15 p.m. to 5:00 p.m.

4pSC1. Tuning phenomena of melodies and resonance frequencies (formants) in infants pre-speech utterances. Kathleen Wermke (Univ. Wuerzburg, Pre-speech Ctr., Dept. of Orthodontics, Pleicherwall 2, 97070 Wuerzburg, Germany, wermke_k@klinik.uni-wuerzburg.de), Werner Mende (Berlin-Brandenburg Acad. of Sci., Berlin, Germany), Claudia Manfredi, Pierro Bruscaglioni (Univ. of Firenze, Italy), and Angelika Stellzig-Eisenhauer (Julius-Maximilians-Univ., Wuerzburg, Germany)

In a former study [Wermke *et al.*, Med. Eng. Phys. **24**, 501–514 (2002)], an increasing tuning between laryngeal (melody) and pharyngeal (resonance frequencies) activity was demonstrated during pre-speech development. This tuning was observed unexpectedly early during development in mitigated cries and earliest non-cry utterances and prepares probably articulation in speech-like vocalizations of older infants. A new retrospective study supported this assumption by comparing tuning processes in cries and early non-cry utterances ($N=2500$) in two groups (low versus high word production performance at 18 months) of term-born healthy infants ($N=20$). Additionally, age-matched comparisons were made in 4 cleft-lip-palate-infants treated with a palatal plate. In order to demonstrate the interaction between melody and resonance control we designed a special graphical representation. The tracking function of the resonance frequencies is displayed synchronously to the melody and its harmonics. Resonance frequencies in pre-speech utterances are not yet identical to formant bands associated with speech sounds. The results support the existence of an early active tuning and its relation to later speech. This behavior seems to prepare formant tuning in later speech. Medical applications are seen for infants with disturbances of the vocal tract transfer function, e.g., infants with cleft-lip-palate. [Work supported by DFG.]

4pSC2. The gradient influence of talker sex and perceived sexual orientation on fricative perception. Benjamin Munson, Sarah V. Jefferson, and Elizabeth C. McDonald (Dept. Speech-Lang.-Hearing Sci., Univ. of Minnesota, 115 Shevlin Hall, 164 Pillsbury Dr., SE, Minneapolis, MN 55455, Munso005@umn.edu)

Listeners are more likely to hear a synthetic fricative ambiguous between /s/ and /ʃ/ as /ʃ/ if it is appended to a woman's voice than a man's voice [Strand, J. Lang. Soc. Psych. **18**, 86–99 (1999)], suggesting that speech perception is sensitive to social-indexical information. This study examined the influence of two variables on listeners' fricative perception: (a) talker sex, and (b) talkers' perceived sexual orientation (PSO, i.e., the probability that a talker is identified as gay, lesbian, or bisexual (GLB) based on speech alone). Stimuli were created by pairing a synthetic nine-step /s/-to-/ʃ/ series with tokens of /æk/ and /ɪp/ taken from natural productions of *shack* and *ship* by 44 talkers (22 women, 22 men), for a total of 88 different continua. Forty listeners rated the 44 talkers' PSO. A different group of 10 listeners participated in a series of two-alternative *sack-shack* and *sip-ship* identification experiments. As expected, listeners identified more /ʃ/ tokens for women's voices than for men's. GLB-sounding women elicited significantly fewer /ʃ/ percepts than heterosexual-sounding women. No consistent influence of PSO on fricative identification was noted for men's voices. Regression analyses showed strong relationships between fricative identification and ratings of PSO for women talkers only.

4pSC3. Recognition of English phonemes in noise. José R. Benkí (Kresge Hearing Res. Inst., Dept. of Otolaryngol., Univ. of Michigan Med. School, Ann Arbor, MI 48109-1346, benki@umich.edu) and Robert Felty (Univ. of Michigan, Ann Arbor, MI 48109-1285)

Native speakers of American English identified the consonants and vowels of American English CV and VC syllables and the vowels of hVd syllables. The CV and VC syllables, consisting of all of the English consonant phonemes, were selected from recordings described by Shannon *et al.* [J. Acoust. Soc. Am. **106**, L71–L74 (1999)], and the hVd syllables were selected from recordings described by Hillenbrand *et al.* [J. Acoust. Soc. Am. **97**, 3623–3641 (1995)]. All syllables were presented binaurally in signal-correlated noise. The resulting confusion matrices and feature analyses will be presented along with a position (initial versus final) analysis for the consonant results. The consonant and vowel results will be used to derive empirical measures of similarity of English words. [Work supported by NIH/NIDCD.]

4pSC4. Perturbed palatal shape and North American English /r/ production. Mark K. Tiede (Haskins Labs, 270 Crown St., New Haven, CT 06511, tiede@haskins.yale.edu and M.I.T.-R.L.E., Cambridge, MA), Vincent L. Gracco, Douglas M. Shiller (McGill U., Montreal, QC, Canada), Carol Espy-Wilson (U. of Maryland, College Park, MD), and Suzanne E. Boyce (U. of Cincinnati, Cincinnati, OH)

It is well established that the lowered $F3$ associated with the acoustic percept of American English /r/ can be achieved with different tongue shapes in production. Broadly speaking these shapes may be grouped into "bunched" and "retroflex" varieties. In this work the effects of somatosensory perturbation on /r/ production are examined. Subjects were fitted with a custom palatal prosthesis incorporating a 0.5 cm protrusion along the alveolar ridge, and tongue position during production of /r/ in vocalic contexts was observed using EMA under four conditions: before prosthesis placement; while wearing the prosthesis immediately following placement; still wearing the prosthesis following an unrecorded 20 min adaptation period; and immediately after prosthesis removal. Acoustic effects of the perturbation were minimal, especially after adaptation; production effects were most pronounced in the low vowel context. One subject showed an unperturbed preference for a retroflex configuration, but increased the degree of retroflexion with the palatal prosthesis in place. The remaining subjects preferred unperturbed bunched shapes, but under the perturbed conditions produced primarily retroflex configurations, which for one subject persisted after prosthesis removal. These results suggest that speaker preference for one shape over another may be determined by palatal morphology. [Work supported by NIH.]

4pSC5. Relational acoustic invariance in the single/geminate stop distinction in Japanese.

Yukari Hirata and Jacob Whiton (Dept. of EALL, Colgate Univ., 13 Oak Dr., Hamilton, NY 13346)

This study addressed an issue in the theory of relational acoustic invariance [Pickett *et al.*, Phonetica **56**, 135–157 (1999)]. The question was whether an invariant acoustic property exists for distinguishing Japanese single and geminate voiceless stops across different speaking rates. Four native Japanese speakers produced disyllabic words with single and geminate voiceless stops spoken in a carrier sentence at three speaking rates. Durations of sentences, words, stop closures, and vowels preceding the contrasting stops were measured. Ratios of geminate to single stop closures, geminate words to single words, closures to preceding vowels, and closures to words were calculated. The stop closure duration significantly overlapped between the single and geminate categories across rates. However, the ratio of geminate to single closure duration was unaffected by rate. Among the measures examined, the ratio of closure to word duration (0.35 as an optimal boundary) best classified all single and geminate tokens with 95.7–98% accuracy. These results suggest that, in spite of overlap in absolute closure duration between single and geminate stops, there is a relationally invariant measure that divides the two phonemic categories across rates and speakers, supporting the theory of relational acoustic invariance.

4pSC6. Articulatory influences on the categorization of speech sounds.

H. Henny Yeung (Dept. of Psych., Univ. of British Columbia, 2136 West Mall, Vancouver, BC, Canada V6T 1Z4, hhyeung@psych.ubc.ca), Bryan W. Gick (Univ. of British Columbia, Vancouver, BC, Canada V6T 1Z4), and Janet F. Werker (Univ. of British Columbia, Vancouver, BC, Canada V6T 1Z4)

This cross-modal study investigates whether production of speech gestures can influence auditory speech perception. Participants made categorization judgments on a continuum between /ba/ and /da/, while making concurrent motor gestures. Ten participants categorized synthesized sounds from a /ba/-/da/ continuum while (a) simply listening, (b) concurrently making a /ba/, /da/, or /ga/ gesture without vocal fold vibration, or (c) making a nonspeech gesture (i.e., tongue protrusion). Preliminary data indicate that making /ba/ and /ga/ gestures, or tongue protrusions, will shift the perceptual boundary away from the /da/ end of the continuum (i.e., towards the /ba/ end of the continuum), compared to a passive listening condition. Making /da/ gestures did not shift the perceptual boundary, compared to this same listening condition. These data offer support for the idea that auditory categorization of speech tokens on a continuum can be influenced by conflicting information in other modalities—specifically articulatory gestures. Further studies will be conducted to examine the influence of (a) other speech gestures on these synthesized sounds, and (b) speech gestures on the perception of naturally-produced speech tokens.

4pSC7. Does intonation have primitive units?

Amebu K. Seddoh (Dept. of Commun. Sci. & Disord., Univ. of North Dakota, P.O. Box 8040, Grand Forks, ND 58202)

This study investigated whether listeners can interpret intonation based on partial rather than global fundamental frequency ($F0$) or pitch information. Matched Statements and echo questions were recorded digitally at 22 kHz sampling rate using a Computerized Speech Lab (CSL). Each production was stored on the CSL and edited by slicing the wave form into two parts that corresponded to terminal (last 150–200 ms span of the $F0$ contour) and preterminal (section of contour preceding the terminal region) $F0$ contour regions. A total of 92 stimuli was thus generated and recorded on a DAT tape. The tape was played to 96 young (20–28 years) and old (42–79 years) adults to determine if each stimulus was part of a statement or a question. For preterminal (PTL) stimuli, both groups successfully identified statements 91% and 88% of the time, respectively, compared to 72% each for questions. The respective scores for the terminal (TL) stimuli were 94% and 91% (statements) versus 91% and 87% (questions). Across groups, statements were easier to identify with both TL ($p=0.002$) and PTL ($p=0.001$) cues compared to questions. These findings suggest that intonation decoding might involve units smaller than global $F0$ contour. [Work supported by NIH (NIDCD) grant 5R03DC04955-02.]

4pSC8. Differential effects of speaking rate and phonemic vowel length on formant frequencies of Japanese vowels.

Kimiko Tsukada (MARCS Auditory Labs., Univ. of Western Sydney, Penrith South DC NSW 1797 Australia) and Yukari Hirata (Colgate Univ., Hamilton, NY 13346)

This study examined the extent to which variation in speaking rate and phonemic vowel length affect the first two formant frequencies of Japanese vowels. If vowel duration is a factor that determines the degree of formant undershoot [Moon and Lindblom, J. Acout. Soc. Am. **96**, 40–55 (1994)], formants of phonemic long vowels or vowels spoken at slower rates would occupy more peripheral areas in the $F1/F2$ vowel space than the short counterparts. Four male native speakers produced Japanese disyllabic non-words, /mVmV/, /mVVmV/, and /mVmVV/ (V=/i e a o u/, VV=/i: e: a: o: u:/) spoken in a carrier sentence at slow, normal, and fast rates. Effects of vowel length were clear for all five vowels: the long vowels occupied more peripheral areas of the vowel space than the short vowels. However, such a systematic effect was not found with speaking rate changes. ANOVAs performed on each vowel type indicated that, for non-high vowels /e(:) a(:) o(:)/, effects of vowel length and speaking rate interacted. The $F2$ of /e:/, for example, did not differ across rates, but the $F2$ of /e/ was significantly lower for faster rates. Implications are discussed in terms of Moon & Lindbloms formant undershoot model.

4pSC9. Enhancement of visual perception of speech via tactile input.

Diana Gibraiel, Bryan Gick, Yoko Ikegami, Kristin Johannsdottir (UBC, Dept. of Linguist., E-270, 1866 Main Mall, Vancouver, BC, Canada V6T 1Z1, gibraiel@interchange.ubc.ca), and Jeff Muehlbauer (UBC, Vancouver, BC, Canada V6T 1Z1)

Motor theories of speech perception predict that perceptual information from modalities other than sound enhance speech perception directly by informing the perceiver of the speaker's gestures [A. Lieberman and I. Mattingly, Cognition **21**, 1–36 (1985)]. Acoustic theories predict that perceptual information from modalities other than sound will only enhance speech perception if there is a learned mapping between the acoustic speech signal speech and that modality [R. Diehl and K. Kluender, Ecol. Psych. **1**, 121–144 (1989)]. As normal subjects are unlikely to have learned a mapping between visual and tactile speech information, this study tests whether and how tactile input enhances visual speech perception. In the control condition, perceivers in noise repeat syllables pronounced by a speaker who they can see clearly. Accuracy is judged on the basis of the repeated syllables. In the experimental condition, subjects additionally have their hand on the speaker's face in the Tadoma position. Results show that subjects are significantly more accurate at perceiving speech when they have both visual and tactile input then when they have visual input alone. In particular, tactile input enhances perceptual accuracy of voice and manner features. [Work supported by NSERC.]

4pSC10. A model of invariant patterns of articulatory movements. Patrizia Bonaventura (Dept. of Linguist., Univ. of British Columbia, 1866 Main Mall, Buchanan E270, Vancouver, BC, Canada V6T 1Z1, bonaventura.8@osu.edu)

The goal of the present study is to model the ''iceberg'' portions of the demisyllables, previously extracted from the microbeam articulatory data (Bonaventura, 2003), by curve fitting. The polynomial analysis was designed to include an appropriate weighting window centering around the threshold crossing point, and aimed to provide an estimate of how, in the vicinity of the crossing point, the curve deviates from a straight line: this deviation would be represented by the higher order coefficients of the polynomial. The model was obtained preliminarily on the basis of 100 curves for the lower lip movement for /f/ and /v/ (in initial and final demisyllable for ''five''), and from 100 curves for the tongue tip displacement (for /n/ in ''nine''). In order to fit the data to the model, a robust least square method (Least Absolute Residuals) has been used, in order to minimize the influence of the outliers, that are present in the read speech data, and cannot be accounted for by ''phrase final lengthening effects.'' The fit results for the cubic polynomials satisfactorily approximated the ''iceberg'' curves. The 95% confidence bounds on the fitted coefficients indicated that they were acceptably accurate.

4pSC11. The effects of auditory-visual vowel and consonant training on speechreading performance. Carolyn Richie and Diane Kewley-Port (Speech and Hearing Sci., Indiana Univ., Bloomington, IN 47405, crichie@butler.edu)

Recent work examined the effects of a novel approach to speechreading training using vowels, for normal-hearing listeners tested in masking noise [C. Richie and D. Kewley-Port, J. Acoust. Soc. Am. **114**, 2337 (2003)]. That study showed significant improvements in sentence-level speechreading for listeners trained on vowels compared to untrained listeners. The present study examined the effects of combining vowel and consonant training on speechreading abilities. Normal-hearing adults were tested in auditory-visual conditions in noise designed to simulate a hearing loss. Using a monosyllable context, one group of listeners received training on consonants, and another group received training on consonants and vowels combined. A control group did not receive training. All listeners performed speechreading pre- and post-tests, on words and sentences. Comparison with the earlier study showed posttest sentence-level speechreading increased by 10 percentage points for participants in the vowel training program, 8 percentage points for participants in the consonant training program, and, unexpectedly, only 2 percentage points for participants in the combined training program. Results from these relatively short training programs suggest that vowels, previously unused in speechreading training, may provide gains in speechreading abilities and play an important role in rehabilitation of hearing-impaired persons. [Work supported by NIHDCD02229.]

4pSC12. Turning speech into music in a two-dimensional space by varying the bandwidth and rate of tone pulses placed along formant tracks. C. J. Darwin (Dept. of Psych., Univ. of Sussex, Brighton, BN1 9QG, UK)

A two-dimensional space is introduced which changes sounds from speech-like to music-like by varying the rate and the bandwidth (decay-time) of individual tones placed along formant contours. At one extreme of a very fast pulse rate and a long decay time, the sounds are equivalent to sine-wave speech. With a more moderate tone rate and a short decay time, the sounds become monotone (at the tone-rate) speech. For tone rates around 10/s with long decay times, individual musical tones are heard at the formant frequencies. Intelligibility is poor for such sounds because of the low sampling rate of the formant contours. Increasing the tone rate and decreasing tone decay time increases intelligibility and makes the sounds less music like. Intelligibility data together with speech-like and music-like ratings of sounds from this space will be presented.

4pSC13. Variability of oronasal coupling in children and adults. Lakshmi Venkatesh and Christopher A. Moore (Dept. of Speech & Hearing Sci., Univ. of Washington, 1417 NE, 42nd St., Seattle, WA 98105)

Oronasal coupling in young children remains poorly understood primarily because of experimental challenges associated with non-invasive techniques for transducing velopharyngeal movement. Productions of the word [bama] were studied in children with normal speech acquisition (NSA) and children with speech delay (SD; 3-to-5 years of age) with respect to adults. Nasal vibration was transduced using a small, lightweight accelerometer affixed to the lateral alar cartilage of the participant's nose. The variability in the nasal acceleration signals obtained for multiple productions of the stimuli for each participant was described using the spatiotemporal index (STI) and functional data analysis (FDA). The spatiotemporal index served as a composite method of variability following linear time normalization of the signals. The FDA technique involving nonlinear time normalization allowed for independent evaluation of amplitude and phase variability. The two groups of children with NSA and SD were more variable than the adults on both measures of STI and amplitude variability, while the amount of phase variability was similar between children and adults. These methods have complementary merits and potential in revealing the properties of development and disruption of oronasal coupling in children and adults. [Work supported by NIH-NIDCD.]

4pSC14. Final devoicing in Russian: Acoustic evidence of incomplete neutralization. Olga Dmitrieva (Dept. of Linguist., Univ. of Kansas, 1541 Lilac Ln., Blake Hall, Rm. 427, Lawrence, KS 66044-3177, olga@ku.edu)

A number of acoustic and perceptual studies conducted on German, Polish, Catalan, and Dutch found evidence of incomplete neutralization of the voicing contrast of obstruents in word-final position. The present study investigates the acoustic correlates of word-final stops and fricatives in Russian. 34 minimal pairs differing in the underlying voicing characteristics of the final segment were incorporated into the body of filler items organized as a stream of associations. This word list was presented to 14 native speakers of Russian. Measurements were obtained for the duration of the vowel preceding the final obstruent, the closure/frication portion of the final obstruent, the burst for the final stops, and duration of the voicing into closure/frication of the final obstruent. Statistical analysis revealed that the effect of underlying voicing, as well as manner of articulation, was significant for all parameters measured. The results strongly suggest that final devoicing in Russian represents a case of incomplete neutralization, at least in the experimental conditions employed.

4pSC15. Haptic-auditory interference from air flow in speech perception. Yoko Ikegami, Diana Gibraiel, Bryan Gick, and Kristin Johannsdottir (Dept. of Linguist., The Univ. of British Columbia, E270-1866 Main Mall, Vancouver, BC, Canada V6T 1Z1)

Previous work on haptic interference in auditory perception has shown McGurk-like effects from manual-tactile contact with the face [Fowler and Dekle, JEP:HPP **17**, 816–828 (1991)]. The present study investigates whether indirect haptic input affects auditory perception. A novel method was developed in which one experimenter blew puffs of air onto a subject's neck while another produced English plosives, creating possible mismatches between the sensation of aspiration and its acoustic presence. Subjects were blindfolded and wore headphones playing white noise. For 50% of the trial, one experimenter, whose presence was hidden from the subjects, blew puffs of air on subjects' necks lightly enough to be perceived but not noticeable as unnatural. Simultaneously, a second experimenter produced syllables with bilabial plosive onsets (aspirated /p/ or unaspirated /b/) and subjects were asked to repeat what was heard. Sessions were videotaped and three observers rated successful simultaneity of stimuli. Results indicate cross-modal interference. Subjects showed higher accuracy of speech perception when appropriate haptic stimuli accompanied the auditory stimuli. Moreover, when presented with mismatched

tactile and auditory stimuli, subjects demonstrated the fusion of the two modes. Subjects perceived /pa/ when auditory stimulus /ba/ was presented with emulating aspiration. [Work supported by NSERC.]

4pSC16. The (non) categorical perception of place assimilated coronal stops. David W. Gow, Jr. (Cognit./Behavioral Neurology Group, Massachusetts General Hospital, Ste. 340, 175 Cambridge St., Boston, MA 02114, gow@helix.mgh.harvard.edu)

English coronal place assimilation generally produces gradient modification of stop place cues. Acoustic analyses are presented that show that labial assimilation of coronal stops often produces a distinctive pattern of formant movement, hereafter referred to as the coronal step. The coronal step is characterized by an initial pattern of $F1$, $F2$, and $F3$ movement consistent with coronal closure, followed by a second wave of movement producing formant values at offset that are intermediate between those associated with coronal and labial stops. The perceptual consequences of this pattern were examined in categorization and 4I2AFC discrimination tasks using a linear /t/-/p/ synthetic VC continuum, as well as a continuum displaying a coronal step but produced by manipulating the same acoustic parameters. The linear continuum produced data consistent with strong categorical perception, while the stepped continuum showed no evidence of categorical perception. These results are discussed in the context of a model of the perception of assimilated speech that relies on the simultaneous activation of competing phonetic categories by assimilated segments. [Work supported by the NIH.]

4pSC17. Position and place effects in Russian word-initial and word-medial stop clusters. Alexei Kochetov (Dept. of Linguist., Simon Fraser Univ., 8888 Univ. Dr., Burnaby, BC, Canada V5A 1S6, alexei_kochetov@sfu.ca) and Louis Goldstein (Haskins Labs./Yale Univ., New Haven, CT 06511-6695)

Studies of inter-gestural timing have shown that (i) word-initial obstruent clusters tend to exhibit less gestural overlap than word-medial or word-boundary clusters, and (ii) the degree of overlap is further affected by the place of articulation of the obstruents (Byrd, 1996; Chitoran, Goldstein, and Byrd, 2002). Both findings have been attributed to perceptual recoverability considerations. This paper presents results of a magnetic articulometer (EMMA) study of Russian word-initial and word-medial stop clusters (e.g., [pt]ashka little bird versus la[pt]a bat). Data collected from 3 native speakers of Russian show that clusters with coronals and dorsals as C1 ([tk], [kt], [kp], [tjm]/[djb]) exhibit less overlap word-initially than word-medially, while the cluster with the labial as $C1$ ([pt]) does not exhibit the same timing pattern. The findings are interpreted as providing additional support for the role of perceptual recoverability in intergestural timing. First, less overlap in word-initial clusters, compared to word-medial clusters, ensures better place recoverability of $C1$ (cf. Chitoran *et al.*, 2002). Second, unreleased labials are more perceptually robust than unreleased coronals (Byrd, 1992; Surprenant and Goldstein 1998) and dorsals (Wright, 2001; Kochetov and So, 2005), and thus do not require the same degree of overlap. [Work supported by SSHRC.]

4pSC18. Self-organizing maps for measuring similarity of audio-visual speech percepts. Hans-Heinrich Bothe (Tech. Univ. of Denmark, Ctr. for Appl. Hearing Res., Oersteds Plads b.352, DK-2800 Lyngby, Denmark, hhb@oersted.dtu.dk)

The goal of this work is to find a way to measure similarity of audio-visual speech percepts. Phoneme-related self-organizing maps (SOM) with a rectangular basis are trained with data material from a (labeled) video film. For the training, a combination of auditory speech features and corresponding visual lip features is used. Phoneme-related receptive fields result on the SOM basis; they are speaker dependent and show individual locations and strain. Overlapping main slopes indicate a high similarity of respective units; distortion or extra peaks originate from the influence of other units. Dependent on the training data, these other units may also be contextually immediate neighboring units. The poster demonstrates the idea with text material spoken by one individual subject using a set of simple audio-visual features. The data material for the training process consists of 44 labeled sentences in German with a balanced phoneme repertoire. As a result it can be stated that (i) the SOM can be trained to map auditory and visual features in a topology-preserving way and (ii) they show strain due to the influence of other audio-visual units. The SOM can be used to measure similarity amongst audio-visual speech percepts and to measure coarticulatory effects.

4pSC19. Perception of "asymmetrical" German vowels by humans, monkeys and gerbils. Joan M. Sinnott, Leigh Ann Long, and Allen C. Ernst (Psych. Dept., Univ. of South Alabama, Mobile, AL 36688)

Polka and Bohn [J. Acoust. Soc. Am. **100**, 557–592 (1996)] report that human infants perceive certain vowels asymmetrically when tested with a repeating-background category-change operant headturn procedure. Specifically, discrimination is easier when the background is a more central vowel (e.g. /y/) and the target is a more peripheral vowel (e.g. /u/), compared to the opposite condition. We are testing human (adult native English listeners) and monkey discrimination of the German vowel pair /dut-dyt/, using both synthetic and natural stimuli (obtained from Polka and Bohn), and a within-subject design. So far, no significant asymmetries have emerged in the data using a percent correct measure. We will also report preliminary data from gerbils being trained in a between-subject design more comparable to that used with human infants. Plans are to continue testing all species until they reach asymptotic performance levels in order to determine if an RT measure will indicate vowel asymmetries. A video will be shown of humans, monkeys and gerbils working on the experiments.

4pSC20. Effects of reduced orosensory feedback on spectral characteristics of medioalveolar /s/. Juha-Pertti Laaksonen, Stina Ojala, Olli Aaltonen (Dept. of Phonet., Univ. of Turku, FIN-20014 Turku, Finland, juhlaa@utu.fi), Matti Niemi, and Risto-Pekka Happonen (Univ. of Turku, FIN-20520 Turku, Finland)

Effects of reduced orosensory feedback on the production of fricative sounds were studied by measuring spectral characteristics of medioalveolar /s/. Five Finnish male speakers produced sibilant /s/ embedded in 8 different word contexts under normal condition and under reduced condition, in which the tactile information from the tongue was reduced by blocking the lingual nerve on the right side by local anesthesia. Parameters of Long Time Average (LTA) spectrum (i.e., center of gravity, standard deviation, skewness, and kurtosis) were measured by Praat software for every speaker. In comparison between the two conditions, center of gravity, standard deviation, skewness, and kurtosis changed for every speaker. However, the changes were variable and individual. The results of acoustic analysis show that reduced tactile sensation have effects on tongue function producing spectral alterations for sibilant /s/. The inter-individual variation between different talkers suggests that there are no general compensatory mechanisms of speech production, but the mechanisms are highly speaker-dependent.

4pSC21. BMSA, a new Bayesian model selection criterion for assessing audiovisual models of speech perception. J. L. Schwartz (Institut de la Commun. Parle, UMR 5009 CNRS, INPG, Université Stendhal, 46 Av. Flix Viallet, 38031 Grenoble Cedex 1, France)

Audiovisual speech perception has provided matter for many model comparison and assessment studies. Most involved "Root Mean Square Error" (RMSE), a criterion based on the differences between predicted and observed probabilities of response in auditory, visual and audiovisual categorization experiments. RMSE is related to the best fit of a model considering data. However, some models are known for their ability to

adapt to almost any data set. This is always associated to a large fit instability: very small variations of the model parameters lead to dramatic modifications of the predicted data. A new criterion is derived from the Bayesian Theory of model selection. Considering that best fit estimation is not error-free, model comparison in this theory involves the total likelihood of the model knowing the data, integrating on the whole parameter space. This results in penalizing a too flexible model, for which local likelihood is high, but global likelihood is low. An approximation of this criterion, called BMSA, is presented. This criterion, easy to compute, results in combining fit and stability. An application of the BMSA criterion in an audiovisual speech perception experiment is proposed, enabling to show for the first time that the fusion process is subject-dependent.

4pSC22. Age differences in detecting gaps in speech and non-speech. M. Kathleen Pichora-Fuller, Bruce A. Schneider (Dept. of Psych., Univ. of Toronto, 3359 Mississauga Rd., Mississauga, ON, Canada L5L 1C6, kpfuller@utm.utoronto.ca), Nancy Benson (The Hospital for Sick Children, Toronto, ON, Canada M5G 1X8), Stanley Hamstra (Univ. of Toronto, Toronto, ON, Canada M5G 1L5), and Edward Storzer (McNeill Audiol., Victoria, BC, Canada V8R 1G1)

Ability to detect gaps in speech and non-speech stimuli was measured in children, young adults, and older adults with good audiograms. The markers varied in duration (40 vs 250 msec) and in spectral symmetry. In spectrally symmetrical conditions, the leading and lagging markers were the same: the vowel [u] in speech conditions and a 500-Hz tone in non-speech conditions. In asymmetrical speech conditions, the lagging marker was the same as in the symmetrical conditions, but the leading marker was the consonant [s] in the speech conditions and a broadband noise (1 to 6 kHz) in the non-speech conditions. For all groups, gap detection thresholds in spectrally symmetrical markers were far smaller than in spectrally asymmetrical markers. Thresholds were significantly smaller in young adults than in either children or older adults. Gaps between spectrally asymmetrical speech markers were detected better than gaps between analogous non-speech stimuli. It is argued that phonological knowledge compensates for auditory processing difficulties. [Research funded by the International Dyslexia Association, the Natural Sciences and Engineering Research Council of Canada, and the Canadian Institutes of Health Research.]

4pSC23. Voice onset time in Mandarin esophageal speech. Manwa Ng (Long Island Univ., 720 Northern Blvd., Brookville, NY 11548-1300, manwa.ng@liu.edu) and Hanjun Liu (Xi'an Jiaotong Univ., Xi'an, 710049, P. R. China)

As an important perceptual cue for voicing and aspiration of stops, voice onset time (VOT) is mainly determined by the aerodynamic interaction between the intraoral and subglottal regions. However, since the PE segment serves as a new vibratory source in esophageal phonation, aerodynamic events are very different from laryngeal phonation. VOT associated with esophageal speech of English has been reported previously. However, few studies have reported VOT characteristics of esophageal speech of tone languages. The present study will investigate the possible VOT difference between esophageal and normal laryngeal speakers of Mandarin Chinese. Seven superior esophageal speakers and 7 normal laryngeal speakers will participate in the present investigation. They will be native male speakers of Mandarin Chinese. The participants will produce the syllable /ta/ embedded in a carrier phrase at a comfortable loudness level for three times. VOT values will be measured from a time domain waveform. With reference to a wide-band (300 Hz) spectrogram, VOT will be defined as the release of stop and the first identifiable period of the vocalic portion. An independent sample t-test will be used to determine if VOT values of esophageal and normal laryngeal speakers were significantly different.

4pSC24. A role for tracheal resonances in speech perception. Asaf Bachrach (Dept. of Linguist. and Philosophy, MIT, Cambridge, MA 02139), Steven Lulich (MIT, Cambridge, MA 02139, lulich@mit.edu), and Nicolas Malyska (MIT, Cambridge, MA 02139)

Acoustic coupling between the vocal tract and the trachea results in the introduction of pole-zero pairs corresponding to resonances of the uncoupled trachea. If the second formant ($F2$) passes through the second tracheal resonance ($T2$) a discontinuity in amplitude occurs. This work explores the hypothesis that the $F2-T2$ discontinuity affects how listeners perceive the distinctive feature [back] in transitions from a front vowel (high $F2$) to a labial stop (low $F2$). We synthesized two versions of an utterance (''apter'') with an $F2-T2$ discontinuity at different locations in the initial VC transition. Subjects heard portions of the utterance with and without the discontinuity, and were asked to identify the utterance. Results show that the presence of the $F2-T2$ discontinuity facilitated the perception of frontness in the vowel. Discontinuities of the $F2-T2$ sort are proposed to play a role in shaping vowel inventories in the world's languages [K. N. Stevens, J. Phonetics **17**, 3–46 (1989)]. Our results support a model of lexical access in which acoustic discontinuities subserve phonological feature identification.

4pSC25. Targetless schwa revisited. Edward Flemming (Dept. of Linguist. and Philosophy, M.I.T., 77 Massachusetts Ave., 32-D808, Cambridge, MA 02139, flemming@mit.edu)

It has been suggested that schwa vowels in English lack vowel quality targets, that is, they are realized as interpolations between the preceding and following segmental contexts or are the result of separating two consonantal constriction gestures, without an inherent vocalic gesture. Tests of this hypothesis have produced mixed results. The present study revisits the question of schwa targets based on acoustic analysis of schwa vowels in a wider range of segmental and morphological contexts. The main conclusions are: (i) It is important to distinguish stem-final reduced vowels (as in ''pandA'') from other schwa vowels (e.g., ''About,'' ''bEgin''). The former clearly have vowel quality targets they are generally mid central vowels whereas the latter are much more contextually variable. (ii) The variable schwa vowels have a target, but this target is not a particular vowel quality, it is to indicate the presence of a vowel. This is achieved through a variety of means, including duration and realization of a local amplitude peak, but is consistent with substantial variation in vowel quality, particularly $F2$.

4pSC26. The role of attention in infant phonetic perception. Monika Molnar, Linda Polka, and Susan Rvachew (McGill Univ., 1266 Pine Ave. W., Montreal, QC, H3G 1A8 Canada monika.molnar@mcgill.ca)

Attention is an important factor underlying phonetic perception that is not well understood. In this study we examined the role of auditory attention in infant phonetic perception using a distraction masker paradigm. We tested infant discrimination of /bu/ vs /gu/ with a habituation procedure and three natural productions of each syllable. For the quiet condition each token was copied into a separate sound file. For the distractor condition, a high frequency noise was added to each sound file so that it gated on and off with the onset and offset of the syllable. The distractor noise was a recording of bird and cricket songs whose frequencies did NOT overlap with the test syllables. Thus, the noise did not change the audibility of the syllable, but it could distract infants if they do not focus their attention well. Infants (6- to 8-month-olds) were tested in each condition. Infants tested in quiet performed significantly better than infants tested in the distractor condition; discrimination scores showed little overlap between

the two groups. These findings indicate that in young infants, attention to subtle phonetic differences is easily disrupted. The implications for developmental models of speech perception will be discussed.

4pSC27. The development of laryngeal coarticulation: Comparison of women, 5-year-olds, and 10-year-olds. Laura L. Koenig (Haskins Labs & Long Island Univ., Brooklyn)

Past work has indicated that adult speakers show effects of laryngeal coarticulation in voice source measures taken in vowels flanking voiceless consonants. In a recent pilot study, we presented time-varying voice source measures from 5-year-old girls producing VCV sequences with varying consonants. The results suggested that, on average, the children produced laryngeal coarticulation over a duration at least as long as adults, but extensive token-to-token variability complicated interpretation of the results. The present analysis extends that work by comparing normal women with normally-developing 5-year-old and prepubertal 10-year-old children. Speakers were recorded producing multiple repetitions of intervocalic /b p h/ in simple carrier phrases such as ''Poppa Hopper'' while oral airflow signals were collected using a Rothenberg mask. After software-inverse filtering, pulse-by-pulse measures of DC airflow, open quotient, and $f0$ are made from the transition out of the consonant into the following vowel. Results will be intepreted in terms of age effects and, for the children, possible gender differences. These data have implications for our understanding of laryngeal motor control in children and, more generally, for theories of the development of coarticulation. [Work supported by NIH.]

4pSC28. Tonal and durational variations as phonetic coding for syllable grouping. Yi Xu (Univ. College London, Wolfson House, 4 Stephenson Way, London NW1 2HE, UK) and Maolin Wang (College of Chinese Lang. and Culture, Jinan Univ., Guangzhou, 510610, China)

While syllables in connected speech are generally believed to be prosodically divided into groups, how such grouping is done phonetically is not fully understood. This study explores the possibility that syllable grouping is partly realized through adjusting articulatory strength and duration. We compared the degrees of tonal undershoot in Mandarin as they are related to syllable position and number of syllables in words or phrases. The sequences consisted of 1–4 syllables with R or F tone and were produced by eight speakers. The all-R and all-F sequences impose great pressure on tone production and hence would best reveal the effects of strength and duration. Results show that as the number of syllables in each sequence increased, both syllable duration and size of $F0$ excursion decreased. Meanwhile, excursion size varied with syllable location in the sequence. But in each case it was the first and last syllables that had the largest excursions, and the excursion variations cannot be fully accounted for by duration. There thus appear to be both an isochrony effect and an ''edge marking'' effect. Taking into consideration known prosodic effects on segmental phonemes, syllable grouping as a communicative function seems to involve a rather complex encoding scheme.

4pSC29. Is speech lazy or just efficient? A control-theoretic analysis. Luis Rodrigues (Dept. of Mech. and Industrial Eng., Concordia Univ., 2160B Bishop St., B-304 Montreal, QC, Canada H3G, luisrod@me.concordia.ca) and John Kroeker (Eliza Corp., Beverly, MA 01915)

This paper introduces a control-theoretic model that allows us to address the energy dynamics of the vocal tract system. The model can be generated directly from articulatory data. The model allows computation of the energy in a state transfer, from an initial to a final articulator configuration. This method can help determine the degree of physical feasibility of various proposed articulatory trajectories. The basic assumption is that the set of articulators evolve through configurations that minimize the energy spent by the system to produce an utterance. Minimum control energy gives a measure of how hard it is to reach a target point for the different articulators. Simulation results are presented corresponding to the computation of the minimum energy from the MOCHA database. The linear model is shown to be adequate for short, well-labeled segments. The results show the intriguing fact that minimum control energy seems to have an oscillatory (swinging) nature for the production of speech. Physical features such as time constants and natural frequencies of the articulators are derived. A control-theoretic model of the dynamics of the mechanical articulators of speech production could be a fundamental tool to understand the mechanism of speech production.

4pSC30. Effects of simultaneously presented pitch- and loudness-shifted voice auditory feedback on voice fundamental frequency. Charles R. Larson, Jean Sun, and Hideki Takaso (Northwestern Univ., 2240 Campus Dr., Evanston, IL 60208)

Recent studies have demonstrated that subjects respond to pitch- or loudness-shifted voice feedback with compensatory changes in voice fundamental frequency ($F0$) or intensity. In the present study, subjects sustaining /u/ vowel sounds were presented with either pitch-shifted auditory feedback (50 or 100 cents) or simultaneous pitch-shifted and loudness-shifted voice feedback (3 or 6 dB SPL) of 200 ms duration. During simultaneous stimulus presentations, the stimulus directions were opposite in direction (e.g., increasing pitch and decreasing loudness feedback). Subjects ($N=24$, normal, age 18–24) responded to pitch-shifted feedback with compensatory adjustments in $F0$. Subjects responded to simultaneous stimuli with independent compensatory adjustments in voice $F0$ and intensity. Latencies of $F0$ responses to simultaneous stimuli ($m=211$ ms) were longer than to pitch-shifted stimuli alone ($m=145$ ms, $F=26$, df 2306; $p<.0001$). Results suggest auditory feedback mechanisms for stabilizing the direction of voice $F0$ against perturbed pitch feedback are independent of mechanisms for stabilizing loudness feedback, but the simultaneous presence of pitch and loudness perturbations slows down the corrective mechanism for pitch-shifted feedback.

4pSC31. Phoneme clustering based on segmental lip configurations in naturally spoken sentences. Jianxia Xue, Abeer Alwan (Dept. Elec. Eng., Univ. of California, Los Angeles, 405 Hilgard Ave., Los Angeles, CA 90095, jxue@ee.ucla.edu), Jintao Jiang, and Lynne E. Bernstein (House Ear Inst., Los Angeles, CA 90057)

It has been shown that face (lips, cheeks, and chin) information can account to a large extent for visual speech perception in isolated syllables and words. Visual speech synthesis has used small reduced sets of phonemes (''visemes''), under the theory that perceivers are limited in their ability to extract visual speech information. In this study, lip configurations from a manually segmented sentence database [L. Bernstein *et al.*, J. Acoust. Soc. Am. **107**, 2887 (2000)] were analyzed to provide phoneme clusters that are algorithmically distinguishable using mouth vertical/horizontal opening and lip protrusion from the middle position of each segment. The lip feature sample spaces for each phoneme were represented by Gaussian mixture models. Maximum posterior probability classification results were computed for each phoneme. Confusion matrices were generated from the classification results, and a set of confusions with 74% or higher within-group classification correct was judged to be a cluster. Preliminary results from 191 sentences by a single talker generated the following clusters: {/p, b, m/(77%), /f, v/(74%), /w, r/(80%), /t, d, s, z, D, k, n/(88%)}. We will present results analyzing the entire English phoneme set across different talkers and compare the results with visual perceptual clusters. [Work supported in part by the NSF.]

4pSC32. The effects of signal-to-noise ratio on auditory-visual integration: Integration and encoding are not independent. Mitchell S. Sommers (Dept. of Psych., Washington Univ., Campus Box 1125, St. Louis, MO 63130, Msommers@wustl.edu), Brent Spehar, and Nancy Tye-Murray (Washington Univ. School of Medicine, St. Louis, MO 63130)

Most current models of auditory-visual speech perception propose a two-stage process in which unimodal information is extracted independently from each sensory modality and is then combined in a separate integration stage. A central assumption of these models is that integration is a distinct perceptual ability that is separate from the ability to encode unimodal speech information. The purpose of the present study was to evaluate this assumption by measuring integration of the same speech materials across three different signal-to-noise ratios. Twelve participants were presented with 42 repetitions of 13 consonants presented in an /iCi/ environment at 3 different signal-to-noise ratios. Integration was assessed using an optimum processor model [L. Braida, Q. J. Exp. Psych. **43A**, 647–677 (1991)] and a new measure termed integration efficiency that is based on a simple probability metric. In contrast to predictions made by current models of auditory-visual speech perception, significant differences were observed for both measures of integration as a function of signal-to-noise ratios. These findings argue against strictly serial models of auditory-visual speech perception and instead support a more interactive architecture in which unimodal encoding interacts with integration abilities to determine overall benefits for bimodal speech perception. [Work supported by NIA.]

4pSC33. The University of South Florida audiovisual phoneme database, v 1.0. Stefan A. Frisch, Sarah Hardin, Dee Adams Nikjeh, and Adrienne M. Stearns (Dept. of Commun. Sci. and Disord., Univ. of South Florida, Tampa, FL 33620, frisch@cas.usf.edu)

An audiovisual database of English speech sounds has been developed for teaching purposes. This database contains a variety of Standard English speech sounds produced in clear speech in isolated words. Phonemes are produced in word initial, word medial, and word final position, unless not allowed by English phonotactics. There is one example of each word spoken by a male talker of Standard American English from the Midwest. The database consists of individual word files that contain simultaneous audio recording, video of the lips, flexible endoscopic image of the pharynx and larynx, and ultrasound video of the tongue in the mid-saggital plane. The files in the database are uncompressed video avi format and are suitable for examination in the Wavesurfer freeware program (Sjolander and Beskow, KTH Stockholm). This database is intended as a multimedia reference for students in phonetics or speech science. Its coverage is relatively complete, though there are some gaps due to technical difficulties with the recording procedure. A demonstration of selected recordings from the database and suggestions for their use will be presented. Plans for further development of the database will also be discussed. [Work supported by a USF Innovative Teaching Grant.]

4pSC34. Generalization of the imitation effect within a natural class. Kuniko Nielsen (Dept. of Linguist., UCLA, 3125 Campbell Hall, Los Angeles, CA 90095)

A natural class refers to a set of segments which share acoustic or articulatory features. Support for this notion has traditionally been provided by phonological alternations and phonotactic constraints, and recently by experiments (e.g., Goldrick, 2004). This study investigates the psychological reality of natural class using the imitation paradigm (Goldinger, 1998) in which subjects speech is compared before and after they are exposed to target speech (= study phase). Although this paradigm has shown that subjects shift their production in the direction of the target, these results do not reveal the size of the linguistic unit(s) influenced by the effect. That is, when a subject shifts production of a particular sound in a word, it is uncertain whether the subject is picking up on the word, the segment, or the feature. In this study, to investigate whether phonetic imitation is generalized across members of a natural class, the study-phase word list includes words with initial /p/ and /t/ (with extended VOT), while the pre- and post-study production list includes (1) the modeled words, replicating Shockley *et al.* (2004), (2) the modeled segments /p/ and /t/ in new words, and (3) the modeled feature [+spread glottis] (aspiration) in a new segment /k/.

4pSC35. Functional MRI reveals two distinct cerebral networks subserving speech motor control. Axel Riecker (Dept. of Psychiatry III, Univ. of Ulm, Leimgrubenweg 12-14, D-89075 Ulm, Germany), Klaus Mathiak (Univ. of Aachen, Pauwelsstr. 30, D-52074 Aachen, Germany), Wolfgang Grodd (Univ. of Tuebingen, D-72076 Tuebingen, Germany), Ingo Hertrich, and Hermann Ackermann (Univ. of Tuebingen, D-72076 Tuebingen, Germany, hermann.ackermann@uni-tuebingen.de)

To further delineate the neural basis of speech motor control, functional magnetic resonance imaging (fMRI) was performed during syllable repetitions synchronized to click trains (8 subjects, 2–6 Hz; vs. passive listening task). (a) Bilateral hemodynamic responses emerged within mesiofrontal and sensorimotor cortex, putamen / pallidum, thalamus and cerebellum (two activation spots at either side). In contrast, dorsolateral premotor cortex and anterior insula showed left-sided activation. (b) Calculation of rate / response functions revealed a negative linear relationship between repetition frequency and hemodynamic activation within the striatum whereas both cerebellar hemispheres exhibited a step-wise response increase at about 3 Hz. (c) Analysis of the temporal dynamics of hemodynamic activation revealed these cortical and subcortical brain regions to be organized into two separate networks (medial and dorsolateral premotor cortex, anterior insula, superior cerebellum versus sensorimotor cortex, basal ganglia, inferior cerebellum). These data provide evidence for two levels of speech motor control bound, most presumably, to motor preparation and executions processes. Furthermore, these findings help to explain clinical observations such as an unimpaired or even accelerated speaking rate in Parkinsons disease and slowed speech tempo which does not fall below a rate of about 3 Hz in cerebellar disorders.

4pSC36. A phonetic study of guttural laryngeals with data from Semitic. Kimary Shahin (Effat College of Al-Faisal U., P.O. Box 34689, Jeddah 21478, Saudi Arabia, kshahin@effatcollege.edu.sa)

Phonetic data from Hebrew and Arabic were examined to determine if laryngeal consonants in those languages have tongue root articulation. This was done in search of an articulatory basis for the phonological patterning of Semitic laryngeals with tongue root consonants like pharyngeals and uvulars. This issue has been problematic for phonological theory because a theoretically comfortable phonetic basis for that patterning has not yet been identified [J. J. McCarthy, Pap. Lab. Phonol. **III**, 191–234 (1994); F. Nolan, *ibid.* IV, 361–367 (1995)]. The present study is the first to address the problem with clear natural-language data. Acoustic and articulatory data are presented (digital audio and video, waveform, spectrogram and video picture) from two laryngoscopic films: Prof. A. Laufer's film of Hebrew and Arabic speech recorded at Haskins Labs in the early 1980s, and a new film of Arabic speech recorded at Speech Technology Research. The data show no tongue root articulation for the laryngeals. The implications for phonology are discussed. Note: Thanks to Prof. Laufer and Haskins Labs for permission to use the Haskins film, and to Prof. John Esling for collaboration in recording the new Arabic data. [Work supported by a SSHRCC grant to J. Esling.]

4pSC37. Quality of American English back vowels before /r/. Michael J. Clark and James M. Hillenbrand (Speech and Hearing Ctr., MS5355, Western Michigan Univ., Kalamazoo, MI 49008)

The vowels /o/ and /ɔ/ are not contrastive before /r/ in most American English dialects, and the phonetics literature is equivocal about the phonetic quality of the nucleus in words such as *board*. Most works use [ɔ] to

represent the first part of the nucleus. In this study acoustic measurements, listening tests, and discriminant analyses were used to determine the phonetic quality of such vowels. Fourteen women recorded monosyllables containing /o,ɔ,ɑ/, with initial /b,g,h/ and final /d/ or /z/ (e.g., *bode, Baud, bod*). Additionally a set with rhotic diphthongs (*board, barred, gored, guard, hoard, hard*) was recorded. The central back vowel in rhotic diphthong syllables (*board, gored, hoard*) showed formant values very similar to those for /o/ and very unlike those for /ɔ/. The low back rhotic diphthong (i.e., *barred, guard, hard*) showed a range of values from /ɑ/ to /ɔ/. Listening tests using brief excerpts from vowel onsets supported the greater similarity of *board* to *bode* than to *Baud* and the intermediate nature of *barred* between *bod* and *Baud*. Discriminant classification of the rhotic diphthong formant measures supported the same conclusions. [Work supported by NIH.]

4pSC38. Articulatory and acoustic characteristics of English /l/ in children's speech production. Sunyoung Oh and Bryan Gick (Dept. of Linguist., Univ. of British Columbia, Vancouver, BC, Canada V6T 1Z1, oh.sun_young@courrier.uqam.ca)

English /l/ is one of the later-developing sounds in language acquisition. It has syllable-based allophones (e.g., light l in initial, dark l in final) and comprises multiple articulatory gestures (e.g., tongue tip fronting and raising, tongue dorsum backing, lateral dipping) with different coordination in timing and magnitude in syllable position (e.g., synchronicity in initial, tongue tip delay in final with greater reduction in magnitude). Using ultrasound, this study examined how articulatory characteristics of /l/ are presented in children's speech production. Replicating previous articulatory studies in adult speech production, /l/ words in isolation and combination were collected from eight monolingual children aged 3;11 to 5;9. Although some children produced more similar to adult /l/, children produced /l/ using different articulatory properties. /l/ was produced with less detail and more variations than in the adults' speech as movements of the tongue were simplified or modified. While articulatory performance varied across subjects as well as syllable positions, acoustic analysis of formants ($F1$, $F2$, $F3$) of sample tokens showed similar patterns for all subjects. This study suggests that the tendency toward late acquisition of /l/ is due directly to the articulatory as well as motor complexity.

4pSC39. The effects of tongue shape categories on tongue segmentation in English. Melissa A. Epstein and Maureen Stone (Biomed. Sci., Univ. of Maryland Dental School, Rm. 5A12, 666 W. Baltimore St., Baltimore, MD 21201)

In our recent work, we have proposed that the tongue moves by compressing and expanding local functional segments. For any single gesture, functional segments may move in similar or opposite directions to compress and expand the tongue locally. High correlations between segments suggest biomechanical constraints. Low correlations suggest independent control of these segments. Our previous studies on English have shown a front-back division of the tongue, where adjacent segments correlate positively with each other (moving in the same direction) and distal segments correlate negatively with each other (moving in opposite directions). Individual segments aggregate with adjacent or distal segments. Furthermore, the phonemic content of the dataset influences these aggregations and the location of the front-back division of the tongue. This study will more deeply explore these phonemic effects. In particular, we will examine the effects of the transitions between consonants and vowels of the four basic tongue shape categories of English [front raising (e.g., /n,i/), back raising (e.g., /ng,o/), complete channel (e.g., /s,ae/) and two point displacement (e.g., /l/)] on tongue segmentation and the location of the pivot point for the front-back division. [Work supported in part by NIDCD/NIH Grant RO1-DC01758 and by NIH Grant T32-DE07309.]

THURSDAY AFTERNOON, 19 MAY 2005 — GEORGIA B, 2:00 TO 3:45 P.M.

Session 4pUWa

Underwater Acoustics: Propagation: Modeling and Experimental Results II

Kevin B. Smith, Chair

Dept. of Physics, Naval Postgraduate School, Monterey, CA 93943

Contributed Papers

2:00

4pUWa1. Multipath cancellation with a two-channel array. John E. Piper (NSWC-PC, Code R21, 110 Vernon Ave., Panama City, FL 32407, john.e.piper@navy.mil)

A test pool experiment with direct and multipath signals incident on a small vertical array was conducted. To separate these signals a maximum likelihood method approach was used. This approach is based on exploiting the orthogonal nature of the signals in the maximum likelihood parameter space, which leads to a theoretically complete decoupling of the desired signal from the multipath interference. This approach has no analog in conventional signal processing. Results of the test pool experiment using this maximum likelihood method cancellation and conventional beamforming methods are presented.

2:15

4pUWa2. Torpedo detection using mutli-path signals and fast orthogonal search techniques. Jeff Collins, Donald McGaughey (Dept. of Phys., Royal Military College of Canada, P.O. Box 17000, Stn Forces Kingston, ON, Canada K7K 7B4, jeff.collins@rmc.ca), Jim Theriault, and Sean Pecknold (Defence Res. and Development Canada (Atlantic), Dartmouth, NS, Canada B2Y 3Z7)

Detecting a high speed torpedo by means of a passive acoustic detector is very challenging for most acoustic operators. Coupled with a very noisy environment, multiple sources in a multi-path scenario and varying environmental factors, a time-constrained assessment will prove difficult. In addition, a passive sensor cannot estimate the range of a torpedo approaching it at a constant bearing. The passive acoustic sensor will receive a direct path signal from the torpedo as well as a signal that has reflected off

the surface. Due to the different angle of arrival, the direct-path and surface-reflected signals have different Doppler shifts. The Torpedo Detection Algorithm (TDA) employs the fast orthogonal search (FOS) algorithm for high-resolution spectral analysis to detect the closely spaced direct-path and surface-reflection signals. When a direct-path and surface-reflection are found, an automatic alert of a torpedo detection is initiated. In simulation, a torpedo is detected 20 times out of 20 as it travels from 5000 to 500 m from the receiver. Simple trigonometric expressions are used to estimate the torpedos range given the two frequencies estimated by FOS and *a priori* information about the torpedo speed and depth.

2:30

4pUWa3. A comparison of mine counter measure performance models. Andrew Holden (Dstl. Winfrith, Dorchester, DT2 8WX, UK, apholden@dstl.gov.uk)

Mine counter measure (MCM) sonar systems perform the task of detection and classification of marine mines that are typically laid in shallow water environments. Currently, there are several MCM performance models in use that can predict the performance of MCM sonars. This paper gives a brief description of some these models and gives a comparison of their performance predictions for several shallow water scenarios. All the models examined are self contained packages that can model the entire problem—the sonar, the environment, and the target. They are considered to be energy models in that only the intensity of sound received from various parts of the environment is modeled while phase calculations are ignored. The results show that the models can give good agreements with each other for some scenarios. In some other scenarios the agreement is not so good and reasons are given to show why this happens.

2:45

4pUWa4. Tank experiments and model comparisons of shallow water acoustics over an elastic bottom. Jon M. Collis, William L. Siegmann (Rensselaer Polytechnic Inst., Troy, NY 12180, collisj@rpi.edu), Michael D. Collins, Erik C. Porse, Harry J. Simpson, and Raymond J. Soukup (Naval Res. Lab., Washington, DC 20375)

A series of tank experiments are being conducted in order to obtain high quality data for acoustic propagation in environments with sloping elastic bottoms. Such problems can now be solved accurately with the parabolic equation method, which is being used to model the experiment. This paper will present results of the initial experiments and discuss plans for upcoming experiments, which will include propagation onto land. The initial experiments involved a broadband source over a block of PVC that was suspended in deionized water. Time series were collected at 100 to 300 kHz on horizontal and vertical arrays for two source positions. [Work supported by the Office of Naval Research.]

3:00

4pUWa5. Adjoint-based control of nonlocal boundary conditions for Claerbout's wide-angle parabolic approximation. Matthias Meyer and Jean-Pierre Hermand (Dept. of Optics and Acoust., Univ. Libre de Bruxelles, av. F-D. Roosevelt 50 - CP 194/05, B-1050 Brussels, Belgium)

This paper applies the concept of optimal boundary control for solving inverse problems in shallow water acoustics. A continuous analytic adjoint model is derived for a wide-angle parabolic equation (WAPE) using a generalized nonlocal impedance condition at the water-bottom interface. While the potential of adjoint methodology has been demonstrated for ocean acoustic tomography, this approach combines the advantages of exact transparent boundary conditions for the WAPE with the concept of adjoint-based optimal control. In contrast to meta-heuristic approaches the inversion procedure itself is directly controlled by the waveguide physics and, in a numerical implementation based on conjugate gradient optimization, much fewer iterations are required for assessment of environments that are supported by the underlying subbottom model. Furthermore, since regularization is important to enhance performance of full-field acoustic inversion, special attention is devoted to applying penalization methods to the adjoint formalism. Regularization incorporates additional information about the desired solution to stabilize ill-posed problems and identify useful solutions, a feature that is of particular interest for inversion of field data sampled on a vertical array in the presence of measurement noise and modeling uncertainty. Results show that the acoustic fields and the bottom properties embedded in the control parameters are efficiently retrieved.

3:15

4pUWa6. Line-integral prediction for horizontal coherence in deep-water propagation. Michael Vera (Univ. of Southern Mississippi, 730 E. Beach Blvd., Long Beach, MS 39560, michael.vera@usm.edu)

Some of the characteristics of an acoustic signal propagating to basin-scale ranges in the ocean can be estimated using line integrals along deterministic ray paths. These line-integral approximations involve the statistics of the internal-wave field. The success or failure of the integral expressions can be analyzed by comparison to parabolic-equation simulations through multiple realizations of the stochastic internal-wave field. One acoustic characteristic of interest is the length scale of acoustic horizontal (cross-range) coherence. Recent work comparing integral predictions of horizontal coherence length to values from simulations of the North Pacific Acoustic Laboratory experiment yielded mixed results. Additional comparisons will be discussed for acoustic propagation paths that are not impacted by range-dependence in the bathymetry or background sound-speed profile. These simulations will employ a single sound-speed profile and a deep, range-independent bottom. The comparisons can yield some insight into the accuracy of the integral estimate; they will not be influenced by bathymetric interaction or range dependence in the background sound speed.

3:30

4pUWa7. Decomposition method in constructing simulation models of parametric location for statistically irregular mediums. Irene Starchenko (TSURE, 347928, Taganrog, GSP-17a, Nekrasovskiy, 44, Russia, star@tsure.ru)

In using parametric arrays for the purposes of distant sounding in a water medium it is necessary to take into account the probable characteristics of acoustic signals. In this case the modeling of processes is especially important, because experiments in natural conditions are not always possible. In the case of parametric location the medium plays the very important function of formation of the parametric array. Results will be discussed.

Session 4pUWb

Underwater Acoustics: Underwater Noise Studies

David E. Hannay, Chair
JASCO Research Ltd., 2101-4464 Markham St., Victoria, BC V8Z 7X8, Canada

Contributed Papers

2:00

4pUWb1. Range estimation of broadband noise sources in an ocean waveguide using the array invariant. Sunwoong Lee and Nicholas C. Makris (MIT, 77 Massachusetts Ave., Cambridge, MA 02139, makris@mit.edu)

A method is developed for range estimation of broadband noise sources in a horizontally stratified ocean waveguide without *a priori* knowledge of the environment. It has previously been shown that the range of a transient source can be estimated using the ''array invariant'' method by analyzing instantaneous beam-time intensity data [Lee and Makris, J. Acoust. Soc. Am. **116**, 2646 (2004)]. This method is now extended to range estimation of continuous, broadband noise sources in an ocean waveguide. It is shown that the cross-correlation of instantaneous beam-time intensity asymptotically reaches the array invariant. The range of the source can then be determined without knowledge of the environmental parameters. This method is applied to localize multiple uncorrelated noise sources in a horizontally stratified ocean waveguide without ambiguity.

2:15

4pUWb2. Effect of shallow water internal waves on the ambient noise notch. Daniel Rouseff, Dajun Tang, and Frank S. Henyey (Appl. Phys. Lab., College of Ocean and Fishery Sci., Univ. of Washington, Seattle, WA 98105, rouseff@apl.washington.edu)

Consider a vertical receiving array in shallow water that is recording ambient noise. If the recorded noise is beamformed, the resulting beam pattern often exhibits a notch at broadside to the array. Modeling of range-independent environments suggests that this ambient noise notch should become more pronounced as the sound speed gradient is increased. In the present work, the effect of range dependence on the ambient noise notch in the 1–5 kHz band is studied. Range dependence is introduced into the model in the form of random shallow water internal waves. Coupling between the propagating acoustic modes is calculated using the Dozier-Tappert formulation [J. Acoust. Soc. Am. **63**, 353–365 (1978)] extended to consider bottom loss. The model is internally consistent as the buoyancy profile determining the internal waves is dictated by the sound speed gradient. Experimental results from the 2001 East China Sea Experiment are also reported. [Work supported by ONR.]

2:30

4pUWb3. Using vertical directionality to study the ambient noise fluctuations due to internal waves in South China Sea. Hsiang-Chih Chan (Natl. Taiwan Univ., No. 1, Sec. 4, Roosevelt Rd., Taipei, 106, Taiwan, d91525003@ntu.edu.tw), Ruey-Chang Wei, Wen-Zheng Hu (Natl. Sun Yat-sen Univ., Kaohsiang, 804, Taiwan), and Chi-Fang Chen (Natl. Taiwan Univ., Taipei, 106, Taiwan)

A range-dependent acoustic propagation model is used to simulate the complex ambient noise field, which offers the key to the understanding of noise varying during the ASIAEX 2001 SCS experiment. The surface noise sources that are generated by wind/waves and ships are distributed horizontally in the simulated model. Observed ambient noise fluctuations in South China Sea are greatly affected by prevailing internal waves in this area, the vertical directionality changes on vertical line array output were believed to be results of interaction between sound and ocean environment. The measured twin-peak beam patterns are not consistent with previous studies on flat or upslope bathymetry. Therefore, in this study the discrepancy is explained by modeling and correlation between the noise distributions and internal wave position.

2:45

4pUWb4. Acoustic spreading and boat noise within shallow Florida waterways. Richard Phillips (Dept. of Mech. and Aerosp. Eng., Univ. of Florida, Gainesville, FL 32611-6250), Christopher Niezrecki (Univ. of Massachusetts Lowell, Lowell, MA 01854-2881, Chris_Niezrecki@uml.edu), and Diedrich O. Beusse (Univ. of Florida, Gainesville, FL 32611-0126)

The West Indian manatee has become endangered partly because of watercraft collisions in Florida's coastal waterways. Several boater warning systems, based upon manatee vocalizations, have been proposed to reduce the number of collisions. One aspect of the feasibility of an acoustically based system will rely upon the distance in which a manatee vocalization can be detected. The magnitude of environmental noise and manatee vocalizations, as well as the acoustic spreading properties of the habitat will help to estimate the detection range of a manatee. This study surveyed several shallow-water coastal areas in Florida (Crystal River, Cedar Key, and Indian River) which are likely to be inhabited by manatees. Using a chirp signal (1–10 kHz) broadcast by an underwater transducer, it was observed that the acoustic pressure data collected was best represented by the mixed and Lurton spreading models. The overall data obtained from passing a boat directly over the hydrophone was most closely represented by a spherical spreading model. However, for a boat that is approaching, a mixed spreading model is the most appropriate. The mean SPL for boat traffic was measured to be 140 dB and the background environment noise level ranged between 69 and 105 dB.

3:00

4pUWb5. One year of background underwater sound levels in Haro Strait, Puget Sound. Val Veirs (Phys. Dept., Colorado College, Colorado Springs, CO 80903, vveirs@coloradocollege.edu) and Scott Veirs (BeamReach Sustainablity Sci. School, Seattle, WA 98115)

Haro Strait, on the west side of San Juan Island, WA, is the home range of the Southern Resident orca whales, a major shipping lane to and from Canada, and a center of private and commercial boating, especially in the summer. Four ITC hydrophones in a near-shore fixed array are used here to localize the underwater vocalizations of Southern Resident orca whales. The system operates 24 hours a day and has a frequency response of 100 Hz 10 kHz. Background sound levels are automatically characterized by half-hour reports that include: statistics and graphics based on mean sound levels (2-min running arithmetic mean pressure); a histogram of mean sound levels binned by frequency; and 2-s sound samples from maximum background events. Sound levels range from ~90 dB re 1 microPa (quiet conditions) to ~130 dB re 1 microPa when loud commercial ships are passing in the nearby shipping lane or speedboats are passing

close to the hydrophone array. Complete results for one year of continuous monitoring will be presented, segmented by time (season, day of the week, hour in the day), frequency spectrum and dominant noise source class. [Work supported by 35 undergraduate researchers and the Colorado College.]

3:15

4pUWb6. An acoustic modeling study of airgun noise from seismic surveys performed offshore British Columbia. Alexander O. MacGillivray (JASCO Res. Ltd., 2101-4464 Markham St., Victoria, BC, Canada V8Z 7X8) and N. Ross Chapman (Univ. of Victoria, Victoria, BC, Canada V8W 3P6)

A recent numerical modeling study has examined the propagation of underwater noise from potential seismic survey activity in Hecate Strait and Queen Charlotte Sound, Canada. Noise level predictions from this study are based on an integrated modeling approach incorporating an airgun array source model, a broadband transmission loss model, and an environmental model based on high resolution bathymetry, historical CTD casts and geophysical data. Details of the source model, transmission loss model and environmental model are discussed. Selected noise level predictions from the modeling study are presented. Results from this study will aid in evaluating potential environmental impacts of seismic exploration activity on marine ecosystems in British Columbia's offshore region. [Work supported by BC MEM and NSERC.]

3:30

4pUWb7. An integrated acoustic modeling infrastructure for underwater noise impact assessment. David E. Hannay and Roberto G. Racca (JASCO Res. Ltd., 2101-4464 Markham St., Victoria, BC, Canada V8Z 7X8, dave@jasco.com)

When industrial activities such as underwater dredging or seismo-acoustic surveying are planned near ecologically sensitive areas, effective forecasting of the associated sound levels is often necessary to allow effective noise management planning and to satisfy permitting requirements. To this end a comprehensive software application, the Marine Operations Noise Model, was developed to enable the estimation of aggregate noise levels from complex operations involving numerous activities over extended regions. The softwares architecture includes a specially adapted Parabolic Equation propagation model, a database of field measured spectral source levels from a wide range of industrial vessels, a run module that coordinates modelling and summing of sound from multiple sources, and a GIS interface for the definition of operational layouts and the display of noise level contours on area maps. This application has been used in the planning and regulatory approval process for future pipeline and offshore platform installation activities to take place in the proximity of marine mammal habitats. Its accuracy in this context was validated through an extensive program of acoustic monitoring of similar activities taking place in a nearby location over the course of a prior construction season.