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The role of principal component analysis in vowel acquisition research

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Session 4aAA**Architectural Acoustics, Noise, Psychological and Physiological Acoustics, Speech Communication and Committee on Standards: Implementation of Classroom Acoustics I**

Michael R. Stinson, Chair

*National Research Council, Institute for Microstructural Science Montreal Road, Ottawa, Ontario K1A 0R6, Canada***Chair's Introduction—7:40*****Invited Papers*****7:45****4aAA1. Forging improvements in classroom acoustical quality.** Angelo Bellomo (OEHS, Los Angeles Unified School District, Los Angeles, CA 90017)

Major investments are currently being made in school infrastructure. In Los Angeles alone, 90 new schools are to be constructed in the next 10 to 15 years, and hundreds of existing schools will undergo major renovation. Similar efforts are underway in large urban school districts throughout the country. This presents a unique opportunity to improve acoustical quality in classrooms for future generations of teachers and students. While some school districts are embracing this opportunity, there are substantial challenges to be faced before improvements in classroom acoustical quality are widely realized. Despite recent adoption of the ANSI standard, there are inconsistent views among school administrators and designers on the extent of the problem and the urgency to act. Additionally, there is insufficient data on the incremental costs of achieving improved acoustical quality in new construction or modernization projects. Meanwhile critical design and purchasing decisions are underway. There are promising examples of how school districts can successfully meet these challenges and forge improved acoustical quality in the learning environment. The focus of this discussion is on the use of policy reforms to drive better design and purchasing decisions, and how these decisions can be further influenced by community involvement and interagency collaboration.

8:05**4aAA2. Experiences implementing ANSI S12.60 in classrooms.** R. Kring Herbert and John Erdreich (Ostergaard Acoust. Assoc. 200 Executive Dr., W. Orange, NJ 07052)

Issues in meeting the ANSI 12.60 classroom acoustics standard fall into two categories: overcoming design team unwillingness to implement recommended criteria and finding practical methods of meeting issues of reverberation control and HVAC noise control. Solutions to HVAC noise control are emerging; an example is the use of fan coil units in new construction. Education of owners and designers is an ongoing process. It is hoped this paper contributes to a dialogue between the parties involved in school construction.

8:25**4aAA3. Source attenuating HVAC equipment—Just the facts.** Arthur Hallstrom (Trane, American Standard, 1515 Mercer Rd., Lexington, KY 40511)

Current classroom designs range from NC 63 to NC 25. The ASA/ANSI 12.60 Classroom Standard sets a relatively low background sound requirement for classrooms. Achieving 12.60 levels economically requires the maximum in source sound attenuation. This paper will provide a manufacturer's perspective on current and low noise HVAC products and the resulting classroom sound levels. Predictive acoustics software will be used to predict the space sound power level and some indication of equipment cost will be provided in order to assist designers in what is the best approach to low noise classroom design.

8:45**4aAA4. Case study of a prototype elementary school designed to meet the ansi classroom acoustics standard.** Gary Siebein, Robert Lilkendey, Hyun Paek, and Stephen Skorski (Siebein Assoc., Inc., 625 NW 60th St., Ste. C, Gainesville, FL)

The school board standards group that prepares design criteria for a large district, an architectural firm specializing in educational environments, a contractor and an acoustical consultant formed a team to develop a series of bid alternates for a prototype elementary school to upgrade the acoustical performance of the school to meet the ANSI classroom acoustics standard. Three primary areas of work included room finishes to reduce reverberation times to 0.60 s or less; selecting wall assemblies and details to reach the STC 50 required in the standard to reduce sound transmission between rooms; and HVAC system design to reduce noise produced by the air-conditioning system to the 35 dBA required in the standard. Only minor adjustments had to be made to the finish materials and wall assemblies in the prototype school design to meet the requirements of the standard. Major changes in design concept and implementation had to be made to the HVAC system design to meet the 35 dBA requirement. The paper describes the prototype design, the design processes used to develop the modifications and the cost impacts of the proposed changes.

4aAA5. Ten years of classroom acoustical design. Angelo J. Campanella (Campanella Assoc., 3201 Ridgewood Dr., Columbus (Hilliard), OH 43026)

Bradley found [J. Acoust. Soc. Am. **80**, 837–845] with word score tests that S/N and room reverberation time (RT) affect speech intelligibility, and that computed U50 [J. Acoust. Soc. Am. **80**, 846–854] was maximum for rooms having RT<0.6 sec and quieter than 35 dBA. This was applied to teleconferencing rooms [J. Acoust. Soc. Am. **111**, 2411(A)], and classrooms for distance learning [J. Acoust. Soc. Am. **107**, 2861(A)]. Economic strategies to reduce classroom RT and HVAC noise were developed. RT from 0.5 to 0.7 s and NC less than 30 are economically achieved: Absorption is provided by acoustical ceiling tiles of NRC>0.6. Carpets help. Chalk and cork boards provide some wall absorption, while in-use posters and book storage add more later. Central air handling systems promote NC-25 HVAC design. Noisy supply (VAV) air boxes are placed in the corridor ceiling. The transmitting manifold is lined. A flexible duct link to diffusers provides more noise attenuation at no added cost. Larger supply air diffusers satisfy NC-25. Common ceiling return air plenums compromise noise isolation between classrooms. High CAC ceiling tiles and return air openings near the room center provide NIC>40 versus recommended NIC50. Sketches and some cost estimates are provided.

4aAA6. Using computer modeling and auralization to provide classroom listening experiences. Robert C. Coffeen (School of Architecture and Urban Design, Marvin Hall, The Univ. of Kansas, Lawrence, KS 66045)

Computer modeling and computer auralization will produce comparative listening experiences for architects, building owners, and building users based on selected interior surface shapes, interior finish materials, and ambient noise levels. Will these techniques provide easily understood audible information that will assist classroom designers, school administrators, and teachers in making proper assessments regarding classroom acoustics? Auralizations for both good and bad classrooms, based on ANSI S12.60 2002, will be presented for comparative listening.

4aAA7. Acoustical renovation of portable classrooms for cochlear implanted pupils. David Lubman (14301 Middletown Ln., Westminster, CA 92683, dlubman@ix.netcom.com) and Tim Lopez (Riverside County Office of Education, Riverside, CA 92502)

Anticipating increased enrollment of children with cochlear implants, the Riverside County Office of Education undertook the acoustical renovation of two portable classrooms in California's Riverside County: Wildomar Elementary School in Wildomar and Katherine Finchy Elementary School in Palm Springs. The aim was to improve the acoustical environment to make the classrooms suitable for pupils with severe hearing disabilities (reduce noise and reverberation). Lacking an acoustical standard specific for cochlear implanted pupils, the acoustical goals chosen were those of the ANSI acoustical standard S12.60-2002. Key changes were intended to reduce interior noise from HVAC and improve the facade insulation to reduce exterior noise intrusion. The resulting acoustical improvements are documented. Costs and lessons learned are discussed.

4aAA8. Classroom acoustics in a Leadership in Energy and Environmental Design (LEED) certified school. Felicia M. Doggett (Metropolitan Acoust., LLC 40 W. Evergreen Ave., Ste. 108, Philadelphia, PA 19118, felicia@metropolitanacoustics.com)

Clearview Elementary School in Hanover, Pennsylvania, is the first school in that state to achieve LEED (Leadership in Energy and Environmental Design) certification. Additionally, it is only one of three K–12 schools nationwide to achieve certification at the Gold Level. A study of the classroom acoustics has been conducted in this school to evaluate how well a Gold Level LEED-certified building compares to criteria set forth in ANSI S12.60-2002. Also addressed is the success of implementing this standard in schools in the greater Philadelphia area and New Jersey, and the receptive attitude that architects in this region have for the standard. Both Philadelphia and the State of New Jersey are in the midst of a large effort to update old school facilities and build new ones.

4aAA9. Classroom acoustics and architects: Sightlines and sound. Terence J. Williams (Terence Williams Architect Inc., 102.2957 Jutland Rd., Victoria, BC V8T 5J9, Canada)

All too often, acoustics in classroom design is a forgotten or ignored subject. Design teams consider structural, mechanical and electrical implications but overlook the fundamental issue of hearing and being heard. As an architect with an interest in architectural acoustics, particularly performance arts venues, we have taken a pragmatic approach, some might say unscientific, to classroom design. A series of case studies is presented that illustrates the challenge of classroom acoustics from an architect's point of view: not "what does it look like," but "how does it sound"! Because classroom acoustics are so closely aligned to the visual presentation of information, seeing and being seen becomes critical also. Sightlines and sound continue to vie for attention. The challenge is making it work.

11:00

4aAA10. Classroom acoustics and ANSI S12.60: A British perspective. Peter A. Mapp (Peter Mapp Assoc., Copford, Colchester, Essex, UK)

At the same time that ANSI S12.60 was being developed, UK recommendations for classroom acoustics were also being drawn up (Building Bulletin BB 93). The resulting standards turned out to require similar goals and set very similar criteria (e.g., 0.6 s RT). The British document however, also set a criterion for intelligibility of 0.6 STI for open-plan classrooms and teaching areas. Apart from making some general comparisons between the standards, the paper looks at the practicality of measuring STI in classrooms. The practical problems of making measurements under occupied conditions as well as the influence that the sound source itself can have on the measurements are discussed. An empirical relationship between STI and the reverberation time of a classroom is established which may be of use to designers. The effects of source directivity on the resultant STI are discussed and measurement data are presented that compare different loudspeaker types including omnidirectional (dodecahedral array), single-cone units, and head/mouth simulators.

11:20

4aAA11. The acoustic design of schools in England and Wales. Bridget M. Shield (Dept. of Eng. Syst., Faculty of Eng., Sci. and Built Environ., London South Bank Univ., London SE1 0AA, UK, shieldbm@lsbu.ac.uk)

Legislation governing the acoustic design of schools was introduced in England and Wales in July 2003. The acoustic performance standards which must be achieved are published by the Department for Education and Skills (DfES) in "Building Bulletin 93: Acoustic Design of Schools." The UK government has recently announced plans for a major school buildings program; over 200 new secondary schools are to be built and many existing schools are to be refurbished. The DfES has published designs for "exemplar schools" by many of Britain's leading architects. However, current trends in architectural design such as the use of highly reflective materials and the introduction of large open plan spaces make it difficult to comply with the specified acoustic performance standards. There are also conflicts between the acoustic and other requirements, such as the desire for natural ventilation in classrooms. This paper outlines the regulations, presents data to show that many existing schools in England do not comply with previous or current standards, and gives examples to illustrate the effects of various acoustic treatments of classrooms. Issues concerning implementation of the regulations are discussed and examples of "good" and "bad" acoustic designs of schools are given.

11:40

4aAA12. A challenge to the Acoustical Society of America and its members to help accelerate adoption of the American National Standards Institute (ANSI) standard for classroom acoustics. Louis Sutherland (27803 Longhill Dr., Rancho Palos Verdes, CA 90275-3908), David Lubman (Westminster, CA 92683), and John Erdreich (Ostergaard Acoust. Assoc., West Orange, NJ 07052)

The ANSI Standard on classroom acoustics, S12.60-2002, has the potential to greatly improve the quality and accessibility of public education. To realize its potential the standard must be put to use. The standard is now voluntary unless it is adopted by a state or school district or is incorporated into individual school construction bidding documents. In some parts of the US, the ANSI standard now gets consideration for construction of new or substantially upgraded schools. But wide scale adoption may take many years at the current rate of progress. To accelerate adoption of the ANSI standard in school construction a major effort is required to educate stakeholders. ASA and its members can play an important role in this endeavor by educating stakeholders, including architects, school building officials, and parent-teacher groups through workshops, seminars and other presentations. Consistent with ASA's mission, to increase and diffuse the knowledge of acoustics and promote its practical application, ASA and its members can become ambassadors for good classroom acoustics in our communities. Some of the existing materials are described that can be used for this vital objective to help remove acoustical barriers to learning.

Session 4aAB

Animal Bioacoustics: Acoustic Monitoring of Large Whales--The Legacy of Paul Thompson I

John A. Hildebrand, Chair

Scripps Institute of Oceanography, University of California, San Diego, Mail Code A-005, La Jolla, California 92093-0205

Chair's Introduction—7:55

Invited Papers

8:00

4aAB1. Appreciated: Forty years with Paul O. Thompson. William C. Cummings (5948 Eton Ct., San Diego, CA 92122-3203, oshundbs@san.rr.com)

As longtime professional associate, ASA Fellow, and close personal friend, I have pleased in knowing Paul O. Thompson for 40 years. In 1965 he was first to join my newly established Applied Bioacoustics Branch at NEL. He authored or co-authored 30 papers and reports on the sounds and related behavior of marine mammals in the North and South Pacific and South Atlantic oceans. Educated at St. Olaf College, MN, and USC, Paul's earliest research resulted in 36 papers and reports on psychoacoustics often in collaboration with John Webster or Bob Gales. (Along with Past President Bob Young, we all were co-workers in the Listening Division headed by Past President Gales.) Ever unassuming, Paul is noted for stubborn devotion to detail that often uncovered new findings in seemingly rote data. Unmatched were his aural memory and skills at analog sound recording and processing. Paul could ferret and dissect whale signals among the ambient like very few contemporaries. Even after "retirement," his research in the Sea of Cortez markedly added to earlier findings of 20-Hz and other low frequency utterances from blue and fin whales. This presentation focuses both on Paul and his contributions that are sure to permanently mark the literature on marine mammal bioacoustics.

8:25

4aAB2. The study of whales with sound: the legacy of Paul Thompson. John A. Hildebrand (Scripps Inst. Oceanogr., Univ. Calif.—San Diego, La Jolla, CA 92093)

Passive acoustic monitoring of whales offers a complementary technique to traditional visual survey for assessment of whale populations. The development of whale acoustic monitoring techniques has proceeded over the last several decades. The work of Paul O. Thompson stands out for his pioneering contributions to the development of acoustic monitoring by: (1) identifying the variety of sounds produced by marine mammals and their mapping by species, (2) recognizing temporal patterns of sound production both on diel and seasonal cycles, and (3) demonstrating the value of long-term monitoring. The contributions of Paul Thompson will be discussed, and their connection to on-going work will be illustrated. The legacy of Paul Thompson's work is that acoustic population estimation techniques are now being developed that offer the potential for efficient and economical monitoring of whale populations.

8:50

4aAB3. Recording blue whales in the North Pacific: Building on the legacy of Paul Thompson. Kathleen M. Stafford (Nat'l. Marine Mammal Lab, 7600 Sand Point Way NE, Seattle, WA 98115)

All of the well-documented blue whale call types for the Pacific were first recorded and described by Paul Thompson. Early work by Thompson and colleagues W. C. Cummings and W. A. Friedl, in particular, laid the foundation for what we know today about blue whale sounds. Among these contributions is the first published confirmation of blue whale sounds recorded off Chile [W. C. Cummings and P. O. Thompson, *J. Acoust. Soc. Am.* **50**, 1193–1198 (1971)]. These sounds have since been recorded seasonally in the eastern tropical Pacific, suggesting a possible migration route. Thompson's early work with Naval data not only provided the first description of northeast and northwest Pacific calls, but also documented diel calling patterns [P. O. Thompson, *US Naval Elec. Lab. Rep.* (1965); P. O. Thompson and W. A. Friedl, *Cetology* **45**, 1–19 (1982); W. C. Cummings and P. O. Thompson, *J. Acoust. Soc. Am.* **95**, 2853 (1994); P. O. Thompson *et al.*, *Marine Mammal Sci.* **12**, 288–292 (1987)]. Based on these descriptions, long-term seasonal and geographic studies of blue whale calls have been used to examine the Pacific basin-wide occurrence of calling blue whales. Lastly, Thompson and colleagues first remarked that blue whale sounds show distinct geographic variation, an observation that has promoted acoustics as a tool for distinguishing among populations of blue whales worldwide.

9:15

4aAB4. A comparison of optimized methods for the detection of blue whale sounds. David K. Mellinger, Sara Heimlich, and Sharon Nieuwkerk (Cooperative Inst. for Marine Resources Studies, Oregon State Univ., 2030 SE Marine Sci. Dr., Newport, OR 97365, david.mellinger@oregonstate.edu)

Paul Thompson and colleagues published one of the first long-term studies of mysticete sounds [Thompson and Friedl, *Cetology* **45**, 1–19 (1982)]. Thompson analyzed sounds manually, finding and tallying vocalizations to arrive at a view of seasonal occurrence. Today the detection and counting tasks are often done by computer, using various methods for pattern recognition. Here we examine and compare three such methods for detecting the sounds blue whales: matched filtering, which may work well because of the

stereotypy of blue whale vocalizations; spectrogram correlation, which may work well for the same reason and also because of the noise removal that can be done with it; and a neural network, which has worked well in other contexts for detecting right whale calls. The methods are configured using optimization procedures specialized for each method, and the results are compared for vocalizations recorded at different signal-to-noise ratios. The optimized detectors are applied to SOSUS data to detect sounds characteristic of blues whales in the northeast Pacific.

9:40

4aAB5. Blue and fin whales in the Southern California Bight: Using long-term acoustic records and acoustic recording tags.

Erin M. Oleson (Scripps Inst. Oceanogr., Univ. Calif. San Diego, La Jolla, CA 92093), John Calambokidis (Cascadia Res. Collective, Olympia, WA 98501), Sean M. Wiggins, and John A. Hildebrand (Univ. Calif. San Diego, La Jolla, CA 92093)

Blue and fin whales have been the subject of acoustic investigation for decades, but recent advances in technology, including acoustic tags, autonomous recorders, computer automated signal detectors, and genetic sex-identification allow us to examine fine scale patterns in acoustic behavior. Northeast Pacific blue whales are known to produce at least five separate call types, and North Pacific fin whales at least three. Since August 2000, we have been acoustically monitoring blue and fin whales in the Southern California Bight using acoustic recording packages (ARPs), acoustic recording tags, and through genetic sampling of vocal whales. There are patterns of occurrence of particular call types with nonacoustic behaviors, and sex-biases exist in the production of certain calls. Acoustic sampling from several sites in the region indicates there are significant changes in the detection of various call types both within and between years, and among sites. Hourly detection patterns also vary, potentially serving as an indicator of feeding activity. An understanding of the temporal and spatial patterns of occurrence of individual call types, and the behavior associated with each, will lead to more efficient methods to monitor habitat, distribution, and abundance of these species.

10:05

4aAB6. Remembrances of Paul O. Thompson at the Naval Ocean Systems Center. Sue E. Moore (Natl. Marine Mammal Lab, 7600 Sand Point Way NE, Seattle, WA 98115) and Sam Ridgway (Naval Command, Control, and Ocean Surveillance Ctr., RDT&E Div., San Diego, CA 92152)

Paul Thompson spent 35 years working at the Naval Electronics Laboratory (later the Naval Ocean Systems Center) in San Diego and Hawaii. His research included psychoacoustics and animal bioacoustics. He authored 66 papers and reports, gaining the distinction of Fellow of the ASA. Paul was noted for painstaking data analysis, excellent aural memory, and outstanding skills at sound recording and processing. We present remembrances of Paul as a scientist, colleague, and friend during his years studying marine mammals for the Navy.

10:30–10:40 Break

Contributed Papers

10:40

4aAB7. Estimates of the percentage of sperm whales missed on combined visual and acoustic surveys in the eastern Pacific Ocean.

Jay Barlow and Shannon Rankin (Southwest Fisheries Sci. Ctr., Natl. Marine Fisheries Service, NOAA, 8604 La Jolla Shores Dr., La Jolla, CA 92037, jay.barlow@noaa.gov)

Abundance estimation from line-transect surveys commonly assumes that all animals on the trackline are detected. For long-diving species, such as sperm whales, this assumption is commonly violated. Marine mammal surveys were conducted in the eastern tropical Pacific in 2000–2002, with visual observers searching by 25× binoculars and acousticians aurally monitoring two elements of a towed hydrophone array. Visual and acoustic teams were independent and did not learn of detections made by the other team until after the animals had passed abeam. Data from 186 visual and acoustic detections were used to estimate the fraction of whales missed by each method. Many of the 178 acoustic detections were at greater distances than could be seen by the visual observers. Based on Palkas' direct-duplicate method, the percentages of missed trackline groups were 38% and 21%, respectively, for visual and acoustic methods. Assuming these two detection methods are independent, the percentage missed by both groups is 8%. Acoustic detections greatly reduce the fraction of missed animals, but by itself this method cannot reliably estimate group sizes and still misses some groups. A combination of acoustic and visual methods is recommended for sperm whale surveys.

10:55

4aAB8. Effectiveness of acoustic localization in documenting spatial and temporal patterns in autumn migration of bowhead whales in the Alaskan Beaufort sea.

Susanna B. Blackwell, Robert G. Norman, Charles R. Greene, Jr., Miles W. McLennan (Greeneridge Sci., Inc., 1411 Firestone Rd., Goleta, CA 93117, susanna@greeneridge.com), Trent L. McDonald (WEST Inc., Cheyenne, WY 82001), and W. John Richardson (LGL Ltd., King City, ON L7B 1A6, Canada)

During September 2001, 2002 and 2003, westbound bowhead whales were localized acoustically using directional autonomous seafloor acoustic recorders (DASARs). Each year, ~10 500–45 600 calls were detected over 23–36 days by an array of 11 DASARs deployed 6.5–22 km north-east of the oil-production island Northstar. Locations of origin for the 29% (2001) and 75% (2002) of the calls detected by two or more DASARs were determined by triangulation. Peak call detection rates occurred near 20 September in 2002 and 2003, but early in the month in 2001. In 2003, peak call detection rates exceeded 550 calls/hour. Upsweeps, downsweeps and constant-frequency calls made up 66%–68% of the calls each year. The migration corridor was similar in 2001 and 2002, but closer to shore by ~10 km in 2003. Comparisons with yearly aerial censuses by the Minerals Management Service showed that the two methods both documented migration timing and the offshore distance of the migration corridor. Aerial surveys covered a larger area and continued after onset of freeze-up. The acoustic method revealed more temporal and spatial details because it operated continuously over long periods independent of weather and darkness, and provided far more detections of whales. [Work supported by BP.]

4aAB9. Acoustic localization of bowhead whales near a Beaufort Sea oil development, 2001–2003: Evidence of deflection at high-noise times? W. John Richardson (LGL Ltd., Environ. Res. Assoc., P.O. Box 280, King City, ON L7B 1A6, Canada), Trent L. McDonald (WEST Inc., Cheyenne, WY 82001), Charles R. Greene, Jr., Susanna B. Blackwell (Greeneridge Sci. Inc., Goleta, CA 93117), and Bill Streever (BP Exploration (Alaska) Inc., Anchorage, AK 99519)

Most bowhead whales migrating west along Alaska's north coast during autumn pass 10–75 km seaward of BPs Northstar oil production island. Our objective was to determine the extent to which underwater sound from Northstar deflected the closest whales farther offshore. Whale calls were localized, via 11 DIFAR-based seafloor recorders, on 23–36 days per year while underwater sound was measured continuously 450 m from Northstar. Totals of 1121, 2057 and 14 304 calls were localized within a 20×26 km analysis area seaward of Northstar. Quantile regression was used to relate the 5th quantile distance-from-shore to underwater sound. A permutation method allowed for lack of independence of closely spaced calls. In 2001, the closest calling bowheads tended to be slightly farther offshore at times with high industrial sound, which were mainly due to vessels near Northstar. The displacement effect was maximal if sound was averaged over 50–80 minutes preceding calls. In 2002, high sound levels were rare and results were inconclusive. In 2003, offshore displacement was not significant despite high sample size and a migration corridor unusually close to shore. The upper 90% confidence bound on the percentage of bowheads displaced 2+ km did not exceed 1.6% in any year. [Work supported by BP.]

4aAB10. Acoustic surveys of killer whales during winter along the Pacific Northwest Coast. Thomas Norris (Sci. Application Intl., San Diego, CA), Brad Hanson, Dawn Noren, and Linda Jones (Northwest Fisheries Sci. Ctr., Seattle, WA 98112-2097)

An acoustic and visual survey was conducted along the outer coast of the Pacific Northwest (Washington to Central California) between 26 February and 14 March 2004 to determine the winter-time distribution and behavior of the southern resident ecotype population of killer whales (SRKWs). SRKWs are vocally active with a vocal repertoire that is well documented. A towed hydrophone array was deployed on 17 of the 20-day cruise resulting in an average of approximately 19 h of acoustic effort per day. Eleven of the 49 unique detections of marine mammal made were attributed to killer whales (22.5% of the total). Three of these killer whale detections occurred at night. Two of the three known SRKW pods (J, K, and L) were detected acoustically and positively identified as the J and L pods. Interestingly, the remaining nine unique acoustic encounters with killer whales were of either the offshore form or the transient form, considered to be vocally less active relative to the resident form. The hydrophone array substantially increased the number of detections of killer whales that would have been made using visual methods alone. Determining estimates of group size using acoustics remains problematic. Planned improvements to the acoustic system will be discussed. [Work funded by NOAA/NWFSC-Seattle, WA.]

4aAB11. Range-depth tracking of humpback whales using autonomous acoustic recorders. Aaron M. Thode, Peter Gerstoft, Melania Guerra, Dale M. Stokes (Marine Physical Lab., Scripps Inst. of Oceanogr., UCSD, La Jolla, CA 92093), William C. Burgess (Greeneridge Sci., Inc., Goleta CA 93117), Michael J. Noad (Univ. of Queensland, St. Lucia, Queensland, Australia), and Douglas H. Cato (Defence Sci. and Technol. Organization, Sydney, NSW, Australia)

Matched-field processing (MFP) is a technique for tracking an acoustic source in range and depth by comparing the output of an ocean acoustic propagation model with measured acoustic data collected across multiple hydrophones [e.g., Thode *et al.*, J. Acoust. Soc. Am. **107**, 1286–1300 (2000)]. In October 2003 a MFP experiment was conducted using humpback whale sounds recorded during the spring migration off the Sunshine Coast in Queensland, in conjunction with a larger experiment conducted by the Humpback Acoustic Research Collaboration (HARC). Humpback whale sounds between 50 Hz to 1 kHz were recorded on a five-element vertical array deployed in 24-m-deep water. The vertical array consisted of a set of flash-memory autonomous recorders attached to rope with an anchor at one end, and a subsurface float at the other. Acoustic data were simultaneously collected and monitored on five sonobuoys deployed over approximately 2-km range. Using the range estimates from the sonobuoys as bounds on the matched-field processing, a geoacoustic inversion using the calls was performed on the vertical array data using a genetic algorithm. Preliminary results of the inversion and resultant 3-D position fixes are presented. [Work supported by ONR.]

4aAB12. Model-based tracking of marine mammals. Ahmad T. Abawi, Paul Hursky, Michael B. Porter (Ctr. for Ocean Res., Sci. Applications Intl. Corp., San Diego, CA), John Hildebrand, Sean Wiggins, and Allan Sauter (Scripps Inst. of Oceanogr., La Jolla, CA)

Data from the August 2003 experiment conducted by the Scripps Institution of Oceanography in the Southern California Offshore Range (SCORE) are used to track marine mammals. SCORE is a naval training area near the island of San Clemente located in relatively shallow water. The water depth where the experiment was conducted is around 360 meters. Data were recorded on a 100-m, eight-element vertical line array (VLA) deployed from the floating instrument platform (FLIP) and four bottom-mounted hydrophones deployed in an area covering approximately 3 square kilometers. During the course of the 7-day experiment continuous recording of the ocean environment was made. The recordings contain calls from various marine mammals, particularly blue and fin whales. Data recorded on the bottom-mounted hydrophones and the VLA are used to track singing marine mammals using two entirely different model-based tracking techniques: The animals are tracked by comparing the predicted (using a propagation model) and measured time difference of arrival recorded in pairs of bottom-mounted hydrophones. Additionally, matched-field processing on the VLA data is used to estimate the range and depth of the animals.

Session 4aEA

Engineering Acoustics: Transduction, Radiation, Ultrasonics, General Engineering Acoustics

Kim C. Benjamin, Chair

Naval Undersea Warfare Center, Newport, Rhode Island 02840

Chair's Introduction—8:25

Contributed Papers

8:30

4aEA1. Single-crystal ferroelectrics for sonar devices. E. A. McLaughlin and H. C. Robinson (NUWC-UTMR, Code 15232, 1176 Howell St., Newport, RI 02841)

Single-crystal ferroelectrics or piezocrystals were recently introduced into the electroactive materials community. The 33-mode electromechanical coupling factor of piezocrystals is typically greater than 0.90, which is significantly larger than typical values for piezoelectric ceramics (0.62–0.74). For sonar projector applications this large, k_{33} has been responsible for more than doubling the bandwidth of active sonar arrays over what is currently achievable with ceramics. More recently, a crystal grower produced a cut of lead magnesium niobate-lead titanate (PMN-PT) single crystal with piezoelectric shear coefficient values of 7000 pm/V and shear coupling factors of 0.97. (For PZT5H, d_{15} is 730 pm/V.) This piezocrystal d_{15} coefficient implies significantly improved sensitivity and signal-to-noise ratio for accelerometers and hydrophones, while the high coupling promises bandwidth increases greater than those realized in 33-mode projectors using piezocrystals. The 31-mode is also being investigated. By measuring the response of the materials to high- and low-level electrical bias and excitation fields, mechanical prestresses, over temperature, the material's effective material properties as a function of these operational variables have been determined. [This work sponsored by ONR and NUWC ILIR.]

8:45

4aEA2. Acoustic radiation from arrays on coated prolate spheroids. Jeffrey E. Boisvert (NAVSEA Newport, Newport, RI 02841, boisvertje@npt.nuwc.navy.mil) and A. L. Van Buren (Middletown, RI 02842)

An important aspect in the design of conformal sonar arrays is the effect of an outer coating on the array response. The design of the coating is subject to the competing requirements of acoustic transparency, toughness for impacts, and flow noise rejection. The present study considers an array of rectangular pistons, conformal to a rigid prolate spheroidal baffle, covered by an outer coating. The coating is modeled as a fluid, i.e., it is assumed to have a vanishingly small value of the shear modulus, typical of many elastomers. Unlike a coating on a spherical or cylindrical baffle, where a uniform thickness coating can be defined by simply increasing the value of the radial coordinate, the corresponding operation in prolate spheroidal coordinates results in a confocal layer, i.e., a layer that is not of constant thickness. In this paper a new method is described that allows for a uniform thickness coating in prolate spheroidal geometry. This method employs the spheroidal addition theorem to satisfy the boundary conditions. Comparisons of the beamformed array response for a confocal coating versus a uniform thickness coating are given for a sample array with various coating properties. [Work supported by the NUWC ILIR Program.]

9:00

4aEA3. A fast time-domain method for calculating the near field pressure generated by a pulsed circular piston. James Kelly (Dept. of Math., Michigan State Univ., East Lansing, MI 48824) and Robert McGough (Michigan State Univ., East Lansing, MI 48824)

A fast single-integral method has been derived for calculations of transient near field pressures generated by circular pistons. This new time-domain method outperforms other methods with respect to speed, accuracy, and ease of implementation. Unlike methods that apply the spatial impulse response, the new method inherently eliminates aliasing problems, which permits a significant reduction in the temporal sampling rate. As a result, the new method is approximately six times faster than the spatial impulse response for a 1% maximum numerical error. This new method converges exponentially with Gauss quadrature, whereas spatial impulse response methods evaluated with fast Fourier transforms only converge linearly with respect to sampling frequency. Thus, the new method achieves small numerical errors in very short times. Since the new fast method evaluates a single-integral expression that is readily computed with standard quadrature techniques, the implementation is greatly simplified relative to the spatial impulse response. The new method, which achieves smaller errors in less time with lower sampling rates, is also much more memory-efficient than the impulse response approach. Each of these advantages represents a significant improvement with respect to the impulse response method for computations of transient acoustic fields.

9:15

4aEA4. New ultrasonic ranging transducer using 200-KHz corrugated polyvinylidene fluoride film. Minoru Toda and Jerry Dahl (Measurement Specialties Inc., 460 E. Swedesford Rd., Ste. 3005, Wayne, PA 19087)

Corrugated PVDF transducers were originally developed for use in parametric arrays to demodulate audio sound in air from modulated high intensity ultrasound. These arrays were designed to operate at ~40 KHz. PVDF has recently been used in low-cost short-range-distance ranging transducers and provides accurate distance measurements in consumer, industrial, and medical applications. For this purpose, we designed, fabricated, and tested a corrugated PVDF transducer that operates at 200 KHz. This results in a small, directional, and low-cost transducer. The transducer has an area of ~1 cm² and produces a beam width of 16°. Combined with a sensor circuit that converts the measured time of flight into distance, we have demonstrated a detection range of 17–500 mm with a resolution of 25 mm. The resonance frequency of curved PVDF film was increased to 200 KHz by reducing the curvature radius to $\frac{1}{5}$ (~1 mm) of that for previous devices operating at 40 KHz. The concave and convex regions of the corrugation vibrate with opposite phase and the radiated ultrasonic waves from both regions have a half-wavelength path difference that cancels the opposite vibration phase due to the half-wavelength effect.

4aEA5. Investigation of vibration of polyvinylidene fluoride cylindrical film by use of acoustic probe. Minoru Toda (Measurement Specialties Inc., 460 E. Swedesford Rd., Ste. 3005, Wayne, PA 19087)

Ultrasonic pen and white board digitizing systems are a rapidly growing market for PVDF transducers, utilizing an ultrasonic transmitter in the pen tip and two stationary ultrasonic receivers. The transmitter is constructed of piezoelectric PVDF welded into a cylindrical shape. Radius vibration (breathing mode) creates an 80-kHz airborne acoustic wave. The initial cycle is used to determine the time of flight, used to calculate the pen position by triangulation. An ideal cylindrical transmitter would have a uniform 360 horizontal radiation pattern. However, practical devices have a 65% to 85% minimum/maximum ratio for the initial pressure peak. A novel acoustic probe was devised to investigate this nonuniformity. The probe tip is a needle with 0.5-mm through hole, and the acoustic wave propagates through the needle to a conventional reference microphone. The probe was positioned using a precision XYZ stage. Investigations with this probe revealed the details of the nonuniformity in horizontal radiated pressure. It was found that (1) vibration at the welded seam of the PVDF cylinder is a maximum, opposite the expected behavior, and (2) the pressure distribution profiled along the height of the cylinder has a central plateau and decreases linearly at the cylinder ends.

9:45–10:00 Break

10:00

4aEA6. Lamb waves in bulk acoustic wave resonators: Analysis of spurious resonances and design of resonators in the UHF-VHF frequency range. Alexandre Volatier, Gregory Carruyer (ST Microelectronics, 38926 Crolles, France, volatier@chartreuse.cea.fr), Emmanuel Defay (LETI-CEA, Grenoble, France), and Bertrand Dubus (UMR CNRS 8520, Lille, France)

Bulk acoustic wave (BAW) resonators exhibit attractive properties in terms of power handling capacity and on-chip integration to realize filters in the GHz range [K. M. Lakin, IEEE Ultrason. Symp. pp. 895–905 (1999)]. In a BAW resonator, a thin piezoelectric layer (a few mm) deposited between two electrodes is driven in its thickness extensional mode of vibration. To get a high quality factor, this structure is decoupled from the substrate by a multilayer Bragg reflector or a back-etched membrane. A current problem in the design of BAW resonators is the existence of spurious resonances close to the thickness extensional mode which generate ripple in the filter passband. In this paper, these spurious modes are analyzed in terms of Lamb waves resonances. Physical modeling of BAW resonators using finite element ATILA code is presented. The influence of lateral dimensions and electrode geometry on spurious resonances is emphasized. With specific electrode design and electrical excitation, it is demonstrated that lateral modes of Lamb waves can be used to realize resonators in the 50–250-MHz range. Experiments on an AlN piezoelectric layer between Pt electrodes on a SiN membrane are presented. [Work supported by a ST Microelectronics grant (CIFRE).]

10:15

4aEA7. Ultrasonic nondestructive inspection of interface defects in anisotropic fiber-metal-laminates. Stefan A. L. Stijlen (Dept. of Imaging Sci. and Technol., Delft Univ. of Technol., Lorentzweg 1, 2628 CJ Delft, The Netherlands, s.a.l.stijlen@tnw.tudelft.nl) and Maarten C. M. Bakker (Delft Univ. of Technol., 2629 HS Delft, The Netherlands)

Ultrasonic inspection is an essential tool for the quality control of the fiber-metal-laminate Glare, which is currently applied in the production of the huge A380 Airbus aeroplane. Nowadays, the ultrasonic inspection of Glare is performed by a transmission C-scan, traditionally, or by a more advanced pulse-echo scan. In this research, a new method of data acquisition and processing is investigated. The data acquisition consists of reflection measurements for variable source-receiver offsets. The first objective is to image delaminations at or nearby interfaces. These defects can, for instance, be associated with kissing bonds due to locally imperfect

coupled interfaces. The algorithm which processes and converts the data into an image is based upon the inverse scattering theory. From the acquisition level in water to the interface levels in Glare, the elastodynamic Green functions are computed including strong anisotropy, multiple reflections and conversions, refractions, and surface waves. Subsequently, interface defects are modeled as additional contrasts in compliance. Finally, an image is obtained through their optimization by minimizing a cost-functional which is the mismatch between the measured data and the synthetic data generated with the Green functions. [Work supported by STW, the Dutch technology foundation.]

10:30

4aEA8. Experimental equalization of a one-dimensional sound field using energy density and a parametric equalizer. Micah Shepherd, Xi Chen, Timothy Leishman, and Scott Sommerfeldt (Acoust. Res. Group, Dept. of Phys. and Astron., Brigham Young Univ., Provo, UT 84602, mrs74@email.byu.edu)

A simple experimental method has been developed to equalize a one-dimensional sound field using acoustic energy density. Energy density is estimated using two methods: the two-microphone transfer-function method and the two-microphone finite-difference method. An equalization filter is manually configured for each using a digital parametric equalizer. The results of the two estimation approaches are compared. Equalization filters are similarly implemented for sound pressure measured at discrete points in the field and for spatially averaged sound pressure. Results of the approaches are compared and benefits of the energy density method are discussed.

10:45

4aEA9. Sound-quality analysis of sewing machines. James J. Chatterley, Andrew J. Boone, Jonathan D. Blotter (Mech. Eng. Dept., School of Eng. and Technol., Brigham Young Univ., 435 CTB, Provo, UT 84602), Scott D. Sommerfeldt, and Thomas L. Lago (Brigham Young Univ., Provo, UT 84602)

Sound quality has become an important factor in consumer product development. Sound quality analysis procedure and results for six sewing machines ranging from entry level to professional grade machines will be presented. The focus of the study was to determine what consumers feel constitutes a pleasant sounding sewing machine and what structural modifications can be made to produce those sounds. The procedure consisted of forming small (15 person) then large (50 person) jury-based tests. These tests consisted of listening to various sound bytes and ranking the sounds as well as indicating why sounds are appealing or unappealing. The sound bytes were from actual sewing machines as well as computer generated sounds and modified machine sounds, which were constructed such that the spectral features most important in achieving a desirable product from an acoustic perspective could be statistically determined. The procedures and analysis of the jury testing results will be presented and discussed. This paper presents sound localization scans, indicating machine sound hot spots and possible sources for undesired sounds. In conclusion proposed modifications to machine structure to alter machine sound signature into a more sensory pleasant sound will also be presented.

11:00

4aEA10. Production of a noise level database of power tools used in the construction industry. Charles S. HaydenII, Edward Zechmann, and Rohit Verma (Nat'l. Inst. For Occupational Safety and Health, 4676 Columbia Pky., C27, Cincinnati, OH 45226, chayden@cdc.gov)

This study focused its efforts on sound power levels and noise reduction of power tools used by workers at both commercial and residential construction sites. The objective was to develop a noise control technology database consisting of power hand tools used in the construction industry with respective sound power levels and workers' sound exposure level (SEL). The database and specific noise control applications information are made available to the public on a searchable web-site. Data includes a particular tool's wavfile and time series signatures to be used for future

qualitative and quantitative analysis by other acousticians accessing the database. With this information, reasonable assumptions can be made as to the potential occupational noise exposure expected for various occupations, tasks, etc. and as to potential sources of noise within the powered hand tool itself. The database captures tool types, tool models, tool manu-

facturers, and decibel power level. The database is expandable to include new tools and new manufacturers as the power tool population changes over time. These efforts support the reduction of noise induced hearing loss among construction workers by providing "buy quiet" and "designed quiet" information to power tool buyers and end users.

THURSDAY MORNING, 18 NOVEMBER 2004

PACIFIC SALON 3, 8:30 TO 11:45 A.M.

Session 4aMU

Musical Acoustics: Physics of Musical Instruments

George A. Bissinger, Cochair

Department of Physics, East Carolina University, Greenville, North Carolina 27858

Daniel O. Ludwigsen, Cochair

Science and Mathematics Departments, Kettering University, 1700 West Third Avenue, Flint, Michigan 48504-4898

Contributed Papers

8:30

4aMU1. Galilei experiments and common sense. R. Dean Ayers (Dept. of Phys. and Eng. [Adjunct], Southern Oregon Univ., 1250 Siskiyou Blvd., Ashland, OR 97520)

Galileo Galilei showed us by example how to put hypotheses about the natural world to the test of scientific experiments. It is very likely that his attitude of healthy skepticism toward his own ideas and those of others was instilled in him by some work in musical acoustics. His father, Vincenzo (or Vincentio), was a prominent music theorist and composer as well as an accomplished lutenist. During his adolescence Galileo assisted Vincenzo in practical studies on the behavior of vibrating strings. That research examined the influence of various parameters on the pitch of a string (and hence its frequency), disproving some incorrect guesses. We are now in an age when many of our students are unduly influenced by popularized reports on exotic and seemingly bizarre open questions in relativity, cosmology, and elementary particles. That exposure encourages them to abandon their common sense and skepticism when they take a science course. Our specialty can provide a valuable counterexample by letting students perform exploratory, quantitative studies on the strings of the original Galilei experiments or on other vibratory systems. We can all benefit by remembering the lessons learned from such work. [Work supported in part by the Paul S. Veneklasen Research Foundation.]

8:45

4aMU2. End correction for an open pipe from measured resonance frequencies. Daniel O. Ludwigsen and Brandon J. Dilworth (Dept. of Sci. and Math., Kettering Univ., 1700 W. Third Ave., Flint, MI 48504)

As briefly noted in freshman physics textbooks, resonance in open-ended pipes depends on an effective length that is greater than the actual length of the tube. The additional end correction varies with the termination at the open end. By measuring several peak frequencies in a response function (pressure in the pipe over source excitation), we have calculated the end correction for open ends of PVC tubes of varying diameters with several flanges of increasing size. Temperature effects were carefully controlled, and resonance frequencies recorded to within 0.01 Hz. Averaged results are in agreement with literature and theoretical expectations, but anomalies and unexpected results will also be presented.

9:00

4aMU3. Period doubling in free reeds coupled to pipe resonators. Evan M. Goetzman (Univ. of Minnesota, Minneapolis, MN 55455) and James P. Cottingham (Coe College, Cedar Rapids, IA 52402)

An earlier study investigated behavior of the reed-pipe combination consisting of an American organ reed installed at the closed end of a cylindrical pipe. Over a wide range of pipe lengths and playing pressures, the sounding frequency is slightly below a pipe resonance frequency and can be pulled considerably below the natural frequency of the reed. [Vines *et al.*, J. Acoust. Soc. Am. **114**, 2349 (2003)]. Additional measurements of reed vibration for these reed-pipe combinations for low frequency (48 Hz) free reeds show that as the playing pressure is increased there is typically a sudden transition to period doubling or, in some cases, tripling or quadrupling. Some pressure intervals exist in which apparently chaotic reed vibration occurs, or in which the reed will not vibrate at all. Period doubling was found to occur over varied values of pipe length, pipe diameter, and the use of two different reeds with matching resonance frequencies. General trends in the onset and steady-states of period doubling are described using measured spectra and spectrograms of both reed motion and radiated sound pressure over a continuous range of playing pressure. [Work supported by NSF REU Grant No. 0354058.]

9:15

4aMU4. The motion of harp strings. Chris Waltham (Dept. of Phys. and Astron., Univ. of British Columbia, Vancouver, BC V6T 1Z1, Canada, waltham@physics.ubc.ca)

A harp is an instrument with a set of plucked strings that excite the sound board directly, without the medium of a bridge. The strings are positioned at an acute angle to the plane of the sound board. The quality of the sound produced depends on the motion of the string and its interaction with the resonances of the sound board. The string motions of several small and large harps have been studied using small, fast position sensors. The results are compared to those of a simple nonlinear model based on the measured elastic properties of the string materials, and those of the sound board. The implications for the sound production are discussed.

4aMU5. Measurements of the acoustics of the kalimba. Daniel O. Ludwigsen (Dept. of Sci. and Math., Kettering Univ., 1700 W. Third Ave., Flint, MI 48504)

The kalimba is an idiophone (self-sounding instrument) of African origin, specifically a standardized type of mbira manufactured with modern facilities by Hugh Tracey. The sound production is somewhat similar to a small glockenspiel, but the free ends of the metal keys are plucked rather than simply-supported struck bars. The body of the kalimba, however, more resembles a guitar with a central sound hole in the top plate. There are additionally two smaller holes in the back plate, which a player can use to create vibrato effects with rapid covering and uncovering with fingertips. Spectral information and modal analysis are used to investigate the dynamics of sound production of the kalimba, and are compared with such measurements in the literature for guitar and mallet instruments. These measurements will inform future modeling efforts to better understand the dynamics of the instrument.

4aMU6. Inexpensive apparatus for studies of the lip reed. Mark T. McLaughlin, Lowell J. Eliason (Dept. of Phys. and Astron., California State Univ., Long Beach, 1250 Bellflower Blvd., Long Beach, CA 90840-3901), and R. Dean Ayers (Southern Oregon Univ., Ashland, OR 97520)

A circuit has been designed and built to provide real-time, precisely controlled video images of a brass player's lips during the relatively steady part of a sustained note. This should be useful for both the scientist who wishes to model the lip reed and the beginning player who is just learning how to control that driver. The circuit uses a phase-locked loop integrated circuit to track the signal from a microphone in the mouthpiece and subdivide the period for triggering a strobe light. The signal from the video camera is used for arming the trigger, so that there will be at most one strobe flash per field. This circuit is stable enough in tracking the playing frequency that a trombone slide can be moved continuously through its full range of positions, yet the live, strobed image advances through equally spaced fractions of a cycle at a rate determined by the settings of the circuit. This is important because the playing frequency is not perfectly constant, especially for beginners. [Work supported in part by the Paul S. Veneklasen Research Foundation and the CSULB Scholarly and Creative Activities Committee.]

4aMU7. Drumstick "Pitching"—A modal analysis approach. George Bissinger (Phys. Dept., East Carolina Univ., Greenville, NC 27858)

"Pitching" drumsticks by ear is too time intensive for a manufacturing process that generates many millions of drumsticks each year. Yet, it is a sought-after attribute that is worth implementing if some pitching technique can be made compatible with typical manufacturing rates. Drumstick pitch is normally determined by our evaluation of the sound produced by the nonharmonic vibrations when it is held and struck against a hard surface. Vibration analysis performed on drumsticks in a free-free support fixture provided normal mode results for over 100 solid wood and wood+plastic tip drumsticks. Modal analysis measurements up to 5 kHz were made parallel and perpendicular to the plane of the grain at 17 points along the length of a few drumsticks. A small sample of four drumsticks, pitch-matched by ear for comparison with these vibration analysis predictions, indicated a promising solution to the problem of real-time pitch matching. Even though the usual problems were encountered—orthotropic, small knots, kinked or twisted grain, density variations (dimensional consistency was within 0.1% but mass varied over 40%), and stiffness variations—the least sensitive to these problems was the lowest bending mode near 0.4 kHz, now the basis for automated pitch matching by a major manufacturer.

4aMU8. The glissando of colliding hard objects. Benjamin Frey and William M. Hartmann (Dept. of Phys. and Astron., Michigan State Univ., East Lansing, MI 48824)

When two hard objects are pressed together they alternately collide and rebound, leading to an audible, decaying sequence of pulses with a decreasing interpulse interval. Therefore, the sequence is heard as a rising pitch or glissando. Virtuoso performances of colliding hard objects (steel spheres) demonstrated a frequency rise that was accurately an exponential function of time over the span of more than three octaves. The amplitude decay also tended to be exponential, though less reproducibly so. Because successive decaying pulses were otherwise virtually identical functions of time, the spectra were harmonic. Four harmonics were clearly evident in spectrograms. Spectrograms also revealed a strong formant near 2000 Hz, presumably indicating the lowest mode of vibration of the steel spheres themselves. Formants had a Q of about 7. With the quantitative acoustical description well in hand, interest now centers on discovering the time-dependent force function employed by experienced performers of colliding hard objects when they achieve the most effective glissandos. [Work supported by the Michigan State University Professorial Assistant Program.]

4aMU9. Multiphonic model of split pressure-phase modes using the finite-element-method. Bader Rolf (Musikwissenschaftliches Institut, Neue Rabenstr. 13, 20354 Hamburg, Germany, r_bader@t-online.de)

The generation of multiphonic sounds in blown musical instruments is modeled by an eigenvalue finite-element-method (FEM) calculation. A common technique for the production of multiphonic tones is the use of complex sound-hole closing techniques, where more than one overtone spectrum appears, with fundamental frequencies being in inharmonic relations. Eigenfrequencies of a tube with both ends open are compared to eigenfrequencies of a tube with at least one sound hole open. It shows up that the tube with one open sound hole leads to additional eigenfrequencies in the tube for each node number. This is because of two possibilities of pressure curves within the tube for each node number, the pressure right and left of the hole being in phase or in opposite phase. The frequencies of equal phase form a new overtone structure which is harmonic within itself and so forms a new tone, a multiphonic. So the multiphonic model is in agreement with the Backus formula of multiphonic sounds as a combination of the tubes fundamental and multiples of the frequency difference of the zero node mode components.

4aMU10. Modeling the nonlinear string body coupled dynamics of bowed musical instruments. Octávio Inácio (Instituto Politécnico do Porto, Escola Superior de Música e das Artes do Espectáculo, Musical Acoust. Lab., R. da Alegria, 503, 4000-045 Porto, Portugal) and José Antunes (Instituto Tecnológico e Nuclear, 2686 Sacavém codex, Portugal)

Most theoretical papers on bowed-string instruments deal with isolated strings, pinned on fixed supports. In addition, the instrument body dynamics has not been accounted at all, or else by using extremely simplified models of the string/body interaction at the bridge. Such models have, nevertheless, been instrumental to the understanding of a very common and musically undesirable phenomenon—the "wolf note"—a strong beating interplay between string and body vibrations. Cellos, bad and good, are prone to this problem. In previous work we developed a modal method to deal with friction-excited strings, enabling effective simulations of such systems in which the strings were assumed decoupled from the instrument body. In the present paper our computational method is extended to incorporate the complex dynamics of real-life instrument bodies, coupled to the string motions. In this approach, the string is coupled with experimental body data—body modes or impulse response at the bridge. Our computa-

tional method is illustrated through extensive parametric computations performed on a bowed cello. These numerical simulations show some light on interesting and less-known features of wolf notes, in particular concerning the ranges of bowing-parameters leading to their emergence, as well as the dependence of the beating frequency on the playing.

11:00

4aMU11. Turbulence eddy viscosity in flutelike instruments. Rolf Bader (Musikwissenschaftliches Institut, Neue Rabenstr. 13, 20354 Hamburg, Germany, r_bader@t-online.de)

The generator area of a flutelike instrument is modeled by the finite-element-method (FEM) using a Navier–Stokes (NS) model and a k -epsilon turbulence model. A steady-state simulation with condition of zero pressure at the tube's end and a transient simulation using a back-impulse from the tube's end to the generator area was performed. Both models NS and k -epsilon agreed in the overall structure of the volume flow from the player's lips split in a part flowing into the tube and one flowing outside, common in eddy and vortex structure. But the k -epsilon model showed a much more realistic description of the relation of flows into and outside the tube. As the NS model results in a one-to-one split of the flow, the turbulence model leads to a much smaller flow part into the tube with most of the flow going outside and so to a much higher vorticity at the flute lip. The transient model also shows a sharpening of the transient pressure curve because of this sharpened lip near vorticity behavior. So as the turbulence model seems to be a good choice in formulating a discrete flute model, the overall eddy damping seems to play an important role in flute dynamics.

11:15

4aMU12. Separation of musical instrument timbre using performance information and real-time harmonic/inharmonic component filtering. Joseph A. Sarlo (Univ. of California, San Diego, 4991 Combe Way, San Diego, CA 92122, jsarlo@ucsd.edu)

Described here is a technique for the real-time separation of a musical instrument timbre from a multitimbre sound source. The separated timbre may be a timbral component of a single musical instrument timbre (e.g., the breath noise from a wind instrument) or an instrument timbre from a multi-instrument sound source. The separation is accomplished by first using performance information, such as pitch tracking or score alignment, to estimate a fundamental frequency for the source of the timbre to be separated. Real-time analysis/resynthesis filtering of the sinusoidal components based on their harmonic relationship to the estimated fundamental frequency is then employed to extract the desired timbre.

11:30

4aMU13. Synthesis of audio spectra using a diffraction model. V. Vijayakumar (FOSEE, Multimedi Univ., Jalan Air Keroh Lama, Melaka 75450, Malaysia, vijaya@mmu.edu.my) and C. Eswaran (Multimedi Univ., Cyberjaya, Selangor, Malaysia)

It is shown in this paper that the intensity variations of an audio signal in the frequency domain can be explained by using a Fresnel diffraction model. By proper choice of two parameters an input spectrum of discrete frequencies of unit intensity can thus be transformed into the known spectra of different musical instruments. Specific examples of musical instruments are considered for evaluating the performance of this method. It is found that this model yields musical spectra with a good degree of accuracy. Synthesized sounds of bells and drums are presented as real examples.

THURSDAY MORNING, 18 NOVEMBER 2004

ROYAL PALM SALON 1, 9:00 TO 11:15 A.M.

Session 4aNS

Noise and Committee on Standards: Active Noise Reduction for Hearing Protection: Technology Opportunities and Assessment of Effectiveness

William J. Murphy, Chair

National Institute of Occupational Safety and Health, 4676 Columbia Parkway, Cincinnati, Ohio 45226-1998

Chair's Introduction—9:00

Invited Papers

9:05

4aNS1. Active noise reduction hearing protectors; 50 years of development. Kenneth A. Cunefare (Woodruff School of Mech. Eng., Georgia Inst. of Technol., Atlanta, GA 30332-0405)

This paper will review the technology development of active noise reduction (ANR) hearing protectors. Such devices have been in development for close to 50 years. Early studies by Meeker on feedback implementation of ANR identified many of the issues and performance criteria that are still relevant today. These issues include compensation, minimum phase filter design, and out-of-band control stability. Numerous patents and publications since the early developments focused on means to address the issues discussed by Meeker, and, to introduce alternative implementations of ANR hearing protectors or headsets. Such alternatives include feed-forward designs, as well as open-back and closed-back designs (open-back designs rely solely upon active control for noise protection; closed-back designs include passive attenuation from an ear muff). Development has also proceeded on designs which include communication elements; here, the active control system must attenuate the noise, but not the communication. Fifty years on, ANR hearing protectors represent the most successful commercial application of active noise control technology.

4aNS2. Design and performance of hybrid feedforward-feedback active noise reduction for hearing protection and communication. Laura Ray (Thayer School of Eng., Dartmouth College, Hanover, NH 03755)

Since the 1990s research and development of digital feedforward active noise reduction (ANR) based on least-mean-square (LMS) algorithms has shown that significant gains in noise reduction can be achieved for circumaural hearing protectors. However, to-date digital feedforward hearing protection devices have not penetrated the market. This presentation reviews prior research in feedforward ANR for circumaural devices, along with the opportunities and challenges for commercialization of the technology. The effect of noise source characteristics, specifically stationarity and dynamic range, on stability and performance of LMS filters is examined. Research on hybrid ANR architectures combining LMS filters and digital feedback ANR is reviewed, showing that robust performance with stability margins required for a commercial product can be achieved with such systems. The role of ANR in enhancing speech intelligibility is outlined, along with research needed to develop standards for measurement of stability, ANR performance, and speech intelligibility afforded by digital feedforward ANR devices. The paper concludes with a presentation of opportunities offered by feedforward ANR in intra-aural and supra-aural hearing protection devices.

4aNS3. Development and performance of an active noise reduction earplug for use in extreme noise environments. Richard L. McKinley, John Allan Hall (Human Effectiveness, Air Force Res. Lab., 2610 Seventh St., Wright-Patterson Air Force Base, OH 45433-7901), Will Saunders, and Mike Vaudrey (Adaptive Technologies Inc., Blacksburg, VA 24060)

High-performance military fighter aircraft at high power settings can generate near-field noise levels typically between 140 and 150 dB SPL at approximately 50 ft. Providing adequate noise attenuation for maintenance personnel working in these locations is a challenging task. This paper describes the basic design of an active noise reduction earplug that, when combined with a passive earmuff, will operate in noise environments up to 150 dB SPL. The design and fabrication was the result of a contractual collaborative effort of the Air Force Research Laboratory and Adaptive Technologies Inc. The basic design includes a custom-molded deep insert earplug with an integrated error microphone and earphone. The active noise reduction controller was realized using digital signal processing. The performance of the ANR earplug and earmuff system was measured in both laboratory and field conditions using both real-ear attenuation at threshold and miniature microphone in real-ear measurement techniques. The performance of the system will be described in terms of both active and passive attenuation in 142–150-dB SPL broadband noise fields.

10:05–10:20 Break

4aNS4. Hearing protector labeling for active noise reduction devices. William J. Murphy, John R. Franks (NIOSH, Hearing Loss Prevention Team, 4676 Columbia Pkwy., MS C-27 Cincinnati, OH 45226-1998, wjm4@cdc.gov, jrf3@cdc.gov), and Alberto Behar (Univ. Toronto, Toronto, ON, M5S3G9 Canada)

The US EPA regulation 40 CFR part 211b does not specify how to label hearing protectors that use active noise reduction (ANR). Real-ear attenuation at threshold (REAT) measurements are appropriate for testing the passive noise reduction performance of the earmuff. The contribution of ANR to the overall attenuation must be measured either with an acoustic test fixture (ATF) or using the microphone in real ear (MIRE) technique. ATFs must adequately mimic the coupling of the protector with either the skin of the ear canal or the side of the head, as well as provide sufficient attenuation to eliminate effects of bone conduction. The MIRE technique uses a miniature microphone positioned in the ear canal either at the entrance or at the tympanic membrane, and can be used to measure the insertion loss between the occluded and unoccluded conditions for both the passive and active modes of the protector. ATF and MIRE measurements made for several protectors demonstrate effective performance below 1000 Hz. This paper will examine methods for combining ATF and MIRE measurements with REAT to develop effective and informative rating for the ANR class of hearing protectors. [Portions of work supported by the U.S. EPA IA 75090527.]

4aNS5. Influence of control structure and sound reproduction on the intelligibility of speech in headsets with active noise reduction. Anthony J. Brammer (Univ. of Connecticut Health Ctr., 263 Farmington Ave., Farmington, CT, 06030-2017, anthonybrammer@hotmail.com), Donald R. Peterson, Martin G. Cherniack, and Subhash Gullapalli (Univ. of Connecticut Health Ctr., Farmington, CT 06030-2017)

A primary requirement for a communication headset is to maintain speech intelligibility under all conditions of use. For headsets equipped with active noise reduction (ANR), the performance of the control system may influence the communication signal reaching the ear. Conversely, the communication signal may perturb the operation of the ANR system. The interaction between the communication and control signals will depend primarily on the control structure, and the electro-acoustic transfer function of the earphone and drive electronics of the communication channel. The effect on speech intelligibility will be described using the speech transmission index, for circumaural headsets employing feedback control with a fixed filter and analog signal processing, an approach commonly used in commercial devices, and one employing adaptive, digital feed-forward control. In each case, the active control systems attempt to control low-frequency noise, as the passive attenuation of the earmuff enclosing the ear substantially reduces noise at frequencies above 500 Hz. Results will be discussed for different environmental noise spectra, and speech-to-noise ratios.

11:00

4aNS6. An objective method for measuring the attenuation of hearing protection devices using otoacoustic emissions. Darin A. Knaus, Anthony J. Dietz (Creare Inc., P.O. Box 71, Hanover, NH 03755, dak@creare.com), and Frank E. Musiek (Univ. of Connecticut, Storrs, CT 06269)

An objective human subject test technique for measuring the attenuation performance of hearing protection devices is presented. The technique is similar to the real-ear attenuation at threshold (REAT) technique, except distortion product otoacoustic emission (DPOAE) measurements in the ear canal replace the standard audiometric threshold measurements. The primary advantages of the new technique are that the measurement is objec-

tive, relatively fast, and can be applied at supra-threshold sound levels. Microphone in real ear (MIRE) data is also available from the DPOAE probe in the ear canal. Disadvantages are the restricted frequency range in which DPOAE signals may be reliably recorded (2–5 kHz) and the need for an in-ear probe. Attenuation data for a set of earmuffs was measured on a pool of seven subjects using DPOAE, MIRE, and threshold methods. The DPOAE and threshold measurements were in agreement, validating the DPOAE method. The MIRE data were significantly higher than the DPOAE or threshold data. The difference in the attenuation levels represents sound reaching the cochlea that is bypassing the ear canal. This difference was smaller for measurements of the attenuation performance of a helmet, which provides increased shielding from bypass transmission paths through the head.

THURSDAY MORNING, 18 NOVEMBER 2004

PACIFIC SALONS 4 & 5, 7:45 A.M. TO 12:00 NOON

Session 4aPA

Physical Acoustics: Applications of Acoustic Radiation Pressure

Gregory A. Kaduchak, Cochair

Los Alamos National Laboratory, P.O. Box 1663, Los Alamos, New Mexico 87545

Thomas J. Matula, Cochair

Applied Physics Laboratory, University of Washington, 1013 NE 40th Street, Seattle, Washington 98105-6698

Chair's Introduction—7:45

Invited Papers

7:50

4aPA1. Manipulation of cells in ultrasound standing waves. W. Terence Coakley (Cardiff School of Biosciences, Cardiff Univ., Cardiff CF10 3TL, UK)

The acoustic radiation force operating on a cell in a 0.5-MPa pressure amplitude 3-MHz standing wave field equals that experienced in a 250-g gravitational field. Consequently, cells may be driven rapidly to preferred points in the sound field. Acoustic radiation pressure can drive bacteria, flowing by an acoustic reflector (prepared as an immunosensor surface), onto that surface where they are then captured. Animal cells have been driven onto the fabric of wide-gauge filters to form, a basis for a supported bioreactor. Real standing wave fields do not have the ideal plane-wave form, particularly when the transducer diameter is much larger than a wavelength (typical transducer diameter of 20 mm, wavelength of 0.5 mm at 3 MHz in water). Pressure variations within a node plane then give rise to regions towards which cells migrate and may be held against flow in potential traps. These traps are exploited commercially to retain cells in fermentors in the pharmaceutical industry. Most recently, ultrasound traps have been employed to concentrate animal cells in the focal plane of a fluorescence microscope so that cell interactions and their consequences at the molecular level can be monitored.

8:15

4aPA2. Nondestructive microscopic concentration, separation, analysis of biomaterials in microfluidic space using acoustic radiation force for cellomics study. Kenji Yasuda (Dept. of Life Sci., Grad. School of Arts and Sci. Univ. of Tokyo, 3-8-1 Komaba, Meguro, Tokyo 153-8902, Japan)

A method for noncontact handling of biological materials such as cells using acoustic radiation force could be incorporated in a fully automated microfluidic analysis system for cellomics studies. Benefits of acoustic radiation force are simple and compact design without moving parts, suitability for continuous concentration, separation and mixing in micrometer-sized spaces, and ready use in microchambers allowing contamination-free handling. Since this can be an advantage in biological applications for on-chip cellomics screening, such handling techniques using ultrasound are pursued. At the meeting, the following three applications are presented: (1) mixing, concentration and fractionation of biomaterials [K. Yasuda *et al.*, J. Acoust. Soc. Am. **99**, 1248–1251 (1996); **102**, 642–645 (1997), Jpn. J. Appl. Phys. **34**, 2715–2720 (1995); K. Yasuda, Sens. Actuat. B. **64**, 128–135 (2000)]; (2) superposition method for improving concentrating efficiency [K. Yasuda, Jpn. J. Appl. Phys. **36**, 3130–3135 (1997)]; (3) separation of different materials exploiting the competition between acoustic radiation force and electrostatic force [K. Yasuda *et al.*, J. Acoust. Soc. Am. **99**, 1965–1970 (1996); Jpn. J. Appl. Phys. **35**, 3295–3299 (1996), K. Yasuda, Jpn. J. Appl. Phys. **38**, 3316–3319 (1999)].

4aPA3. Acoustic levitation in the field of analytical, physical, and atmospheric chemistry. Rudolf Tuckermann (GKSS-Res. Ctr., Inst. for Coastal Res., Max-Planck-Str.1, D-21502 Geesthacht, Germany, rudolf.tuckermann@gkss.de)

In this contribution an overview of fundamentals, possibilities, and applications of acoustic levitation in the field of analytical, physical, and atmospheric chemistry investigated by Neidhart and co-workers since the 1990s will be given. Started with sample preparation procedures in acoustically levitated drops, e.g., analytical enrichment, liquid-liquid extraction, and acid-base titration [E. Welter and B. Neidhart, *Fresenius J. Anal. Chem.* **357**, 345–350 (1997); R. Eberhard and B. Neidhart, *ibid.* **365**, 475–479 (1999); O. Rohling, C. Weitkamp, and B. Neidhart, *ibid.* **368**, 125–129 (2000)], the work has been continued in evaporation processes of liquids [R. Tuckermann, S. Bauerecker, and B. Neidhart, *Analytical Bioanal. Chem.* **372**, 122–127 (2002)], crystallization, and freezing processes in acoustically levitated drops, formation and characterization of monomolecular films on the surface of levitated drops, formation and growth of particles [S. Bauerecker and B. Neidhart, *J. Chem. Phys.* **109**(10), 3709–3712 (1998); *Science* **282**, 2211–2212 (1998)] and trapping of heavy gases [R. Tuckermann, B. Neidhart, E. G. Lierke, and S. Bauerecker, *Chem. Phys. Lett.* **363**, 249–354 (2002)] in stationary ultrasonic fields. Examples and results of these investigations will be presented.

4aPA4. Acoustically assisted phase-separation processes. Donald Feke (Dept. of Chem. Eng., Case Western Reserve Univ., Cleveland, OH 44106)

The separation and recovery of micron- to millimeter-sized secondary phases suspended in liquids is of fundamental importance in a variety of chemical, biochemical, and materials processing applications. These secondary phases can be solid particles, immiscible liquid droplets, or gas bubbles. In our laboratory, we have developed a separation technique in which the application of acoustic radiation pressure enables the phase separation within meshes having pore size up to 100 times the size of the secondary phase being retained. For the case of droplets and bubbles, the transport and coalescence induced by the acoustic radiation forces augments the separation efficiency. As a means to understand the fundamental physics of this process, we have developed both micro- and macro-scale models of the behavior of dispersed phases in our laboratory-scale acoustic devices and have performed experiments to validate these models. Micro-scale models predict the motion of individual dispersoids in response to the applied acoustic and flow fields, while the macro-scale models predict overall collection dynamics and breakthrough times. Generally, the correspondence between model predictions and experimental observation is excellent. These models thus reveal the fundamental physics that underlies the enhancement of the separation process by the acoustic radiation pressure.

4aPA5. Manipulation of flow fields using acoustic radiation pressure. Eugene H. Trinh (Exploration Systems Directorate, NASA HQ, Washington, DC 20546)

The accurate trapping and positioning of single drops and bubbles in outer host fluids is a capability that can be utilized to study the detailed fluid motion within these particles, as well as in the outer host fluid. The understanding of this motion and the ability to control its magnitude impacts the use of the mechanisms involved in mass and energy transport across the drop and bubble boundaries. We used acoustic radiation pressure generated by an ultrasonic field between 25 and 60 kHz in degassed water to trap centimeter-size air bubbles and to observe the correlation between the onset of capillary waves on the bubble surface and the generation of internal air flow inside the bubbles. The modulation of the ultrasonic field has also allowed the observation of coupling between the resonant bubble radial oscillation mode and the shape oscillation modes for smaller diameter samples. Other applications of radiation pressure combined with electrostatic levitation of charged drops will be discussed together with some applications in the field of crystal growth and liquid surface property measurements. [Work funded through NASA.]

4aPA6. Manipulation of fluid objects using acoustic radiation pressure: Beyond drops and bubbles and retrospection. D. B. Thiessen, J. B. Lonzaga, C. F. Osterhourdt, W. Wei, and P. L. Marston (Phys. Dept., Washington State Univ., Pullman, WA 99164-2814)

Some standard applications of acoustic radiation pressure include the manipulation of drops and bubbles for the study of interfacial dynamics and light scattering. After a review, this presentation summarizes recent applications including radiation pressure effects on small flames and liquid cylinders [Wei *et al.*, *J. Acoust. Soc. Am.* **116**, 201–208 (2004)]. The dependence of the radiation pressure on the acoustic frequency has important consequences including liquid cylinder stabilization. While the radiation pressure is normally associated with sound propagation in the outer gas or liquid, our recent experiments also show that interfacial responses to modulated radiation pressure can be significant when the sound is applied internally to the manipulated object. For example, a liquid jet in air is an acoustic waveguide having a cutoff frequency inversely proportional to the jet diameter. When modulated ultrasound is applied to the jet supply liquid with the ultrasonic carrier frequency near-to or above the cutoff frequency, the jet can be made to oscillate at the modulation frequency of the radiation pressure, thus stimulating jet breakup. [Work supported by NASA.]

4aPA7. Acoustic particle manipulation devices driven by resonantly excited, cylindrical structures. Gregory Kaduchak, Greg R. Goddard, Christopher S. Kwiatkowski, and Dipen N. Sinha (Los Alamos Natl. Lab., Electron. and Electrochem. Mater. and Devices Group, Los Alamos, NM 87545)

An overview of acoustic particle manipulation devices that use resonantly excited structures as the displacement generators will be discussed. These devices rely on coupling of the structural modes of a hollow, cylindrical tube to the acoustic modes set up in its interior cavity. For practical applications, discussed for both particles in air and in liquids, the structural vibrations are tuned such that desired modal responses of the cavity are obtained. The effects on particle manipulation due to different internal configurations and symmetry-breaking of the cross-sectional geometry will be shown in experimental data and videos. Specific designs of these devices have demonstrated that they can be engineered into power-efficient assemblies. It will be shown that different structural and cavity modifications can be made to meet the resonance matching condition for several different azimuthal mode numbers. Material and geometry considerations for nonresonantly driven devices will also be discussed. Applications include sample positioning for real-time analysis in flow cytometers, and aerosol concentration and positioning for in-line aerosol detectors.

Contributed Papers

11:00

4aPA8. Forces generated in biological tissue-like media due to second-order effects in the focal region of an ultrasonic beam. Lev Ostrovsky (Zel Technologies/Univ. of Colorado, Boulder, CO 80305), Alexander Sutin, and Armen Sarvazyan (Artann Labs., Lambertville, NJ 08530)

Several techniques of medical diagnostics have been developed in the last decade based on the application of radiation forces of focused ultrasound. The most commonly considered physical mechanism of the radiation force generation is essentially related to ultrasound beam attenuation. However, in the case of short ultrasonic pulses typical for conventional ultrasonic imaging systems, an additional mechanism of force generation exists that is caused by acoustic nonlinearity of the media and does not explicitly depend on viscous dissipation. For water-like media such as human tissue, our theoretical description of the preceding effects is based on Eckart's theory of second-order effects in fluids, although modified by taking into account the shear modulus in the rotational component. In our theory, nonlinearity in a pulsed beam produces also a longitudinal component of an averaged force and a resulting displacement. Estimates show that for short acoustic pulses generated in conventional diagnostic ultrasound systems, the latter component can make a significant and even dominant contribution to the averaged displacement in the focal area of an ultrasonic beam.

11:15

4aPA9. An investigation of the effect of sound field distribution on the stability of the levitated object in the near-field acoustic levitation. Cheol-Ho Kim (35-3, HongChonRi, IbJangMyun, ChonAnSi, (330-825) Korea), Jeong-Guon Ih (KAIST Sci. Town, Guseong-dong, Yuseong-Gu, Taejon, Korea), and Sung-II Lee (HongChonRi, IbJangMyun, ChonAnSi, (330-825) Korea)

Near-field acoustic levitation (NFAL) is a method that levitates an object or group of particles using the ultrasonic force. It is observed that, when a circular piston type of source is employed, stability problems can occur which makes the object position unpredictable. For example, a thin, circular planar object wobbles to lateral direction like a pendulum when its center is not precisely coincident with the center of source. In this study, the motive force, which makes the levitated object swivel through the center of source, is investigated by varying the sizes of the object and the source. To this end, a visualization of the sound field in the micro gap between the levitated object and source is made by using a photo view of the motion of fine powder scattered inside. The result is compared with the calculation result by BEM. A further investigation is experimentally made on any relation between the temperature distribution on the surface of the levitated object and the acoustic force. The experimental visualizations and numerical modeling results suggest that one may have a model describing the instability of the levitated object in NFAL, if a proper simplified mathematical model is set up in the further study.

11:30

4aPA10. Optical detection of the response of liquid jets to internal modulated ultrasonic radiation pressure. Joel B. Lonzaga, Curtis F. Osterhourdt, David B. Thiessen, and Philip L. Marston (Phys. Dept., Washington State Univ., Pullman, WA 99164-2814)

A liquid jet in air is an acoustic waveguide having a cutoff frequency inversely proportional to the jet diameter. Ultrasound applied to the jet supply liquid can propagate down the jet when the carrier frequency is near-to or above the cutoff frequency. The jet response to the internal acoustic radiation pressure of amplitude modulated ultrasound was monitored along the jet using an optical pseudo-extinction method. The jet profile oscillates at the frequency of the radiation pressure modulation and, where the response is small, the amplitude was found to increase in proportion to the square of the acoustic pressure amplitude as previously demonstrated for oscillating drops [P. L. Marston and R. E. Apfel, J. Acoust. Soc. Am. **67**, 27-37 (1980)]. Small amplitude deformations initially grow approximately exponentially with axial distance along the jet. Modulated radiation pressure can be used to stimulate large amplitude deformations and the breakup of the jet into drops. [Work supported by NASA.]

11:45

4aPA11. Use of a slightly curved reflector to enhance an airborne ultrasonic standing wave in a semi-infinite rectangular channel. Michael Anderson, Andrew Cluff, E. Clark Lemmon, and Gabriel Putnam (Dept. of Mech. Eng., Univ. of Idaho, Eng./Phys. Bldg. 324O, Moscow, ID 83844-0902)

One way to separate small particles from a moving air stream is to pass the stream through an intense ultrasonic standing wave. If the standing wave propagates perpendicular to the fluid flow direction, acoustic radiation pressure will move the particles to specific locations in the stream, where they can be collected at the stream outlet. Of primary importance is to achieve very high pressure amplitude in the standing wave, in spite of the presence of openings for fluid flow. In this presentation, the use of a slightly curved reflector within a flow channel to increase the amplitude of an ultrasonic standing wave is discussed. A finite element analysis was used to predict the amplitude of the standing wave that would be excited in a semi-infinite channel. A perfectly matched layer was used to account for the semi-infinite geometry. The finite element analysis showed that a significant gain in amplification within the channel could be achieved with a surprisingly small amount of reflector curvature. Experiments show that much of this gain can be obtained in practice.

Session 4aPP

Psychological and Physiological Acoustics: Psychological and Physiological Acoustics (Poster Session)

Li Xu, Chair

School of Hearing, Speech and Language Sciences, Ohio University, Athens, Ohio 45701

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 10:30 a.m. and contributors of even-numbered papers will be at their posters from 10:30 a.m. to 12:00 noon.

4aPP1. A psychophysical measure of level-dependent shifts in the peak of the traveling wave. Christopher A. Brown, Changmo Jeung, and Sid P. Bacon (Arizona State Univ., P.O. Box 870102, Tempe, AZ 85287-0102, c-b@asu.edu)

Thresholds for detecting a 50-ms pure-tone probe in the presence of a 200-ms pure-tone forward masker were measured as a function of masker level (50, 70, or 90 dB SPL). The masker frequency was 250, 1000, 4000, or 6000 Hz, and the probe frequency was 1.00, 1.10, 1.20, 1.30, 1.40, or 1.60 times the masker frequency. For subjects 1–3, a 50-ms delay was used. For subjects 4–6, subject-dependent delays were chosen to keep the amount of masking to about 10 dB or less, to restrict the influence of off-frequency listening. These delays ranged from 50 to 110 ms. Subjects 1–3 showed the greatest amounts of masking on frequency in almost all conditions, whereas subjects 4–6 tended to show the most masking at frequencies above the masker frequency, but only at the highest masker levels and only at masker frequencies above 250 Hz. Although inconclusive, these results are in general agreement with both psychophysical and physiological studies that suggest a level-dependent basalward shift in the peak of the traveling wave along the basilar membrane. [Work supported by NIDCD.]

4aPP2. A systems and neurobiological model of nucleus angularis and nucleus laminaris in the barn owl. Brian J. Fischer (Dept. of Elec. and Systems Eng., Washington Univ. in St. Louis, Campus Box 8108, 660 S. Euclid, St. Louis, MO 63110) and Charles H. Anderson (Washington Univ. in St. Louis, St. Louis, MO 63110)

The representation of binaural sound localization cues was investigated in a model of the barn owl's auditory nerve, nucleus angularis (NA), nucleus magnocellularis (NM), nucleus laminaris (NL), and superior olivary nucleus (SON) using spiking leaky integrate and fire neurons. A phenomenological model of the barn owl's auditory-nerve spike response was created to reproduce responses to tonal stimuli and exhibit two-tone rate suppression effects. The representation of stimulus level by NA was investigated by decoding the envelope of the input signal from populations of NA model neurons where the envelope was defined as in a previous computational model of sound localization in the barn owl [B.J. Fischer and C.H. Anderson, *Adv. NIPS* **16** (2004)]. Simulations demonstrated that inclusion of NA neurons with type IV responses increased the accuracy of the representation of the stimulus envelope. The effect of inhibition on the output of the population of NL model neurons was investigated and compared to a cross-correlation model. The output of the model SON provides inhibitory feedback to ipsilateral NA, NM, and NL model neurons. Simulations demonstrated that inclusion of inhibition of NL neurons by SON output increased peak–trough differences in ITD tuning curves.

4aPP3. The influence of spread of excitation on the temporal modulation transfer function (TMTF) for sinusoidal carriers. Rebecca E. Millman and Sid P. Bacon (Dept. of Speech and Hearing Sci., Arizona State Univ., P.O. Box 870102, Tempe, AZ 85287-0102)

The improvement in modulation detection thresholds with increasing stimulus level has been attributed to listening on the high-frequency side of the excitation pattern, where the growth of excitation is more linear, or to an increase in the number of channels via spread of excitation. TMTFs were measured for sinusoidal amplitude modulation (SAM) applied to a 1-kHz sinusoidal carrier, whose level varied between 10 and 80 dB SPL. Modulation detection thresholds improved as the carrier level increased. To minimize the use of spread of excitation at a carrier level of 80 dB SPL, various tonal restrictors were used alone and in combination. The restrictors were presented continuously and their frequencies were 501, 801, 1210, or 1510 Hz. Only the combination of the 801 and 1210-Hz restrictors resulted in an increase in modulation detection thresholds to values comparable to those obtained at 30 dB SL. These data suggest that listening on the high-frequency side of the excitation pattern alone cannot account for better modulation detection thresholds at higher levels, and that spread of excitation above and below the carrier is required for optimal performance. [Work supported by NIDCD.]

4aPP4. Auditory evoked potentials to changes in interaural cross correlation. Helmut Riedel, Helge Lüddemann, and Birger Kollmeier (Medizinische Physik, Universität Oldenburg, 26111 Oldenburg, Germany)

The objective of this study is to analyze the representation of specific binaural stimulus features in the human brain and to compare the results to psychoacoustic data. Late auditory evoked potentials to changes in the interaural cross correlation (IACC) of continuous bandpass noise (100–2000 Hz) were recorded, 1000 sweeps per stimulus condition were averaged. In experiment 1 a brief “binaural gap” with a gap IACC was interposed in the noise with a reference IACC. N1 and P2 amplitudes increased with increasing gap duration. For uncorrelated reference IACC (0/1/0 and 0/–1/0 conditions) objective thresholds were comparable to psychoacoustic data. For uncorrelated gap IACC (1/0/1 and –1/0/–1 conditions) objective thresholds were about four times larger than in psychoacoustic measurements. In experiment 2 reference and test interval had the same length of 800 ms and the magnitude of change in IACC was varied. Objective IACC-JNDs are larger for an uncorrelated reference than for IACCs of ± 1 . For the uncorrelated reference, test IACCs towards +1 elicited larger AEP components than test IACCs towards –1. In contrast to binaural difference potentials used at the brain stem level, employing changes in IACC constitutes a more direct approach to objectively investigate the binaural system in humans.

4aPP5. Psychophysical evidence for the existence of envelope information in the internal auditory representation of signal fine structure. Frederic Apoux, Rebecca E. Millman, Christopher A. Brown, Sid P. Bacon (Psychoacoust. Lab., Dept. of Speech and Hearing Sci., Arizona State Univ., Tempe, AZ 85287-0102), and Neal F. Viemeister (Univ. of Minnesota, Minneapolis, MN 55455)

This study examined the extent to which the auditory system can recover envelope information from phase information contained in signal fine structure [O. Ghitza, *J. Acoust. Soc. Am.* **110**, 1628–1640 (2001)]. A 2IFC paradigm was used to assess amplitude modulation (AM) detection performance for stimuli whose envelope was removed using the Hilbert transform, leaving only the fine structure. One interval contained the fine structure of an amplitude modulated stimulus, the other the fine structure of an unmodulated stimulus. The first experiment measured AM detection for a fixed modulation index ($m=0.95$). The original signal carriers were broadband noise, harmonic tone complexes, and positive Schroeder-phase complexes modulated at rates of 5, 10, or 15 Hz. Listeners were unable to detect AM when presented with the fine structure only, suggesting little envelope recovery. The second experiment measured AM detection thresholds for modulation rates of 5, 10, or 15 Hz imposed, before combination, on one of three nonharmonically related tones. Preliminary results indicate that thresholds are comparable whether listeners are presented with the full signal or fine structure only, suggesting that the depth of the recovered AM is similar to that of the original AM for these stimuli. [Work supported by NIDCD.]

4aPP6. Effects of talker variability on vowel recognition by cochlear implant users and normal-hearing subjects listening to cochlear implant simulations. Yi-ping Chang, Qian-Jie Fu (Dept. of Biomed. Eng., Univ. of Southern California, Los Angeles, CA 90089, yipingch@usc.edu), and Qian-Jie Fu (DAIP, Los Angeles, CA 90057)

Different talker characteristics can significantly affect speech recognition, especially under conditions of reduced spectral resolution. In the current study, single- and multi-talker vowel recognition was tested with both cochlear implant (CI) patients and normal-hearing (NH) subjects listening to CI simulations. CI users were tested with their everyday processors. NH subjects were tested using four-channel acoustic processors in which the envelope cutoff frequency (20, 160 Hz) and the carrier band (noise-band, sine-wave) conditions were varied. Results showed that CI users' single-talker vowel recognition was significantly better than multi-talker recognition. Among the simulations, both single- and multi-talker vowel recognition was best with the 160 Hz/sine-wave processor. Also, single-talker performance was significantly better than multi-talker performance only with the 160 Hz/sine-wave processor; there was no difference between single- and multi-talker recognition with the 20 Hz/sine-wave or noise-band processor. While the effects of talker variability were less for the 20 Hz/sine-wave and noise-band processors, overall vowel recognition was also significantly lower. Listeners' sensitivity to talker variability may indicate their ability to register the relative vowel space for individual talkers. Without sensitivity to talker variability, multi-talker vowel space may collapse, causing cross-talker differences to be confused with phonetic differences. [Research is supported by NIDCD.]

4aPP7. Revisiting the quartic model for early identification of noise-induced hearing loss. William J. Murphy and John R. Franks (NIOSH Hearing Loss Prevention Team, 4676 Columbia Pkwy., MS C-27, Cincinnati, OH 45226-1998, wjm4@cdc.gov)

In a 1976 article by Cooper and Owen ["Audiologic profile of noise induced hearing loss," *Arch. Otolaryngol.* **102**, 148–150 (1976)], they advanced a quartic model to perform a least-squares fit to an audiogram for the purpose of identifying a noise-induced notch indicative of early onset of noise-induced hearing loss. This model has been revisited for the purpose of better quantification of a set of rules that will best identify the presence or absence of a noise notch. A subset of 1660 subjects' right and left audiograms collected for the National Health and Nutrition Examination

Survey (NHANES IV) was fit with a quartic model. As well, several hearing scientists and audiologists within NIOSH examined and classified the audiograms as having a notch or not. Preliminary results suggest that the choice of notch criteria dramatically affects the identification accuracy when correlated with clinical judgments. This paper will present how these rules were developed and how the quartic model may be interpreted and applied in a hearing conservation program.

4aPP8. Early indicators of noise-induced hearing loss: Issues for consideration. Mark R. Stephenson, William J. Murphy, and John R. Franks (NIOSH, Hearing Loss Prevention Team, 4676 Columbia Pkwy., MSC-27, Cincinnati, OH 45226-1998, mos9@cdc.gov)

A recent article in the NHCA Spectrum [Dobie and Rabinowitz, **20**, 8–11 (2003)] considered several methods for identifying the early onset of noise-induced hearing loss (NIHL). Among the methods they examined was the Notch Index: the difference in the pure-tone average of 2, 3, and 4 kHz and the pure-tone average of 1 and 8 kHz. This statistic was expected to be positive for NIHL where noise has produced a threshold shift in the 2- to 4-kHz region but not adversely affected the thresholds at 1 and 8 kHz. The Notch Index was examined for a subset of 1660 subjects' right and left audiograms collected for the National Health and Nutrition Examination Survey (NHANES IV). The Notch Index did not prove to be effective as expected. However, modifications to the calculation, the inclusion of 6 kHz, or calculation of the area difference provided better identification of the early onset of NIHL. This paper will examine other issues such as work-relatedness and potential gender effects that might be influenced by differential trends in ear canal size.

4aPP9. Relating the acoustic vowel space to perceptual vowel space in cochlear implant simulation. Chuping Liu, Qian-Jie Fu (Dept. of Elec. Eng., Univ. of Southern California, Los Angeles, CA 90007, chupingl@usc.edu), and Qian-Jie Fu (Los Angeles, CA 90057)

Because of limited spectral resolution, the acoustic differences between important speech features may not be preserved by cochlear implants (CIs). The present study compared the acoustic vowel space of spectrally degraded speech with the perceptual vowel space of normal-hearing (NH) subjects to CI simulations in which the spectral resolution, frequency allocation, carrier bands (noise-band or sine-wave) and signal-to-noise ratios (SNRs) were varied. For each speech-processing condition, acoustic differences between vowel tokens were calculated by extracting the mel-frequency cepstrum coefficients, time-aligning the tokens, then computing the weighted Euclidean distances. Results showed that the acoustic vowel space monotonically expanded with increasing SNR and spectral resolution; sine-wave carriers also expanded the acoustic vowel space, relative to noise-band carriers, for all SNR conditions. We will compare the acoustic vowel space to NH subjects' perceptual data for these speech-processing conditions. [Research is supported by NIDCD R01-DC-004993.]

4aPP10. Comparison of acoustic features of time-compressed and natural speech. Akiko Kusumoto and Nancy Vaughan (NCRAR, Portland VA Medical Ctr., 3710 SW US Veterans Hospital Rd., Portland, OR 97207)

Time-compressed speech is used as a speech material to examine the upper limits of processing speed in young and old listeners with and without hearing loss. Generally, older normal-hearing listeners at 60% time compression perform similarly to younger normal-hearing listeners at 70% time compression on speech recognition test. This may be due to slowing of processing speed that typically accompanies aging. It has not been determined if time-compressed speech material carries spectral distortions or other acoustic changes that affect the ability to process rapidly. If so, differences in performance between younger and older listeners may be attributable in part to acoustic distortions other than temporal changes. In

this work, we examine acoustic feature changes in time-compressed speech processed using synchronized overlap-add (SOLA) method. Also, we discuss which parts of speech (consonants or vowels) are more affected by time compression. In the temporal domain, we compare the duration of consonants, vowels and voice onset times (VOT) in both 50% compressed speech and natural speech. We also examine the long-term power spectra of both signals. Possible effects on speech intelligibility will be discussed.

4aPP11. Correlation of speech recognition by cochlear implant users with metrics based on thresholds and loudness levels. Li Xu (School of Hearing, Speech and Lang. Sci., Ohio Univ., Athens, OH 45701) and Bryan Pfingst (Univ. of Michigan, Ann Arbor, MI 48109-0506)

Intersubject variability in speech recognition is a persistent characteristic of cochlear implant users, despite significant improvements in average performance. This study examines simple psychophysical measures, maximum comfortable loudness levels (C-levels) and dynamic ranges (DRs) that may serve for diagnosis of conditions that affect speech recognition. Eighteen subjects who had used Nucleus CI24 cochlear implants for a minimum of 1 year participated in the study. T and C levels were measured using the method of adjustment. Consonant, vowel, and sentence recognition were tested using the subjects everyday speech processors (SPEAK or ACE). Across-site variation (ASV) was quantified for each subject using the variance of the T and C levels across all tested stimulation sites. Mean DRs (C levels minus T levels) for all tested sites were calculated for each subject. The results showed a moderate but significant negative correlation between speech-recognition performance and ASV of the T and C levels measured with both monopolar and bipolar electrode configurations. Mean DRs also correlated with speech perfor-

mance. These results support the hypothesis that the large across-site variation in detection thresholds and small dynamic range reflect conditions that are detrimental to speech-recognition with cochlear implants. [Work supported by NIDCD Grant DC03808.]

4aPP12. Effects of short-term training on the recognition of spectrally shifted vowels by non-native listeners. Tianhao Li, Qianjie Fu, and Geri Nogaki (DAIP, House Ear Inst., 2100 W. 3rd St., Los Angeles, CA 90057, tianhaol@usc.edu)

Previous studies have shown that normal-hearing (NH) native listeners can rapidly adapt (though not always completely) to spectrally mismatched speech with short-term training. However, it is unclear how short-term training might affect the performance of non-native listeners. The present study investigated whether non-native listeners' recognition of spectrally shifted English vowels could be improved by moderate short-term training. Recognition of 12 American English vowels (in c/V/c format) was trained and tested in five Chinese-speaking NH subjects; speech tokens were processed by an eight-channel noise-band vocoder in which the spectrum was upwardly shifted by one octave. After baseline testing subjects were trained with shifted speech for 5 consecutive days; daily training consisted of three to four sessions, each of which included a 5-min preview of the test tokens and a 10-min test. Results showed that for all subjects, recognition of spectrally shifted English vowels improved significantly over the 5-day training period. These results show that non-native listeners are able to rapidly adapt to spectrally mismatched speech with short-term training. The ability to fully accommodate spectrally shifted speech may depend on non-native listeners experience with the second language. [Research supported by NIDCD R01-DC-04993.]

THURSDAY MORNING, 18 NOVEMBER 2004

ROYAL PALM SALON 5, 9:00 TO 11:15 A.M.

Session 4aSA

Structural Acoustics and Vibration: Vibration of Sports Equipment

Daniel A. Russell, Chair

Science and Mathematics Departments, Kettering University, 1700 West Third Avenue, Flint, Michigan 48504-4898

Invited Papers

9:00

4aSA1. Vibration and sound radiation of solid wood and tubular metal baseball bats as a function of ball-bat location.

Robert D. Collier (Thayer School of Eng., Dartmouth College, Hanover, NH 03755), Kenneth Kaliski (Resource Systems Group, Inc., White River Junction, VT 05001), and James A. Sherwood (Univ. of Massachusetts Lowell, Lowell, MA 01854)

The "crack of the bat" is an important part of the game of baseball played with solid wood bats. The spectral characteristics of the radiated sound depend on the location of the ball-bat impact location along the length of the barrel and the resulting bat vibration. Balls hit on the sweet spot generally are hit harder and result in a distinct and recognizable "crack" sound. On the other hand, balls hit in on the handle or off the end of the barrel result in different modes of vibration and low-frequency radiation, i.e., a "thunk-like" sound. Analytical predictions and modal analyses, supported by both laboratory and field measurements, provide a more comprehensive picture of bat vibration and radiated sound relationships. Comparisons are made with tubular metal bats which exhibit the narrow-band signature due to excitation of the cylindrical breathing (bell) modes which are essentially independent of hit location. The differences in bat vibration and sound radiation can provide important clues for baseball players in the field.

9:30

4aSA2. Cricket bats: Some performance and vibration control issues.

Sabu John, Cao Jialong, Fatmir Gutaj (RMIT Univ., School of Aerosp., Mech. and Manuf. Eng., Bundoora East Campus, Bundoora, VIC 3083, Australia), and Tom Molyneux (RMIT Univ., Melbourne, VIC 3001, Australia)

This paper will address some of the issues in an on-going research project involving the performance optimization and vibration control of Cricket bats. In many ways, the rules of the game have made our task a little easier by restricting exactly what we can modify in order to improve performance. For example, the blade or the hitting proportion of the Cricket bat has to remain as wood but

4a THU. AM

its geometry can be altered to fit within a volume-based specification. The handles specification, however, is not specified. Hence by our interpretation of the rules of the game, this lack of specificity for the handle in the rules provides an opportunity to improve the performance of the Cricket bat as a whole by making material and geometric modifications to the handle. This paper will address the investigation of bat performance from a dynamics and vibration analyses perspective. This paper will report on parameters such as bat-ball contact time and translation, clustering of distal node points (DNPs) to mimic an enhanced sweet spot. Results showing performance comparisons between traditional and composite material-handle bats will be presented along with computational and analytical results for various handle shapes.

10:00–10:15 Break

10:15

4aSA3. Dynamic characterization of golf equipment. Tom Mase (Composite Mater. and Structures Ctr., Michigan State Univ., East Lansing, MI 48824)

In the golf swing the shaft, head, and ball respond at different time scales. The shaft is swung from the top of the back swing to impact in less than a half second. In that time, the shaft loads and unloads in different ways for different shafts. Natural frequencies and mode shapes characterize different shafts uniquely making some shafts better than others for individual golfers. Shafts are benchmarked by measuring their stiffness profile along the length of the shaft. Knowing this, a shaft finite element model is generated from which the natural frequency is found. Shaft zone frequencies, an important fitting method, are cantilevered natural frequencies found from the stiffness profile. As for golf balls, their core and cover mechanical properties influence performance and feel. The loading of the golf ball occurs in 0.0005 s. Golf ball stiffness and damping properties are measured using DMA and universal testing machines. Distance balls have cores that are nominally twice as stiff as performance balls, making them feel harder to the golfer. Performance and distance balls also respond to strain rate changes differently. Ultimately, stiffness properties affect the initial velocity, spin, and launch angle which dictate the distance the ball flies.

10:45

4aSA4. The sweet spot of a hollow baseball or softball bat. Daniel Russell (Sci. and Mathematics Dept., Kettering Univ., Flint, MI 48504)

The “sweet spot” of a baseball bat is often defined as a region approximately 5–7 in. from the barrel end of the bat where the performance (as measured by a batted-ball speed) is the greatest and the vibrational sensation (the sting felt in the hands) is minimized. In terms of the vibrational properties of a bat, the sweet spot region includes the nodes of the first two bending vibrational modes and is often also identified as being related to the center-of-percussion (COP). This paper will discuss recent research which shows that the COP has no bearing on performance and little, if any, on the perception of feel. We will also discuss the contribution of the bending vibrational modes to the location of the sweet spot and compare results for a wide variety of wood, metal and composite baseball and softball bats. We will attempt to correlate the sweet spot defined in terms of performance and that defined in terms of vibrational sensation. Finally, we will discuss the contribution of the hoop mode, unique to hollow bats, which correlates strongly to performance, and which may also influence the perception of feel.

THURSDAY MORNING, 18 NOVEMBER 2004

SUNRISE AND SUNSET ROOMS,
8:00 A.M. TO 12:00 NOON

Session 4aSC

Speech Communication: Foreign-Accented Speech: Production, Perception and Applications (Lecture/Poster Session)

Amee P. Shah, Chair

*School of Communication Sciences and Disorders, McGill University, 1266 Pine Avenue West,
Montreal, Quebec, H3V 1C2 Canada*

Chair's Introduction—8:00

Invited Papers

8:05

4aSC1. Why do non-native speakers have a foreign accent? A three-dimensional perspective. Amee P. Shah (School of Commun. Sci. and Disord., McGill Univ., 1266 Pine Ave. West, Montreal, QC H3V 1C2, Canada, amee.shah@mail.mcgill.ca)

A three-dimensional perspective, following the speech-chain model, is taken in arriving at the variables that influence the production and perception of foreign-accented speech. Essentially, research to date indicates the interactive role of all three communication components of the speech-chain model. First, speech-related variables, i.e., the interlanguage differences in the phonetic patterns of the speech, of L2 speakers compared to the L1 speech patterns influence listeners perception of accentedness of non-native

speech. Second, speaker-related variables (i.e., differences in age, other psychological variables) cause the non-native speakers to have difficulties in learning to map new sounds of the L2 onto their existing L1 phonetic system, thus resulting in foreign-accented speech patterns. Lastly, differences in listener-related factors (i.e., L1 of the listener, prior linguistic experience, amount of exposure, listening-conditions in which they hear the accented speech) have been found to influence the perception of foreign-accentedness of speech. Past and current findings will be brought to bear upon this issue that has implications in theoretical understanding of speech perception, as well as practical applications in accent-modification and ESL classroom training programs.

8:25

4aSC2. Understanding foreign-accented speech. Ann Bradlow (Linguist. Dept., Northwestern Univ., 2016 Sheridan Rd., Evanston, IL 60208)

Why is foreign-accented speech hard to understand? In one view, foreign-accented speech intelligibility is poor due to its marked deviation from the native talker norm. An alternative view claims that if the talkers and listeners shared knowledge systems are sufficiently aligned with each other, foreign-accented speech intelligibility can be good despite its deviation from the native talker norm. Results from two studies in support of this latter alternative will be presented. Study 1 investigated whether intelligibility of Chinese-accented English could be improved for native English listeners through perceptual training. This study showed that if given exposure to multiple talkers of Chinese-accented English during training, native English listeners could achieve talker-independent adaptation to Chinese-accented English. Study 2 compared intelligibility of foreign-accented and native-accented English for non-native and native listeners. Results showed that native listeners found native-accented English more intelligible than foreign-accented English, but non-native listeners found foreign-accented English as intelligible as native-accented English. Interestingly, this pattern was obtained even for non-native listeners and talkers from different language backgrounds. Taken together, these findings demonstrate that intelligibility of foreign-accented speech varies depending on the degree of talker-listener attunement rather than on the extent of its deviation from a native talker norm *per se*.

8:45

4aSC3. Theoretical models of non-native speech perception: Implications for perception of foreign-accented speech. Catherine Best (MARCS Auditory Labs., Univ. Western Sydney, Penrith NSW 1797 Australia, and Haskins Labs., 270 Crown St., New Haven, CT 06511)

Linguistic experience systematically affects speech perception, specifically for non-native phonemes and contrasts. Native language influences are strong in adults, and emerge in infancy. Several theoretical models of non-native speech perception have been proposed, including the Perceptual Assimilation Model (PAM: Best), Speech Learning Model (LSM: Flege), and Native Language Magnet model (NLM: Kuhl). Findings that infants learn statistical distributions of phoneme sequences, and phonetic details, converge on the notion that exposure to specific phonetic patterns affects the perception of phonological structure. A reasonable extrapolation is that perception of even native utterances should be influenced by their pronunciation with either familiar or unfamiliar phonetic structure. That is, if a listener is inexperienced with a given foreign accent, or another native language dialect, perception should be predictably affected when native phonemes/words are produced with those accents. This issue has received modest research attention, which has focused on sentences/words rather than phonemes. Findings suggest that unfamiliar accents do reduce recognition and intelligibility of native sentences and words. The implications of non-native speech models, particularly PAM and SLM, for understanding and studying the impact of foreign accent on perception of native phonemes and contrasts, as well as longer utterances, will be discussed. [Work supported by NIH.]

9:05

4aSC4. Bridging the gap between speech research and L2 pedagogy. Murray J. Munro (Dept. of Linguist., Simon Fraser Univ., Burnaby, BC V5A 1S6, Canada, mjmunro@sfu.ca)

In studies of L2 speech learning it is common to draw conclusions (sometimes as an afterthought) about the potential pedagogical value of research findings, particularly when they provide insights into specific pronunciation difficulties faced by L2 learners or when they indicate improved perception or production as a result of laboratory training. However, many important research findings have, as yet, had only a minor impact on second language teaching, and there remains a significant gap between what has been empirically established in the research laboratory and what is actually practiced in the classroom. This presentation will identify some of the reasons for this disparity through an examination of goals and priorities that are commonly accepted by pedagogical specialists. In particular, it will be argued that the implementation of research findings depends on establishing practical ways of improving L2 learners' speech intelligibility (as opposed to mere accent reduction) for a diverse audience of interlocutors in authentic interactive settings. In general, there is a need for research that is specifically motivated by pedagogical concerns and that is informed by an awareness of a wide range of current issues in applied linguistics. [Research supported by SSHRC.]

9:25

4aSC5. Please understand me: Toward modeling of non-native speech in automatic speech recognition Laura Mayfield Tomokiyo (Cepstral, LLC 1801 E. Carson St., Pittsburgh, PA 15203)

Non-native speech represents a fundamental challenge for automatic speech recognition (ASR). Most major advances in ASR technology assume consistency in certain features that native speakers of a language share, such as coarticulatory patterns, control of syntax, and conventions in lexical choice. Learners of a language, however, traverse very individual, and often unstable, trajectories toward spoken competence. While ASR systems can often handle highly proficient non-native speech with little or no modification of their statistical models, recognition performance can be very poor as proficiency levels and speaker fluency fall. Interest in handling foreign-accented speech in ASR has emerged from two different directions: large-scale applications such as dictation and telephony systems, and educational applications offering tutoring and diagnostics. Developers of the former have seen the need for adaptation to

non-native speech as their systems become more widely deployed in the real world. And, as speech technology has improved, it is seen as a potentially viable tool in foreign-language education. This presentation will highlight approaches to and advances in treatment of non-native speech in ASR for these two domains, examining both characteristics of learner speech and methods for improving recognizer performance.

9:45–9:55 Summary Remarks

9:55–10:05 Break

10:05–10:55 Panel Discussion

10:55–11:00 Break

Contributed Papers

All posters will be on display and all authors will be at their posters from 11:00 a.m. to 12:00 noon in the Sunset Room.

4aSC6. Relations between native Spanish speakers' perception and production of English vowels. Kimberly A. Lydtin (Dept. of Speech, Commun. Sci., and Theatre, St. John's Univ., Jamaica, NY 11439; Grad. Ctr, CUNY; NYC Dept. Educ.), Fredericka Bell-Berti, Peggy Jacobson, and José Centeno (St. John's Univ., Jamaica, NY 11439)

This study is about the relation between native Spanish speakers' perception and production of English vowels and their age of arrival (AoA) in the U.S. The participants were also grouped by dialect, Central American/Caribbean or South American Spanish. In a listening task, the participants identified the words in the set "heed, hid, head, had, hod, hawed, hood, who'd, hud, heard" and then produced four repetitions of each of the ten words for spectral and temporal analysis. Finally, they indicated their sensitivity to contrasts between the vowels of the English vowel pairs /i–I/, /I–e/, and /e–æ/ (and /æ–ɜ:/), rating them from "very similar" to "very dissimilar." The results of our acoustic analyses will be compared with the participants' identification data to determine how the vowel space of L1 (primary language) affect the production and perception of L2, and also with the listeners' sensitivity to the vowel–contrast pairs.

4aSC7. Why are non-native (L2) utterances longer than native-produced utterances? James E. Flege and Anders Højen (Univ. of Alabama at Birmingham, CH19 301, 1530 3rd Ave. S., Birmingham, AL 35294-2041)

Research has repeatedly shown that utterances produced by non-native speakers are longer than those of native speakers, even when produced as rapidly as possible. The non-native speaking rate effect is known to be greater for late than early L2 learners, but may disappear for late learners when very short-duration L2 utterances are examined. This study examined the duration of English utterances produced by native English (NE) speakers and late native Spanish (NS) learners of English ($n=20$ each). For normal-rate utterances, durations were significantly longer for the NS than NE group for syntactically complex sentences (e.g., The boy saw that Gary forgot a box of pencils) but not simple sentences (Gary forgot a box of pencils) or noun phrases (a box of pencils). However, the NS-NE difference reached significance for all three utterance types at a fast rate. In Experiment 2, the NS group took longer than the NE group to produce the digit string 1 2 3 4 5 6 7 as rapidly as possible. Taken together, the results suggest that differences in both the articulation of L2 speech sounds and higher-order planning (including the formulation of syntactic representations) contribute to the nonnative speaking rate effect. [Work supported by NIH.]

4aSC8. Perceptual shift of the /d/-t/ voice onset time boundary following exposure to French-accented English. Constance M. Clarke and Paul A. Luce (Dept. of Psychol., Univ. at Buffalo, 245 Park Hall, Buffalo, NY 14260, cclarke2@buffalo.edu)

Previous work suggests native English listeners perceptually adapt to foreign-accented English, as indicated by faster reaction times, after only brief exposure [C. M. Clarke, Proc. ICSLP, 253–256 (2002)]. The present study explored whether changes in the perceptual criteria for phonetic categories may contribute to this increased processing speed. Categorization of /d/ and /t/ was tested using a voice onset time (VOT) continuum. French-accented English was used as the exposure accent because it typically contains prevoiced /d/'s and short-lag /t/'s. Native English has short-lag /d/'s and long-lag /t/'s. Categorization was tested prior to exposure and again following 20, 40, and 60 accented sentences. Each set of 20 sentences contained 20 syllable-initial /t/'s and /d/'s. It was predicted that native English listeners categorization boundaries would shift to lower VOTs after exposure to French-accented speech. Native English speech was used in a control condition. To test the contribution of exposure to specific exemplars of /d/ and /t/ versus exposure to more general characteristics of an accent, two other conditions were included: native English speech with /d/'s and /t/'s digitally modified to be French-like, and French-accented speech with /d/'s and /t/'s digitally modified to be English-like. [Work supported by NIH.]

4aSC9. Accuracy and variability in vowel targets produced by native and non-native speakers of English. Shawn L. Nissen (Dept. of Audiol. and Speech-Lang. Pathol., Brigham Young Univ., 138 TLRB, Provo, UT 84602), Bruce L. Smith (Univ. of Utah, Salt Lake City, UT 84112), Ann Bradlow, and Tessa Bent (Northwestern Univ., Evanston, IL 60208)

The primary issue explored in this investigation concerned how accurately and consistently non-native, adult speakers learning English as a second language produce vowel targets when compared with native speakers. One possibility is that non-native speakers show average formant and duration values similar to those of native speakers but are nonetheless more variable across repetitions. Alternatively, they might be "off-target" but still reasonably consistent in such productions. A group of native speakers (3F and 3M) and a group of Mandarin-speaking non-native subjects who had recently come to the United States (3F and 3M) produced at least ten repetitions of each of 20 different English words embedded in a carrier phrase. The first two formant frequencies and duration were measured for seven different vowels. Preliminary findings suggest that the non-native speakers' vowel formants were generally quite similar to those of the native speakers, whereas their vowel durations deviated from those of the native speakers. Moreover, the non-native speakers exhibited greater within-speaker variability in their vowel productions (across word repetitions) than the native speakers, as well as a greater range in vowel formants and durations across speakers. Findings will be discussed in reference to their implications for second language acquisition and foreign-accented speech perception.

4aSC10. Global foreign accent in English sentences produced by native Spanish learners of English: A 5-year longitudinal study. Satomi Imai, James E. Flege (Div. of Speech and Hearing Sci., Univ. of Alabama at Birmingham, Birmingham, AL 35294), Ratree Wayland (Univ. of Florida at Gainesville, Gainesville, FL 32611), and Surya P. Talla (Univ. of Alabama at Birmingham, Birmingham, AL 35294)

Second language acquisition theorists posit that the performance of adult second language (L2) learners improves up to a certain point but may then fossilize, that is, show little further improvement. However, few empirical studies have tested for changes in L2 production by individual adult L2 learners over an extended period of time. In the present study, sentences produced by adult native Spanish (NS) learners of English over a 5-year period will be evaluated for changes in pronunciation. Fifteen NS adults (mean age=34 years when first tested) who had lived in Birmingham, AL and three native English (NE) controls produced four English sentences at 6-month intervals for 5 years. NE listeners will rate the sentences for foreign accent. One-way repeated measures ANOVAs will be carried out to determine if individual NS participants, or the group as a whole, became less foreign-accented over the study interval. Individual differences in demographic variables that might have affected degree of foreign accent (e.g., age of arrival in the U.S., chronological age of testing, percent English and Spanish use, family status) will be discussed in relation to any improvement in L2 production. [Work supported by NIH.]

4aSC11. Perceived accents and comprehensibility of native and nonnative English. Ayako Ikeno (Robust Speech Processing Group, The Ctr. for Spoken Lang. Res., Dept. of Linguist., Univ. of Colorado at Boulder, CO 80309-0295), John H. L. Hansen, and Barbara Fox (Univ. of Colorado at Boulder)

This study investigates the relation between perceived accentedness and comprehensibility of utterances spoken by native and nonnative English speakers. Native and nonnative English listeners heard six types of stimuli: single words, 2–10-word utterances from native and nonnative English speakers. Listeners classified the stimuli as either native or nonnative English speech and transcribed some utterances. Overall, results from the classification and transcription tasks indicate that perceived native-ness or nonnative-ness does not directly correspond to good or poor comprehensibility, as measured by transcription scores. Some nonnative utterances were classified as native speech by both native and nonnative listeners, although the transcription accuracy was not necessarily high. In other cases, native speech was correctly identified as native; however, dialectal pronunciation variation seemed to contribute to low transcription accuracy in some cases. On the other hand, although the transcriptions were accurate, native speech was sometimes classified as nonnative. In other cases, nonnative utterances were correctly identified as nonnative, and the speech was transcribed accurately. These results suggest that familiarity with certain dialects or accents influences listeners' perception and comprehension for both native and nonnative speech, and that perceived nonnative-ness may not be the main cause for nonnative speech to be less comprehensible.

THURSDAY MORNING, 18 NOVEMBER 2004

ROYAL PALM SALONS 3 & 4, 8:00 A.M. TO
12:00 NOON

Session 4aSP

Signal Processing in Acoustics, Underwater Acoustics, Animal Bioacoustics, Noise, Acoustical Oceanography and Engineering Acoustics: Time Delay Estimation, Localization and Tracking in Acoustics Part I

Christopher O. Tiemann, Chair

Applied Research Laboratory, University of Texas at Austin, P.O. Box 8029, Austin, Texas 78731-8029

Chair's Introduction—8:00

Invited Papers

8:05

4aSP1. An introduction to time-delay estimation, localization, and tracking in Acoustics. Leon H. Sibul (Appl. Res. Lab., Penn State Univ., P.O. Box 30, State College, PA 16804-0030)

Acoustic source localization and tracking has a long, productive history in both passive and active sonar signal processing. More recently, microphone arrays have been used to localize and track air acoustic noise sources. An overview of basic localization and tracking techniques of acoustic radiating sources and scatters is presented. For a wide range of applications, many *ad-hoc* passive and active source localization techniques have been investigated. Concepts from the statistical estimation theory, such as maximum likelihood estimators and Cramér-Rao lower bounds (CRLB), are useful benchmarks for evaluation of these practical *ad hoc* systems. In both air and underwater media, bearing and range estimates are determined by time-delay estimates; hence, properties of the time-delay estimates are of critical importance. In the case the of narrow-band active sonar, localization and tracking is based on time delay, Doppler, and direction of arrival estimation. In the case of wideband sonar and tracking of high-speed air acoustic sources, tracking and range estimates are determined from time-scale dilation estimates. CRLBs for active sonar are determined by signal-to-noise ratio, signal properties of signal ambiguity functions such as effective duration and mean-square bandwidth of the transmitted signal, and array parameters. [This work has been supported by ONR.]

4aSP2. Obtaining optimal time-delay, source localization and tracking estimates in free space and in an ocean waveguide.

Nicholas C. Makris, Purnima Ratilal, Michele Zanolin, and Ioannis Bertsatos (MIT, 77 Massachusetts Ave., Cambridge, MA 02139)

Asymptotic methods for obtaining optimal time-delay, source position and track estimates are derived for measurements in free-space and in an ocean waveguide using nonlinear statistical inference theory. The methods are useful for determining necessary conditions on controllable variables such as signal-to-noise ratio and sample size to attain desired design error thresholds. For free-space time-delay estimation, it will be shown both analytically and intuitively that the matched filter cannot be unbiased and attain minimum variance unless the kurtosis of the signal's energy spectrum exceeds the signal-to-noise ratio [Naftali and Makris, J. Acoust. Soc. Am. **110**, 1917–1930 (2001)]. In an ocean waveguide, it will be shown that signal-to-noise ratios and sample sizes necessary to attain practical localization error thresholds can become prohibitively large when random waveguide inhomogeneities such as internal waves degrade intermodal coherence.

4aSP3. Navigation from time differences of arrival. John L. Spiesberger (Dept. of Earth and Environ. Sci., 240 South 33rd St., Univ. of Pennsylvania, Philadelphia, PA 19104, johnsr@sas.upenn.edu)

A brief history and a few modern developments are given for hyperboliclike navigation systems in which location is estimated from measurements of the difference in the propagation time of a signal from two or more known locations. Today, hyperbolic navigation is used in locating ships, calling animals, GPS receivers, cellular telephones, and other objects. Topics discussed include the minimum number of receivers required for location, the effects of environmental fluctuations and receiver positioning errors on location accuracy, methods for obtaining probability density functions for location, and methods for allowing the speed of signal propagation to vary from path to path. Recent developments indicate that this 67-year-old navigation problem is not mature.

4aSP4. The use of acoustic multipath for localization of sperm whales. Aaron M. Thode (Marine Physical Lab., Scripps Inst. Oceanogr., 9500 Gilman Dr., La Jolla, CA 92093-0238)

Sperm whales produce impulsive sounds, or “clicks,” that are ideally suited for acoustic tracking. If a widely distributed passive acoustic array is available, then the relative time delays from a click's direct path can be used to derive an animal's location. There are many interesting circumstances, however, where deploying a wide-baseline 3D array geometry is either unavailable or impractical. In such situations acoustic reflections of clicks from the ocean surface and bottom can be used to effectively convert a few hydrophones into a wide-aperture array that permits localization in range, depth, and azimuth. In addition acoustic multipath, particularly surface reflections, can also simplify automated detection and tracking procedures, and permit a single hydrophone to be used to distinguish between multiple whales calling simultaneously. Reviewed here is recent research on how multipath effects detected on one or two hydrophones have been used to recover dive profiles of sperm whales in the Gulf of Alaska and the Gulf of Mexico. This information, in turn, has been used to explore whether the animals exploit multipath for their own purposes, to estimate signal directionality, and to observe how sperm whales remove fish from longlines. [Work sponsored by ONR, the Minerals Management Service, and the North Pacific Research Board.]

4aSP5. Passive detection, data association, and localization of marine mammal calls using traditional 3D hyperbolic tracking algorithms. David Moretti, Susan Jarvis, Nancy DiMarzio, and Ron Morrissey (NUWC, Bldg. 1351, Newport, RI 02841)

The Office of Naval Research Marine Mammal Monitoring on Navy Ranges (M3R) project has developed a toolset for passive detection and localization of marine mammals using the existing infrastructure of Navy undersea ranges. The tools are designed to work across a broad class of calls including clicks, sweeps, and whistles. Using vocalizations received by widely spaced omnidirectional hydrophones, the M3R tools can detect and localize marine mammals in real-time. Signal detection algorithms, data association algorithm, and tracking algorithm have been developed, and demonstrated. The tools have also been successfully fielded to conduct extended monitoring of a 500 square nautical mile area located in the Tongue of the Ocean. This presentation will present the passive acoustic monitoring and localization tools developed under M3R. It will also present results of the application of these tools to the real-time detection and tracking as well as extended monitoring of various toothed whale species at the Atlantic Undersea Test and Evaluation Center (AUTEC), Andros Island, Bahamas.

4aSP6. Perception of multimodal time differences in virtual environments. Durand R. Begault and Elizabeth M. Wenzel (Human Information Processing Res. Branch, NASA Ames Res. Ctr., MS 262-2, Moffett Field, CA 94035, durand.r.begault@nasa.gov)

Presentation of multimodal information with a virtual environment human interface can enable teleoperation of remote tools, sensors, and vehicles for space applications. However, the performance advantages of providing multimodal information can be potentially reduced by intermodal timing asynchronies between different rendering systems (system–system asynchrony). Additionally, asynchronies between the rendering system and proprioceptive, vestibular, and kinesthetic cues that are produced by the actions of the human user can also be problematic (human-system asynchrony). Preliminary data on tolerable asynchrony for audio-haptic interfaces for virtual environments is presented, along with data relevant to perceptually acceptable update rates and latency to minimize computational and perceptual artifacts within virtual auditory displays. [Work supported by the Human Measurement and Performance Project within NASA's Airspace Systems Program.]

4aSP7. Dynamics of bullfrog choruses studied by time-delay localization with an acoustic array. James A. Simmons (Dept. of Neurosci., Brown Univ., Providence, RI 02912) and Andrea M. Simmons (Brown Univ., Providence, RI 02912)

Understanding the dynamics of bullfrog choruses, particularly the occurrence of acoustic interactions among the calling males, requires segregating and characterizing the sequences of vocalizations from individual frogs making up the chorus night after night for a calling season lasting almost a month. Chorusing takes place in the dark, often at ponds surrounded by vegetation, in conditions where visual observation to locate calling individuals is rarely possible. To record choruses, we used two acoustic sensor units, each consisting of four electret condenser microphones mounted on the faces of a 3-cm cube, located 5–10 m apart and aligned alongside a pond. Signals were stored on an 8-channel digital instrumentation recorder. These signals were used to estimate the horizontal directions and, from these, the locations of multiple bullfrogs (*Rana catesbeiana*) vocalizing at night. The array's mixture of short and long apertures was conceived for acoustic surveying to avoid problems caused by variations in acoustic propagation due to wind and vegetation, and to avoid ambiguities caused by multiple overlapping signals detected only over long apertures. Several kinds of acoustic interactions are evident when calls are assigned to individual animals. [Work supported by DARPA, ONR, NIH.]

Contributed Papers

11:15

4aSP8. Time-delay estimation in underwater acoustics: A global optimization approach. Zoi-Heleni Michalopoulou (Dept. of Mathematical Sci., New Jersey Inst. of Technol., Newark, NJ 07102) and Michele Picarelli (Saint Peter's College, Jersey City, NJ 07306)

Arrival times and amplitudes of individual ray paths provide significant information for source localization and geoacoustic inversion in underwater acoustics. Considered in a statistical framework, estimates for such quantities can be obtained via maximum likelihood or maximum *a posteriori* methods. These approaches, however, require function maximization on a multidimensional space, demanding extensive computations depending on the number of distinct paths. Local search methods have been proposed as computationally low-cost alternatives; such techniques, however, often get trapped in local maxima. In this work, a global optimization approach is proposed for the maximization of the posterior probability distribution of arrival times and amplitudes. To avoid an exhaustive search, an estimate of the posterior distribution is obtained employing Gibbs sampling; it is, subsequently, straightforward to obtain values for time delays and amplitudes for which the distribution attains a maximum. Simulations show that the new method provides excellent estimates that are comparable to those of the analytical maximum *a posteriori* approach. Real data results from the Haro Strait Experiment further validate the method. [Work supported by ONR.]

11:30

4aSP9. Concurrent visual and acoustic tracking of fin whales offshore Southern California. Sean M. Wiggins, Allan W. Sauter, John A. Hildebrand (Scripps Inst. of Oceanogr., La Jolla, CA 92093-0205), Ahmad T. Abawi, Michael B. Porter, Paul Hursky (Sci. Applications Intl. Corp., San Diego, CA), and John Calambokidis (Cascadia Res., Olympia, WA)

Ocean acoustic techniques can be used to monitor whale presence within a region over long time periods. However, using the number of recorded calls to provide an estimate of the number of total whales present has not yet been realized. To help provide a transfer function from the

number of acoustic calls to the number of whales present, we conducted a 10-day, concurrent visual and acoustic experiment aboard the Floating Instrument Platform (FLIP) focusing primarily on fin whale monitoring. FLIP is a stable platform with a visual observation deck approximately 25 m above the sea surface providing excellent visual range. Suspended beneath FLIP at 90 m below the sea surface was a 105-m-long acoustic array consisting of 8 hydrophones, 15 m apart. FLIP was placed in a stationary 3-point mooring centered above three seafloor-mounted hydrophone recorders at depths between 250 and 400 m. The FLIP vertical and seafloor horizontal arrays provide excellent geometry for acoustically tracking whales via arrival-time difference and model-based techniques. These fin whale call tracks are compared to the visually observed fin whale surfacing to provide call-to-whale relationships.

11:45

4aSP10. Location and range of calling blue and fin whales off the Western Antarctic Peninsula. Ana Sirović, John A. Hildebrand, Sean M. Wiggins (Scripps Inst. of Oceanogr., 9500 Gilman Dr., MC 0205, La Jolla, CA 92093-0205), Mark A. McDonald (Whale Acoust., Bellvue, CO 80512), Sue E. Moore (Univ. of Washington, Seattle, WA 98105), and Deborah Thiele (Deakin Univ., Warrnambool VIC 3280, Australia)

Blue and fin whales were heavily whaled from the Southern Ocean during the 20th century and their populations today remain at extremely low levels. Both species produce loud, low-frequency calls that are well suited for long propagation. Eight acoustic recording packages (ARPs) were deployed off the Western Antarctic Peninsula, with instrument spacing from 100 to 180 km. Blue and fin whale calls were commonly heard and two different methods were used to determine locations and ranges to the calling whales. Blue whales were localized using the differences in call arrival times to three different instruments. It also was possible to calculate ranges to blue whales from multipath arrivals of their calls and verify the results obtained by the difference in arrival times method. Short detection ranges for fin whales precluded localization, but multipath arrival difference allowed range estimation. Source levels for both species were calculated using these ranges and the measured received levels.

Session 4aUW

Underwater Acoustics, Signal Processing in Acoustics and Acoustical Oceanography: Long-Range Acoustic Propagation Part I

Peter F. Worcester, Chair

*Scripps Institution of Oceanography, University of California San Diego, 9500 Gilman Drive,
La Jolla, California 92093-0225*

Chair's Introduction—8:15

Invited Papers

8:20

4aUW1. Fifteen years of long-range propagation experiments in the North Pacific. Robert C. Spindel (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, spindel@apl.washington.edu) and Peter F. Worcester (Scripps Inst. of Oceanogr., La Jolla, CA 92093)

Over the last 15 years a series of long-range acoustic propagation experiments has been conducted in the North Pacific Ocean using various combinations of low-frequency, wide bandwidth transmitters and horizontal and vertical line array receivers, including a two-dimensional array with a maximum vertical aperture of 1400 m and a horizontal aperture of 3600 m. These measurements were undertaken to further our understanding of the physics of low-frequency, broadband propagation and the effects of environmental variability on signal stability and coherence. The first experiment in this series, which was conducted in 1989, combined a broadband, 250-Hz transmitter with a 3-km-long vertical receiving array 1000 km distant. The transmissions revealed a traditional SOFAR arrival pattern of steep, raylike time fronts in the early portion building to a crescendo of near-axial arrivals that cut off abruptly. The near-axial finale was found to be highly scattered, however, and included significant energy in the geometric shadow zone. The prevailing theory of scattering from internal-wave-induced sound-speed fluctuations predicted far more resolvable arrivals than were observed. These observations stimulated an intense research effort to reconcile data with theory. In this presentation we summarize the central issues these experiments have addressed, and identify the many still remaining unanswered questions.

8:45

4aUW2. Experimental measurements of normal-mode statistics at megameter ranges. Kathleen Wage (ECE Dept., George Mason Univ., 4400 Univ. Dr., MSN 1G5, Fairfax, VA 22030, kwage@gmu.edu)

The Acoustic Thermometry of Ocean Climate, Alternate Source Test, and North Pacific Acoustic Laboratory experiments generated extensive vertical array data sets for studying normal-mode signals at megameter ranges. These data sets include receptions for 28-, 75-, and 84-Hz sources at ranges of 3.5 and 5 Mm. Recent analysis of the low-mode arrivals in these receptions produced estimates of important mode statistics, including the modal time spread, cross-mode coherence, and temporal coherence. This talk will present the experimental results and place them in context, making comparisons to theoretical predictions about long-range mode propagation and to numerical simulations. The implications for future research will be discussed. [Work sponsored by an ONR Ocean Acoustics Entry-Level Faculty Award.]

Contributed Papers

9:10

4aUW3. Analysis of multipath acoustic field variability and coherence in the finale of broadband basin-scale transmissions in the North Pacific Ocean. John Colosi (Woods Hole Oceanogr. Inst., Woods Hole, MA 02543)

The statistics of low-frequency, long-range acoustic transmissions in the North Pacific Ocean are presented. Broadband signals at center frequencies of 28, 75, and 84 Hz are analyzed at propagation ranges of 3252 to 5171 km, and transmissions were received on 700- and 1400-m-long vertical receiver arrays with 35-m hydrophone spacing. The analysis focuses on the energetic "finale" region of the broadband timefront arrival pattern where a multipath interference pattern exists. Two-dimensional (depth and time) phase unwrapping is employed to study separately the complex field phase and intensity.

9:25

4aUW4. Spatial coherence of acoustic signals measured during the 1998–1999 North Pacific Acoustic Laboratory (NPAL) experiment. Vladimir E. Ostashev and Alexander G. Voronovich (NOAA/Environ. Technol. Lab., 325 Broadway, Boulder, CO 80305), and the NPAL Group (J.A. Colosi, B.D. Cornuelle, B.D. Dushaw, M.A. Dzieciuch, B.M. Howe, J. A. Mercer, W.H. Munk, R.C. Spindel, and P.F. Worcester)

Little experimental data have been reported in the literature about spatial coherence of low-frequency sound waves propagating over megameter ranges in the ocean. In 1998–1999, the North Pacific Acoustic Laboratory (NPAL) carried out a comprehensive low-frequency long range sound propagation experiment with the use of the receiving billboard acoustic array. By signal-processing of the NPAL data, we calculated spatial coherence of the broadband acoustic signals (with 75 Hz carrier) after their propagation of about four thousand km in North Pacific. The results ob-

tained are presented in this paper. First, the horizontal coherence of the NPAL signals is discussed. The results show that the horizontal coherence radius varies in the range 500–1000 m. This value is in a good agreement with predictions of a 3D modal theory of low-frequency long-range sound propagation in a fluctuating ocean. Secondly, the vertical coherence of the NPAL signals is presented. After averaging over several days of the experiment, the vertical coherence reveals non-monotonic dependence on the distance between hydrophones. Finally, the horizontal-vertical coherence is studied. This coherence rapidly degrades with the increase of vertical separation between hydrophones. [Work supported by ONR.]

9:40

4aUW5. Transverse horizontal spatial coherence of deep arrivals at megameter ranges. Rex K. Andrew, Bruce M. Howe, and James A. Mercer (Appl. Phys. Lab., 1013 NE 40th St., Seattle, WA 98105)

Predictions of transverse horizontal spatial coherence from path integral theory are tested for ranges from 2000 to 3000 km. A low-frequency (75 Hz) source bottom-mounted at depth 820 m near Kauai transmitted “m-sequence” signals over several years to bottom-mounted U.S. Navy horizontal line arrays in the North Pacific. The arrays acquired only the early, deep arrivals. Coherence lengths estimated from the datasets, on the order of 400 m, show good agreement with lengths calculated from theory. Both measurements and predictions show a dependence on path as well as path length. These coherence lengths correspond via a simple incoherent source model to about 1° in horizontal arrival angle variability. Estimates of scintillation index, log-amplitude variance, and variance of decibel intensity indicate that the fields were partially saturated. There is no significant seasonal variability in these measures. Scintillation index predictions from path integral theory also agree well, although theory identifies these fields as fully saturated. [Work supported by ONR.]

9:55–10:10 Break

Invited Paper

10:10

4aUW6. α , β , and wave-field statistics. Michael G. Brown, Francisco J. Beron-Vera, Irina Rypina, and Ilya A. Udovydchenkov (RSMAS-AMP, Univ. of Miami, 4600 Rickenbacker Cswy., Miami, FL 33149)

Recent analysis of measurements made in the AET experiment, in which 75 ± 15 Hz signals propagated a distance of 3.25 Mm in the eastern North Pacific Ocean, show that long-held assumptions and theoretical ideas about wave-field statistics at long range require significant revision. Progress towards developing a new theory is discussed in this talk. In contrast to the older theoretical ideas, the background sound-speed structure plays an active and critically important role in the new theory, while details of the perturbation field play only a minor role. The importance of the background sound-speed structure is quantified by the ray-based stability parameter α , or the asymptotically equivalent mode-based waveguide invariant β . [Work supported by ONR and NSF.]

Contributed Papers

10:35

4aUW7. Comparison between ocean-acoustic fluctuations in parabolic-equation simulations and estimates from integral approximations. Stanley M. Flatté (Phys. Dept., Univ. of Calif., Santa Cruz, CA 95064) and Michael D. Vera (Univ. of Southern Mississippi, Long Beach, MS 39560)

Line-integral approximations to the acoustic path integral have been used to estimate fluctuations due to internal waves. Approximations for the root-mean-square (rms) fluctuation and the bias of travel time, rms fluctuation in vertical arrival angle, and the spreading of the acoustic pulse out to 1000-km range are here compared to estimates from simulations that use the parabolic equation (PE). Integral-approximation (IA) estimates of rms travel-time fluctuations were within statistical uncertainty at 1000 km for the Slice89 profile, and in disagreement by between 20% and 60% for the canonical profile. Bias estimates were accurate for the first few hundred kilometers of propagation, but often disagreed beyond. The PE structure functions of travel time with depth were quadratic for vertical separations of 20 m or less, in qualitative agreement with the IA prediction. Pulse spreads in the PE results were much smaller than predicted by the IA estimates. Implications of these results will be discussed.

10:50

4aUW8. Eliminating unnecessary diffractive features in ocean sound-speed models for long-range acoustic propagation. Katherine C. Hegewisch, Nicholas R. Cerruti, and Steven Tomsovic (Dept. of Phys., Washington State Univ., Pullman, WA 99164-2814, khegewisch@wsu.edu)

A major focus of long-range acoustic experiments is the study of the properties of the ocean. In order to explore these properties, it is necessary to have an effective model of the ocean that is sufficient to understand the physics of the wave propagation. Large structures in the ocean cause re-

fraction of the acoustic waves, while smaller structures cause diffraction. Since long-range time fronts are mainly due to refractive effects, adding diffractive features to a model does not significantly change the long-range wave propagation. However, such features cause important theoretical techniques based on refraction, e.g., ray methods, to be inaccurate and they also increase computational effort in the wave propagation calculations. A prescription to remove the diffractive features in an ocean sound-speed model is outlined and the effects on the time fronts is shown. This removal eliminates the “microfolds” in the phase space manifolds [discussed by Simmens *et al.* in J. Acoust. Soc. Am. **102**, 239 (1997)] and improves the wave/ray correspondence. Therefore, it is necessary to remove diffractive structures from ocean models before attempting to use ray methods to understand long-range propagation. [Work supported by ONR.]

11:05

4aUW9. Acoustic beam dynamics in deep ocean environments. Francisco J. Beron-Vera, Ilya A. Udovydchenkov, Irina Rypina, and Michael G. Brown (RSMAS-AMP, Univ. of Miami, 4600 Rickenbacker Cswy., Miami FL 33149)

Parabolic equation simulations of broad-band directional beams in highly structured deep ocean environments are used to illustrate the important role played by the background sound speed structure on controlling wavefield structure and stability, and to test theoretical predictions of time spreads. The predictions are based on both ray- and mode-based theoretical considerations, in which the stability parameter α and the waveguide invariant β , respectively, play important roles. [Work supported by ONR and NSF.]

4aUW10. Mean field of a sound wave propagating in an oceanic waveguide with random inhomogeneities. Alexander G. Voronovich and Vladimir E. Ostashev (NOAA/Environ. Technol. Lab., 325 Broadway, Boulder, CO 80305)

The mean field is an important statistical characteristic of a sound wave propagating in an oceanic waveguide with random inhomogeneities such as internal gravity waves. A closed equation for the mean sound field propagating in a 3D random waveguide can be obtained with the use of the diagram technique. Using the Fourier transform, this equation is reduced to a one-dimensional integro-differential equation with respect to the vertical coordinate. The latter equation determines acoustic modes which are generally different from those in a regular waveguide. For low frequency sound waves, the integro-differential equation is solved approximately. This results in an explicit formula for the mean sound field as a sum of modes which attenuate exponentially with the propagation distance. The extinction coefficient of each mode is linearly related to the spectrum of random inhomogeneities. Therefore, measuring the extinction coefficients for different frequencies and different mode numbers, one can try to reconstruct this spectrum. The mean field of a sound wave propagating in a 2D random waveguide is also calculated. The result obtained is compared with that for a 3D random waveguide. This allows us to study the range of applicability of 2D approximation.

4aUW11. Interference wave associated with a multitude of cusped caustics in long-range acoustic propagation in the ocean. Natalie S. Grigorieva and Gregory M. Fridman (Dept. of Appl. Math. and Math. Modeling, St. Petersburg State Marine Tech. Univ., 3 Lotsmanskaya Str., St. Petersburg, 190008, Russia, nsgrig@natalie.spb.su)

In long-range propagation experiments, the observed time-of-arrival patterns show early geometrical-like arrivals followed by a crescendo of energy that propagates along the sound-channel axis and is not resolved into individual arrivals. In these experiments the sound source and receiver are placed close to the depth of the axis. As a result, rays emerging from the source at sufficiently small angles intersect the axis many times and form in its vicinity a large number of caustics with caustic cusps located repeatedly along the axis. It is well known that a description in terms of geometrical acoustics is not valid in neighborhoods of cusped caustics because such a description presupposes that the waves associated with individual ray paths do not interfere with one another. These neighborhoods of interference grow with range and at long ranges they overlap. The interference of the wave fields that correspond to near-axial rays results in the formation of a complex interference wave, or axial wave, that propagates along the waveguide axis being a component of the axial crescendo. In this paper for a range-independent ocean the axial wave formation is discussed as well as its main properties in the frequency and time domains. [Work supported by ONR.]

THURSDAY AFTERNOON, 18 NOVEMBER 2004

PACIFIC SALON 2, 1:00 TO 4:45 P.M.

Session 4pAAa

Architectural Acoustics, Noise, Psychological and Physiological Acoustics, Speech Communication and Committee on Standards; Implementation of Classroom Acoustics II

Lily Wang, Chair

Architectural Engineering, University of Nebraska Lincoln, Omaha, Nebraska 68182

Contributed Papers

1:00

4pAAa1. A case study: School band and practice rooms in Harrington Park (NJ) school. Daniel R. Raichel (Eilar Assoc. and CUNY Grad. Ctr., 2727 Moore Ln., Ft. Collins, CO 80526, draichel@comcast.net)

Several years ago, the Harrington Park (NJ) Board of Education constructed a new gym/cafeteria addition and converted the old multipurpose room in the elementary school (K-8) into a band practice room with two adjacent individual practice rooms. In an attempt to allay the sound level of the band, which can reach as high as 102 dB(A), sound panels having an effective absorption coefficient of 0.74 were installed over approximately 20% of the plaster wall surfaces, resulting in little acoustic effect. An analysis indicates that using wall and ceiling panels with a higher values of absorption coefficients over the entire wall surface and the ceiling would bring the sound level down by only 3 dB, insufficient to bring the prevailing SPLs to more reasonable levels, due to the band's sound power. However, the reverberation time of 0.8 s would be reduced to 0.3 s, which would allow the music to be more discernible. The walls and the doors separating the individual practice rooms from the band room proved to be ineffective in isolating these rooms acoustically. It was recommended that modular practice rooms be purchased and installed within these two practice rooms, with ventilation ducts leading outside the music complex.

1:15

4pAAa2. Acoustical design of Kumamoto Health Science University. Masahiro Ikeda (Yamaha Corp. of America, 6600 Orangethorpe Ave., Buena Park, CA 90620, miked@yamaha.com)

Kumamoto Health Science University was built in 2002. The key of the project is how to include variety of facilities required for medical school within a limited cost. For this purpose, a large flat circle formed with folded metal plate was adopted. The circle roof covers the rooms, which is located on the ground, except a gymnasium. The prior acoustical concern was the noise caused by rain drops on the metal roof. Through two times experiments carried out in a design and a construction stage, three types of the details were utilized depending on the purposes of the rooms. The effect of inserted rubber, which is cost effective compared to laminated damping steel sheets, is 10 dB in the structure borne transmission loss. Regarding to the room acoustics, the ceilings of lecture rooms are designed to reflect frontal reflection from talkers effectively to the seating area. Acoustical panels are located at a higher wall part instead of the ceiling. Geometrical simulation showed the effect of the acoustical design. The interior of the gymnasium was acoustically designed for utilization of ceremony and assembly, while keeping a liveness suitable for a sports utilization. The highly designed campus was realized in low cost.

1:30

4pAAa3. Speech intelligibility in real and virtual classrooms.

Wonyoung Yang and Murray Hodgson (Rm. 313, Univ. of BC, 2206 East Mall, Vancouver, BC V6T 1Z3, Canada)

In current research by the authors, auralization is being used to identify the optimal reverberation times and signal-to-noise level differences for speech in classrooms. The work presented here was initiated to validate the auralization procedure in comparison with live listening tests in real classrooms. Reverberation and noise conditions were created in both real and virtual classrooms, and speech intelligibility tests performed. Two architecturally identical classrooms which have different amounts of sound absorption were selected for real classroom speech intelligibility tests, and their acoustical parameters were measured. Speech and noise sources were placed at the fronts and backs of the classrooms. Speech intelligibility tests were performed at three positions with normal-hearing subjects. The modified rhyme test (MRT) and the noise babble were generated by loudspeakers. The classrooms and speech intelligibility test conditions were then simulated for auralization using CATT-Acoustics, based on the real classroom measurement results. Virtual speech intelligibility tests were performed on the same subjects in the laboratory using headphones. The MRT results in the real and virtual classrooms were compared.

1:45

4pAAa4. Acoustics in a high school library.

Sergio Beristain (P.O. Box 12-1022, Narvarte, 03001, Mexico D. F., Mexico, sberista@hotmail.com)

A new and up-to-date library is under development within a Mexican high school educational institution. Usually, libraries are supposed to be quiet places where people can read without disturbance, but nowadays with the new data storage technologies and their capabilities for storing much more than only text, where people can interact with the material

under study, for instance, read the description of the words Water Fall, see the water sparkling, and hear the sound of the fall, plus the sound of the computers and projection equipment, libraries tend to be more and more noisy. The approach taken for this project includes separate areas and acoustic absorption material distribution for different activities within the library, such as reading, computer consultation of texts, and computing consultation of multimedia stored books either in DVD format or by the network.

2:00

4pAAa5. Articulation tests of the first and the second languages in classrooms.

Coriolanus C. L. Lam, Kai Ming Li, and Stanley M. L. Cheung (Dept. of Mech. Eng., The Hong Kong Polytechnic Univ., Hung Hom, Hong Kong)

The current study is motivated by the needs of improving acoustic environments in classrooms for the primary and secondary schools in Hong Kong because one of the main education policies is to enable pupils be biliterate and trilingual. Being biliterate means that the pupils are expected to write good English and Chinese. Being trilingual requires that pupils be able to speak English, Cantonese (dialect spoken in Hong Kong) and Mandarin (dialect spoken in China). There are many related studies investigating the speech intelligibility in both English and Mandarin. However, these studies are targeted at the respective native listeners. On the other hand, there are few research studies using Cantonese as the first language of listeners. A series of articulation tests, which were based on the phonetical characteristics of these different languages, was carried out for university students in Hong Kong. Other relevant parameters, such as the speech transmission index, reverberation time and signal-to-noise ratio, were also examined. The outcomes of the present study will provide a basis for further investigations involving pupils of primary and secondary schools. [Work supported by the Research Grants Council of the Hong SAR Government and the Hong Kong Polytechnic University.]

2:15–2:45 Panel Discussion

2:45–4:45 Brainstorming Session

THURSDAY AFTERNOON, 18 NOVEMBER 2004

PACIFIC SALONS 6 & 7, 3:00 P.M. TO 5:05 P.M.

Session 4pAAb

Architectural Acoustics, Speech Communication and Committee on Standards: Speech in Architectural Spaces—Both Intelligibility and Privacy I

Kenneth P. Roy, Cochair

Armstrong World Industries, Innovation Center, 2500 Columbia Avenue, Lancaster, Pennsylvania 17604

Peter A. Mapp, Cochair

Mapp Associates, 5 Worthington Way, Colchester CO3 4JZ United Kingdom

Chair's Introduction—3:00

Invited Papers

3:05

4pAAb1. An evaluation of source speakers for open-plan privacy measurements. Kenneth W. Good, Jr. and Kenneth P. Roy, Ph.D. (Armstrong Innovation Ctr., 2500, Columbia Ave, Lancaster, PA 17601)

A variety of source speakers is used today when making field measurements of interzone attenuation in open-plan spaces for the purpose of calculating the privacy index. These speakers have a variety of performance characteristics and often do not meet ASTM E 1179. The question is, do these variations matter more or less for field or laboratory measurements? In this study we look at the speaker performance related to their impact on interzone attenuation in field measurements. An omnidirectional source speaker, a speaker meeting ASTM E 1179, and another typically used speaker are compared.

3:25

4pAAb2. Sound fields and speech intelligibility in rooms. Kenneth P. Roy (Armstrong Innovation Ctr., 2500 Columbia Ave., Lancaster, PA 17604), John Stowell, Mike Walter, and Mathew Schuepbach (Univ. of Kansas, Lawrence, KS 66045)

A key objective in the architectural design of buildings is to choose building elements that provide both speech intelligibility within a space, and speech privacy between spaces. Standard measurement and evaluation methods per ASTM standards on speech intelligibility and privacy will be discussed as they relate to field measurements. Field data on the reverberant sound fall-off with distance from the source will be presented for a range of small to mid-sized rooms. The variation in reverberant sound within a space can result in an estimation error for the AI and PI between rooms. A possible approach to resolving this issue will be discussed.

3:45

4pAAb3. Assessment of Architectural speech security of closed offices and meeting rooms. Bradford N. Gover and John S. Bradley (Inst. for Res. in Construction, Natl. Res. Council, Ottawa, ON K1A 0R6, Canada, brad.gover@nrc-cnrc.gc.ca)

The degree to which conversations taking place within closed offices and meeting rooms are audible or intelligible to a listener outside the room depends on the listener's hearing ability, the talker's voice level, the sound attenuation afforded by the building, and the background noise at the position of the listener. Sensible assumptions can be made about the listener's ability so that signal-to-noise measures can be used to reliably indicate audibility or intelligibility. A survey of actual meetings was conducted to acquire estimates of typical voice and noise levels, which can be used in the prediction of security. In existing buildings, the background noise can be measured, leaving only the need to measure sound attenuation through the building. A new measurement procedure has been developed that accurately indicates the attenuation from within a source room to a single receiving point outside the room. The results of this procedure can be used to derive a reliable indicator of the level of speech security at the receiving point. This procedure is capable of handling localized weak spots ("leaks" or "hot spots") in the partitions. An overview of the measurement and assessment procedures will be discussed.

4:05

4pAAb4. Speech privacy for healthcare. Joel A. Lewitz (Lewitz and Assoc. Inc., 1505 Bridgeway, Ste. 128, Sausalito, CA 94965)

The Healthcare Insurance Portability and Accountability Act, also known as HIPAA, requires healthcare providers to implement reasonable safeguards to protect the privacy of healthcare information. This includes privacy for oral communications. Architects and facility managers are aware of this mandate, and are calling for these capabilities to be designed into current projects. HIPAA contains no objective, measurable criteria for speech privacy, leaving the design team to select specific criteria for their project. Criteria options will be discussed including Privacy Index and other descriptors that have been developed and correlated with subjective evaluation of privacy. This paper describes several recent projects, which began with evaluations of existing facilities and led to establishment of criteria and design solutions. These studies included various aspects of the healthcare facility such as offices, exam rooms and conference rooms as well as open plan reception, waiting and registration areas. Field conditions, measurement techniques, measurement results, criteria selection and design solutions will be presented.

4:25

4pAAb5. A case study of speech privacy in healthcare and other closed-plan professional spaces. Kenneth W. Good, Jr. (P.O. Box 595, Maytown, PA 17550-0595)

Speech privacy is of growing importance in everyday life. Most folks assume that their conversations will be confidential when the door is closed. The need for the measurement and rating of a room's performance is driven by both consumer demand for respect of privacy, and by regulations such as HIPAA and GLBA. This case study will present field measurements of various professional spaces such as healthcare counseling, medical treatment rooms, and pastors' offices and places of counseling in ministry.

4:45

4pAAb6. Application of intelligibility metrics to privacy measurements. Steven McManus (Gold Line/TEF Box 500, W. Redding, CT 06896, steve.tech@bigfoot.com)

The scale of perceived intelligibility runs from the easily understood, down through hard to understand, to private, and finally confidential. When attempting to use STI to measure privacy, it is found that the currently published translations to Articulation Index and therefore Privacy Index do not correlate well with subjective experience at the lower end of the scale. The transition from privacy, where the aim is not to be disturbed, and confidentiality, where the aim is not to be able to eavesdrop, is very rapid. Here, a study is presented that proposes a new mapping from STI to AI and therefore PI, in the low intelligibility range using the accepted threshold of 95% PI as the boundary for confidentiality. This new mapping deviates from the mathematical relationship between STI and AI, which is based solely on signal to noise ratio.

Session 4pAB

Animal Bioacoustics: Acoustic Monitoring of Large Whales II—The Legacy of Paul Thompson

Sue E. Moore, Chair

National Marine Mammal Laboratory, 7600 Sand Point Way, NE, Seattle, Washington 98115

Contributed Papers

1:25

4pAB1. Detection of a unique calling pattern during the respiratory cycle of blue whale pairs. Catherine Berchok (Grad. Program in Acoust., Penn State Univ., P.O. Box 30, State College, PA 16804), David Bradley, Thomas Gabrielson (Penn State Univ., State College, PA 16804), and Richard Sears (MICS Inc, 285 rue green, St. Lambert, QC J4P1T3, Canada)

Although blue whales are known for their long (10–20 s) low-frequency (<20 Hz) vocalizations that occur in regularly patterned series, in coastal waters they produce short (1–4 s) higher-frequency (30–150 Hz) unpatterned calls just as often. Paul Thompson was among the group of scientists who first attributed these short calls to blue whales. They found the majority of these calls to be downsweped in frequency but noted the presence of others that were not. This paper describes a special patterning of these non-downsweped sounds that was detected in the St. Lawrence estuary in both 2000 and 2001. This patterning is variable but normally includes both regularly and irregularly spaced components. In at least a dozen cases, these mixed-pattern bouts were detected in the presence of a pair of blue whales at the surface for their respiratory cycle. Characterization of these bouts and their daily, seasonal, and spatial distributions will be presented and compared with the distributions of the short downsweps and the long low-frequency calls. [Work supported by the Applied Research Lab (Penn State), NSF, and the American Museum of Natural History.]

1:40

4pAB2. Insights into underwater behavior and vocalizations of blue and fin whales from attachments of acoustic recording tags. John Calambokidis (Cascadia Res. Collective, Olympia, WA 98501), Erin M. Oleson (Scripps Inst. of Oceanogr., La Jolla, CA 92037), William C. Burgess (Greeneridge Sci., Inc., Goleta, CA 93117), Mark A. McDonald (Whale Acoust., Bellvue, CO 80512), and John A. Hildebrand (Scripps Inst. of Oceanogr., La Jolla, CA 92037)

The underwater behavior and vocalizations of blue and fin whales in the northeast Pacific were examined using three types of electronic archival tags attached with suction cups. The bulk of the attachments were of the “Bioacoustic Probe,” a commercially available acoustic recording tag developed by Greeneridge Sciences under ONR support. We completed more than 30 deployments of this tag between 2002 and 2004 at multiple locations off California and in the Sea of Cortez, Mexico. Acoustic data recorded by the tags were used to investigate: (1) vocalizations made by the tagged subject whale and nearby animals; (2) anthropogenic noise to which the subject was exposed, including airguns and vessel noise; and (3) the variability of flow noise, which indicated the relative speed of the subject through the water. Results to date have revealed a number of new findings on blue whale vocalizations, including that only a small proportion of blue whales are producing regular calls, that more blue whales may be producing calls irregularly than regularly, and that callers are males. The new data have also clarified the behavioral context in which calling takes place. [Work supported by ONR, SERDP, CNO N45.]

1:55

4pAB3. Techniques for separating interfering blue and fin whale calls. Paul Hursky, Ahmad T. Abawi, Michael B. Porter (Ctr. for Ocean Res., SAIC, San Diego, CA 92121), John A. Hildebrand, Sean M. Wiggins, and Allan W. Sauter (Univ. of Calif. in San Diego, La Jolla, CA 92093)

Recordings of blue and fin whales were made at the Southern California Offshore Range (SCORE) in August 2003. In analyzing this data, previously successful tracking algorithms, based on time differences of arrival (TDOA) measured on geographically distributed hydrophones, were frustrated by the mutual interference between the shorter, broadband, but higher amplitude fin whale calls and the longer duration, narrow-band, blue whale calls. We present a series of progressively more powerful detectors designed to isolate these two types of calls, by zeroing in on specific features of these calls. Because the limited bandwidth of both types of calls presents a challenge to time delay estimation, we also present a comparison of a number of TDOA estimators, including correlation of suitably isolated time-series and spectral features. We apply these techniques to data from different years and geographic sites to assess previous anecdotal evidence of strikingly similar calls within groups of blue whales, despite the presumably wide range of physical characteristics among individuals within each group.

2:10

4pAB4. Vocalizations of blue whales in the Antarctic: Implications for the use of passive acoustics for population studies on baleen whales. Shannon Rankin (Southwest Fisheries Sci. Ctr., NMFS, NOAA, 8604 La Jolla Shores Dr., La Jolla, CA 92037, rankin_shannon@hotmail.com), Don Ljungblad (Ljungblad Assoc., Elk Mountain, WY 82324), Chris Clark (Cornell Univ., Ithaca, NY 14850), and Hidehiro Kato (Natl. Res. Inst. of Far Seas Fisheries, Shimizu, Japan)

Stereotypical calls of baleen whales can be a diagnostic field characteristic for determining species and stocks. Repeated series of these calls may be used for long-term population studies using passive acoustics. Vocalizations were recorded from 15 groups of blue whales (consisting of 42 individuals) during the 2001–02 and 2002–03 International Whaling Commission-Southern Ocean Whale and Ecosystem Research (IWC-SOWER) surveys in Antarctic waters north and east of the Ross Sea. Long-duration vocalizations attributed to Antarctic blue whales included 28-Hz “tones,” 28 Hz “downsweps,” and the more complex “3-unit” calls (considered to be song units). The 28-Hz component was stable for all three calls, regardless of signal strength. The 3-unit calls varied in their 2nd and 3rd units and were heard in only 10 out of the 15 groups. There was no evidence of repeated patterns (song phrases) of 3-unit calls. The results of this study suggest that the 28-Hz peak frequency may be used as a diagnostic feature for acoustic identification of Antarctic blue whales in the field. Additional research on the gender, group size, and associated behavior of these animals is necessary to ground-truth long-term acoustic population studies of baleen whales.

4p THU. PM

4pAB5. Studies of waveguide propagation effects on frequency properties of sperm whale communication signals. Natalia A. Sidorovskaia (Phys. Dept., Univ. of Louisiana at Lafayette, UL BOX 44210, Lafayette, LA 70504, nsidorovskaia@louisiana.edu), George E. Ioup, and Juliette W. Ioup (Univ. of New Orleans, New Orleans, LA 70148)

The Littoral Acoustic Demonstration Center (LADC) conducted a series of passive acoustic experiments for studying ambient noise environment and marine mammal (namely, sperm whales) acoustic phonations in the Northern Gulf of Mexico by long-term autonomous recordings of broadband acoustic signals on a single bottom-moored hydrophone. It was previously reported that extensive spectrogram analysis of sperm whale phonations revealed persistent null pattern in animal spectral signatures that could be associated with either waveguide propagation effects or with a uniqueness of an animal phonation apparatus. The previous modeling addressed the propagation effects on echolocation signals of deep-water foraging animals. It was demonstrated that the null patterns in the analyzed spectrograms could not be explained by the propagation effects for an animal echolocating at foraging depths. This paper presents the results of modeling waveguide propagation effects on spectral properties of sperm whale near-surface communication signals, codas, that could be associated with distinctive sequences of clicks repeatedly present in the LADC recordings. The possibility of the individualization of sperm whale phonations based on the frequency properties of click recordings on a single hydrophone is discussed. [Research supported by ONR.]

2:40

4pAB6. Variation in acoustic detection distances of delphinid whistles using a towed hydrophone array in several geographic areas. Julie N. Oswald (Scripps Inst. of Oceanogr., Univ. of California San Diego, La Jolla, CA 92093, joswald@ucsd.edu), Shannon Rankin, and Jay Barlow (NOAA, La Jolla, CA 92037)

Dolphins were detected using a towed hydrophone array during cetacean abundance surveys in the eastern tropical Pacific (ETP) and in U.S. west coast waters up to 300 nmi offshore (USWC). Cross-correlation algorithms were used to estimate bearing angles to vocalizing dolphins. The convergence of bearing angles as the ship traveled indicated the probable location of dolphins. The study areas were stratified based on sound speed profiles, sea surface temperature, and depth of the thermocline to examine the effect of oceanographic conditions on acoustic detection distances (the greatest distance at which bearing angles can be determined) for delphinid whistles. Significant differences in mean detection distances were found between offshore ETP (2.9 nmi) and USWC waters (1.5 nmi, $p < 0.001$), and between USWC waters and waters off Peru and Baja pooled (2.9 nmi, $p = 0.005$). Ray trace models confirm these empirical differences in acoustic propagation. Results suggest that the effectiveness of acoustic methods during cetacean abundance surveys varies with oceanographic conditions. Regional variability in whistle propagation may partially explain why whistles are seldom or never used by some northern delphinid species but are commonly used by virtually all tropical delphinid species.

2:55

4pAB7. Automatic species identification of odontocete calls in the Southern California Bight. Marie Roch (Dept. of Computer Sci., San Diego State Univ., 5500 Campanile Dr., San Diego, CA 92182-7720), Melissa Soldevilla, and John Hildebrand (The Univ. of California at San Diego, La Jolla, CA 92093)

In this work, a method for the automatic classification of odontocete species from their vocalizations is presented. We consider the vocalizations of three dolphin species which were recorded in a pelagic environment of the southern California Bight over a period of approximately three years. 57B naval sonobuoys sampling at 48 kHz were used to record the calls of common (*Delphinus delphis*), Risso's (*Grampus griseus*) and Pacific white-sided (*Lagenorhynchus obliquidens*) dolphins. In this study, approximately 3 h of calls were selected, with a disproportionate number of common dolphin calls appearing. An emphasis has been made on clas-

sifying calls across recording sessions, thus illustrating the system's ability to perform classification in different acoustic environments with a high likelihood of different individuals performing the vocalizations. Cepstral feature vectors are extracted from manually endpointed call data and classified using a Gaussian mixture model system. The performance of the automatic classification compares favorably with human performance.

3:10

4pAB8. Brydes whale (*balaenoptera edeni*) sounds collected from autonomous hydrophones in the eastern tropical Pacific, 1999–2001. Sara L. Heimlich, Sharon L. Nieuwirth, David K. Mellinger, Robert Dziak, Haru Matsumoto, and Matthew Fowler (Cooperative Inst. for Marine Resources Studies, Oregon State Univ., Newport, OR 97365)

An array of seven autonomous hydrophones, deployed within 12 N 8S and 95 110W, continuously recorded the frequency band 1–110 Hz from November 1999 to November 2001. Four low frequency sound types were observed, similar to Brydes whale vocalizations previously reported [E. Oleson *et al.*, Mar. Mammal. Sci. **19**, 407–419 (2003)]. Two averaged 2.5-s duration and were commonly high amplitude sounds that exceeded the sampling frequency. Both had a distinct tone averaging 18.6 and 25.4 Hz, respectively, several frequency-modulated higher tones that resembled harmonics but were not, and often an initial short broad band pulse which overlapped the tonal components. Two were lower amplitude sounds that typically began with a single constant-frequency upper tone, averaging 36.6 Hz, 1.1-s duration, and 28.7 Hz, 1.1-s duration, respectively. Sometimes, an overlapping lower tone followed 0.7 s later, lasting 1.6 s and averaging 21.4 and 15.3 Hz, respectively. Occasionally the upper and lower tones alternated two or more times, with decreasing amplitude on each repetition. Differences in seasonal occurrence and geographic distribution amongst the four types were found. These results support the use of passive acoustics as a technique for examining existing and proposed stock definitions.

3:25–3:40 Break

3:40

4pAB9. Acoustic observations of longline depredation by sperm whales. Aaron M. Thode (Marine Physical Lab., Scripps Inst. Oceanogr., UCSD, La Jolla, CA 92093-0238), Janice M. Straley (Univ. of Alaska Southeast Sitka Campus, Sitka, AK 99835), Tory M. O'Connell (Alaska Dept. of Fish and Game, Sitka, AK 99835), and William C. Burgess (Greeneridge Sci., Inc., Goleta, CA 93117)

Sperm whales have learned to take sablefish off longline gear in the Gulf of Alaska. Over the past two years the Southeast Alaska Sperm Whale Avoidance Project (SEASWAP) has collected biopsy and photo-ID data concerning longline depredation, via collaboration with local fishermen in Sitka, AK. In 2004, in collaboration with SEASWAP, sets of autonomous acoustic recorders were attached to the anchor lines of several longline deployments, effectively converting the fishing gear into a vertical acoustic array deployed 200 m beneath the surface. The fishing vessels then left the area, leaving the acoustic instruments behind to monitor the region. Several hours later, typically the next morning, each vessel would return and begin hauling in the gear, a procedure that was also recorded by the instruments. In May 2004 an interaction of two sperm whales with the F/V Cobra was recorded. It was found that the animals started producing sounds within 20 min of the longline recovery and continued being extremely vocally active throughout the recovery. Acoustic multipath was exploited to estimate the range and depth of the whales. [Work sponsored by the North Pacific Research Board.]

4pAB10. Source levels of free-ranging killer whale (*orcinus orca*) social vocalizations. Val Veirs (Phys. Dept., Colorado College, Colorado Springs, CO 80903)

An array of four hydrophones spaced over 200 m near the shore of San Juan Island, WA, has been used to localize the underwater vocalizations of Southern Resident orca whales. The hydrophones (ITC) have home-built pre-amps and amplifiers and signals are digitized simultaneously by two computers. The system has been calibrated with an underwater speaker (J-9). Automatic call detection is used to activate data logging on one computer and real-time localization on a second that directs a video camera to record the water's surface at the predicted source location. The system requires about 1 s to detect a call, locate it and redirect the video camera. This is the first report of source levels of social vocalizations of free-ranging Southern Resident orcas. To date (July 2004), 160 orca calls have been localized within 200 m of the center of the array. The average acoustic intensity of these calls is 145 dB *re* 1 μ Pa@1 m. The range of intensities is from 137 to 157 dB *re* 1 μ Pa@1 m. Complete results including intensity angular distributions with respect to the probable direction of whale travel will be presented. [Work supported by 32 undergraduate researchers over 5 years and the Colorado College Physics Department.]

4:10

4pAB11. Calculation of repetition rates of the vocalizations of killer whales. Judith C. Brown (Phys. Dept., Wellesley College, Wellesley, MA 02481 and Media Lab., MIT, Cambridge, MA 02139), Andrea Hodgins-Davis (Wellesley College, Wellesley, MA 02481), and Patrick J. O. Miller (Univ. of St. Andrews, St. Andrews, Fife KY16 9QQ, UK)

A large number of whale sounds recorded from the Captive Killer Whale Population at Marineland of Antibes, France have recently been classified into call types using acoustic input and visual examination of spectral patterns [A. Hodgins-Davis, "An Analysis of the Vocal Repertoire of the Captive Killer Whale Population at Marineland of Antibes, France," thesis, Wellesley College, 2004]. The repetition rate of the pulsed component of an example of each call type has been calculated using algorithms which have proven successful for pitch tracking of musical signals [J. C. Brown and B. Zhang, J. Acoust. Soc. Am. **89**, 2346–2354 (1991), J. C. Brown, J. Acoust. Soc. Am. **92**, 1394–1402 (1992)]. A new method based on pattern recognition of the Fourier transform has been the most successful of these methods as it best circumvents the problem of background pump noise. Results will be presented and compared. [Sound recording supported by grants from WHOI's Ocean Life Institute and the Royal Society.]

4:25

4pAB12. Consistency in received echo characteristics with an echolocating *Tursiops truncatus*. Stuart Ibsen, Whitlow Au, Paul Nachtigal, and Marlee Breese (46-007 Lilipuna Rd., Kaneohe, HI 96744, stui1@aol.com)

In this experiment the dolphin was asked to perform a discrimination task between two simulated targets. The echoes from these targets were created using a phantom echo generator which replicated the actual echo characteristics of solid spherical stainless-steel or brass targets. The system provided full knowledge of each click produced by the dolphin and the corresponding echo received by the dolphin. During the discrimination task the dolphin used a range of clicks that differed widely in peak fre-

quency and bandwidth. However, the echoes that were returned to the dolphin showed significantly less variability in these characteristics. The dolphin in this study carefully maintained consistency in just a portion of the full bandwidth of its clicks, the region that was both within the dolphin's hearing range and had the best reflectivity off the target. The rest of the click bandwidth outside this region showed a higher degree of variability. The dolphin most likely demonstrated less control over these frequency regions because they were outside its hearing range or did not provide the dolphin with nearly as much information.

4:40

4pAB13. The spatial context of free-ranging Hawaiian spinner dolphins producing acoustic signals. Marc O. Lammers, Michiel Schotten, and Whitlow W. L. Au (Hawaii Inst. of Marine Biol., P.O. Box 1106, Kailua, HI 96734)

An important requirement for understanding the function of animal signals is the ability to determine the context in which they occur. To improve our understanding of how dolphins use acoustic signals in the wild a three-hydrophone towed array was used to investigate the spatial distribution of Hawaiian spinner dolphins (*Stenella longirostris*) relative to each other as they produced whistles, burst pulses and echolocation clicks. Groups of 10 to 60+ animals were recorded while they traveled and socialized in nearshore waters off Oahu, HI. Broadband recordings up to 62.5 kHz were obtained using a PC-based multi-channel system with simultaneous analog-to-digital converters. Signaling animals were localized using time of arrival difference cues on the three channels. Localized whistles produced closely in time typically originated from individuals spaced widely apart (10–100+ m), supporting the hypothesis that whistles play a role in maintaining contact between animals in a dispersed group. Multiple burst pulses, on the other hand, usually came from individuals spaced closer to one another (5–20 m), suggesting they function as a more intimate form of signaling between adjacent individuals. The spacing between echolocating animals was variable.

4:55

4pAB14. Will acoustically enhanced nets reduce odontocete gillnet bycatch because they are better sound reflectors? T. Aran Mooney, Whitlow W. L. Au, and Paul E. Nachtigal (Marine Mammal Res. Program, Hawaii Inst. of Marine Biol., 46-007 Lilipuna Rd., Kaneohe, HI 96744, mooneyt@hawaii.edu)

Tens of thousands of odontocetes are caught annually as bycatch in gillnet fisheries worldwide. Of particular interest are bottlenose dolphins and harbor porpoises, both caught in U.S. waters. One method to potentially reduce this incidental-take is to use nets that are easier for echolocating marine mammals to detect. These acoustically enhanced gillnets are designed to reflect odontocete sonar signals better than regular nylon nets. They have shown to reduce bycatch. Whether or not this is due to increased acoustic reflectivity remains to be determined. We measured the target strength of cod and monkfish gillnets, including those enhanced with barium sulphate, an iron oxide, using dolphin- and porpoise-like echolocation signals. From the target strengths we then predicted the detection distances of the nets by the respective animals. It was determined that at reasonable distances bottlenose dolphins should be able to detect all nets. But because of lower probable source levels, harbor porpoises may not be able to detect any gillnet before it becomes entangled. Stiffness of the nets was also quantified and experimental nets proved to be greater in stiffness than regular nets, thus likely a significant factor in reducing odontocete bycatch.

Session 4pBB**Biomedical Ultrasound/Bioresponse to Vibration and Physical Acoustics: Light and Sound Imaging in the Body**

Ronald A. Roy, Cochair

Department of Aerospace and Mechanical Engineering, Boston University, 110 Cummings Street, Boston, Massachusetts 02215

Todd W. Murray, Cochair

*Department of Aerospace and Mechanical Engineering, Boston University, 110 Cummings Street, Boston, Massachusetts 02215***Invited Papers****2:00****4pBB1. Ultrasound-aided high-resolution biophotonic imaging.** Lihong V. Wang (Dept. of Biomed. Eng., Texas A&M Univ., College Station, TX 77843-3120, lwang@tamu.edu)

Using non-ionizing electromagnetic and ultrasonic waves, we develop novel biophotonic imaging for early-cancer detection, a grand challenge in cancer research. Unlike ionizing x-ray, non-ionizing electromagnetic waves such as optical waves are safe for biomedical applications and reveal new contrast mechanisms and functional information. For example, our spectroscopic oblique-incidence reflectometry can detect skin cancers based on functional hemoglobin parameters and cell nuclear size accurately. Unfortunately, electromagnetic waves in the non-ionizing spectral region do not penetrate biological tissue in straight paths as x rays do. Consequently, high-resolution tomography based on non-ionizing electromagnetic waves alone is limited to superficial tissue imaging as demonstrated by confocal, two-photon, and optical-coherence microscopic technologies. Ultrasonic imaging, on the contrary, furnishes good imaging resolution but has poor contrast in early-stage tumors and in addition has strong speckle artifacts. We developed ultrasound-mediated imaging modalities by combining electromagnetic and ultrasonic waves synergistically. The hybrid modalities yield speckle-free electromagnetic-contrast at ultrasonic resolution in relatively large biological tissue, where the spatial resolution is scalable with the ultrasonic frequency. In ultrasound-modulated optical tomography, a focused ultrasonic wave encodes diffuse laser light in scattering biological tissue. In photo-acoustic (thermo-acoustic) tomography, a low-energy laser (rf) pulse induces ultrasonic waves in biological tissue due to thermoelastic expansion.

2:25**4pBB2. Acousto-photonics imaging: Modulation of light by ultrasound for medical imaging.** Charles DiMarzio (Northeastern Univ., 440 Dana, 360 Huntington Ave., Boston, MA 02115)

Dual-wave technologies hold the promise of new medical imaging techniques. One such combination is acousto-photonics imaging, in which light waves are "tagged" by ultrasound waves. The promise is the spectroscopic capability of optical imaging, to measure hemoglobin, with the resolution of ultrasound. A further promise is the ability to measure both optical and acoustic properties in a single measurement. The mechanisms of interaction include modulation of the optical index of refraction and modulation of the distance between optical scatterers, both resulting in phase shifts of the detected light. Because these phase shifts are random, they are difficult to detect. Detectors average over many speckles and thus do not enhance the signals. The use of a photo-refractive crystal permits large detectors to be used, and also provides resolution in the axial direction, leading to the possibility of using the ultrasound to introduce virtual sources of diffusive waves, which can be treated with processing techniques employed in diffusive optical tomography. This talk will provide an overview of the field, including various source and detector configurations, measurement techniques, and suggestions for practical implementation.

2:50**4pBB3. Dynamic holographic imaging of acoustic motion.** Ken L. Telschow and Vance A. Deason (Idaho Natl. Eng. and Environ. Lab., Idaho Falls, ID 83415-2209, telsch@inel.gov)

Optical measurement of acoustic motion at a surface is usually accomplished with a scanning point probe. A full-field view acoustic imaging microscope is described that records subnanometer displacement amplitude and phase over a surface without scanning. Images are recorded at video frame rates and heterodyne principles are used to allow operation at any frequency from Hz to GHz. Fourier transformation of the acoustic displacement image provides a mapping of excited mode wavenumbers at any frequency. This method readily measures (a) the acoustic displacement of transducers, (b) acoustic bulk wave reflection from subsurface objects in solids and liquids, (c) resonant ultrasound vibration modes of complex shapes, (d) surface acoustic waves, and (e) plate wave modes in isotropic and anisotropic materials. These and other examples are presented along with discussion and analysis for characterizing material properties, such as physical dimensions, material elastic properties, crystal orientation, electromechanical coupling, and the identification and visualization of defects.

4pBB4. Schlieren imaging of sound at the turn of the century. Claudio I. Zanelli and Sam M. Howard (Onda Corp., 592 Weddell Dr., Ste 7, Sunnyvale, CA 94089, cz@ondacorp.com)

The mystery of sound fields lies largely in that, despite the complex presence of diffraction effects, at a human scale, they cannot be seen. We sample an acoustic field two points at a time, and mapping it completely, even in two dimensions, requires a constant field and a lot of patience. Vision allows capturing two-dimensional images, and even mental three-dimensional images. Yet light can only be seen when it is interfered with, and scattered into, our retinas. Like sound, we must destroy its field in order to sense it. Quite possibly, the lack of sensory perception of acoustic field maps is responsible for our slowness in understanding the nature of waves. The limitations of point-to-point mapping of fields leads many to take a few samples and then work directly with mathematical models. Schlieren is an ancient yet rediscovered, fascinating technique to generate instantaneous maps of acoustic fields based on the interaction of light and sound. Its value as a guide to intuitively connect with the underlying physics is undeniable, and new developments have turned it into a quantitative method, even where no other mapping methods are possible.

4pBB5. A novel approach to high-frequency laser-based acoustic microscopy. Todd W. Murray and Oluwaseyi Balogun (Dept. of Aerosp. and Mech. Eng., Boston Univ., 110 Cummington St., Boston, MA 02215)

A laser-based acoustic microscopy system has been developed which uses a modulated continuous wave (cw) laser source for acoustic wave generation. An amplified electro-absorption modulated diode laser source is used for narrow-band excitation of acoustic waves generation at frequencies up to 200 MHz. The resulting surface displacement is measured using a stabilized Michelson interferometer fed into an rf lock-in amplifier. The detection bandwidth reduction afforded by this technique allows for a significant improvement in signal-to-noise ratio (SNR) over systems using pulsed laser excitation, and displacement sensitivity in the femtometer range is obtained with only minimal surface heating. The source modulation frequency is scanned over the bandwidth of interest and the transient response of the specimen reconstructed from the frequency domain data. In addition, the use of cw excitation allows for resonant excitation of micro- and nanoscale structures for materials characterization or sensor applications. Experimental results demonstrate that the system is well suited for thin-film characterization, and for the nondestructive analysis of nanometer scale electromechanical devices.

Contributed Papers

4pBB6. Further investigation of acousto-optical imaging in highly diffusive media using a photorefractive crystal based detection system: Sensitivity and contrast. Lei Sui, Emmanuel Bossy, Ronald A. Roy, and Todd W. Murray (Dept. of Aerosp. and Mech. Eng., Boston Univ., Boston, MA 02215, suilei@bu.edu)

Acousto-optical imaging takes advantage of good optical contrast and high ultrasonic resolution to facilitate imaging and characterization of biological tissues. By detecting ultrasound modulated optical signals, the local opto-mechanical properties can be extracted. Among the state-of-the-art detection techniques, there is a trade-off between the axial resolution (or ultrasound bandwidth) and signal-to-noise ratio. Recently, a photorefractive crystal (PRC) based system was shown to enhance the detection of ultrasound-modulated optical signals [Sui *et al.*, Proc. SPIE **5320**, 164–171 (2004)]. This system achieved improved axial resolution by using short-duration acoustic pulses [Sui *et al.*, J. Acoust. Soc. Am. **115**, 2523 (2004)]. Previous experiments were done in tissue-mimicking phantoms whose optical diffusivities were less than that of most biological tissues. In this paper, the PRC-based detection system is applied to acousto-optical imaging in highly diffuse media and in biological tissues *in vitro*. Both transmission and right-angle detection geometries are studied. The relationship between the amplitude of the detected signals and acoustic driving pressure is determined. The ability of this system to detect both optical absorption contrast and optical scattering contrast is explored. [Work supported by the Center for Subsurface Sensing and Imaging Systems via NSF ERC Award No. EEC-9986821.]

4pBB7. Combining acousto-optical imaging with diagnostic ultrasound: resolution, contrast and speed. Emmanuel Bossy, Lei Sui, Todd W. Murray, and Ronald A. Roy (Dept. of Aerosp. and Mech. Eng., Boston Univ., Boston, MA 02215)

Ultrasound modulation of light can be used to image optical properties in highly diffusive media such as biological tissues. In a previous study [Bossy *et al.*, J. Acoust. Soc. Am. **115**, 2523 (2004)], a commercial ultrasound scanner was combined with a photorefractive crystal (PRC)-based optical detection scheme to generate simultaneous images of acoustic and optical properties of inclusions embedded in diffusive tissue phantoms. In this paper, we quantitatively investigate the performance of the system in terms of resolution, contrast, and acquisition time. Measurements are performed on gel-based highly diffusive tissue phantoms, with embedded targets possessing optical contrast relative to the surrounding medium. The resolution of the acousto-optical images turns out to be given by the ultrasound beamwidth for the lateral resolution, and by the pulse duration for the axial resolution. Phantoms and targets with different optical contrasts are imaged to assess the feasibility using this technique to delineate tissue types *in vivo*. The time needed to acquire acousto-optical images depends on the desired quality of the image (signal-to-noise ratio) and ranges from 1 to 30 s per line. [Work supported by the Center for Subsurface Sensing and Imaging Systems via NSF ERC Award Number EEC-9986821.]

Session 4pEA

Engineering Acoustics: Microphones and Loudspeakers

Gilles A. Daigle, Chair

Institute of Microstructural Science, National Research Council, Ottawa, Ontario K1A 0R6, Canada

Chair's Introduction—1:25

Contributed Papers

1:30

4pEA1. Precision calibration of measurement microphones. Gunnar Rasmussen (G.R.A.S. Sound & Vib. A/S, Staktoften 22D, DK-2950 Vedbaek, Denmark, gr@gras.dk)

The calibration of measurement microphones is often carried out as a rough check using a comparison between the measurement microphone and another microphone. A calibration with lower uncertainty involves either a complete reciprocity calibration which is hard to carry out in the field or the use of a pistonphone, where the calibration is related to well known physical quantities easy to measure. The largest contribution to the uncertainty is often the barometric pressure and the use of less accurate barometers. The use of a precision barometer built into the pistonphone greatly reduces the uncertainty due to human error and due to varying atmospheric conditions independent of altitude. Normally the ambient temperature at which the calibration takes place is also reported. Practical experience shows that it may be more relevant to report the actual temperature of the transducer under calibration. The final working temperature for a microphone is reached several minutes after the start of the calibration. It is therefore practical to measure and report the temperature of the actual calibration coupler and microphone combinations in order to minimize the uncertainty. Direct data transfer of all relevant data calibration level, barometric pressure, and temperature to a data system to ensure immunity to human error is also practical.

1:45

4pEA2. Design and performance of a micro-probe microphone. Gilles A. Daigle and Michael R. Stinson (Natl. Res. Council, Ottawa K1A 0R6, Canada)

It is often necessary to measure sound fields in confined spaces where minimum disturbance of the sound field is important. The B&K 4170 is a horn-coupled probe microphone for which a 1.25-mm-i.d. probe is attached to the narrow end of the horn and a 1/2-inch microphone is connected to the wide end. In applications where the confinement of the space is extreme, this probe is too large. The probe section of the B&K 4170 has been redesigned using smaller diameter probes. Simulations were made for probes having outside diameters ranging from 1.25 mm down to 0.2 mm. It was found that acceptable sensitivities could be expected from a probe with 0.2 mm o.d. (0.1 mm i.d.) if the length is less than 1 cm. The micro-probe is connected to a larger cylindrical tube with the correct o.d. for mounting on the B&K 4170 casing. Viscous resistance due to the small diameter provides an almost flat response. Several micro-probes were constructed and their performance measured and compared to the simulated results.

2:00

4pEA3. Harsh environment windscreen analysis and design. Jeffrey Hill, Jonathon Blotter, and Timothy Leishman (Brigham Young Univ., 435Q CTB Provo, UT 84602, jblotter@byu.edu)

This paper presents the analysis and design for a multimicrophone probe placed outdoors in relatively harsh environments including high temperature, high humidity, rain, wind, and soil particles. The windscreen was designed to minimize the insertion loss as well as maintain the phase relationship in a circular 5-probe microphone array. A numerical CFD model, which accounts for wind, soil particles, the windscreen material, and transient pressure fluctuations, will be presented. The parametric design using this numerical model will be presented and discussed, along with experimental results. The experimental testing consists of measuring the wind attenuation, the insertion loss, as well as any errors in the phase relationship. Testing is done in an anechoic chamber using two methods: a stationary microphone array, and attaching the microphone system to a rotating arm to simulate the wind. Measurement results which correlate the density and thickness of the foam with the insertion loss and the drop in wind noise level caused by the foam will be presented. Also, the insertion loss due to different weather-proof coverings is reported.

2:15

4pEA4. Reduction of airflow noise in telephone handsets. Michael R. Stinson, Gilles A. Daigle, and John F. Quaroni (Inst. for Microstructural Sci., Natl. Res. Council, Ottawa, ON K1A 0R6, Canada, mike.stinson@nrc-cnrc.gc.ca)

The bursts of airflow that accompany plosive sounds such as a "p" or "t" can cause significant noise in a telephone handset when the talker's mouth is close to the mouthpiece of the handset. Measurements of the generated noise have been made on a series of modified handsets, using various arrangements of ports and tubes between microphone and the mouthpiece exterior. The use of one, two and more exterior holes, and the introduction of different geometrical coupling between these holes and the interior microphone were explored. The noise could be reduced by over 20 dB by using an array of small (less than 0.5 mm diameter) exterior holes coupled with an offset to a sound tube leading to the microphone. [Work supported in part by Mitel Networks.]

2:30

4pEA5. Acoustical nonlinearities in compression chambers and horns of horn drivers. Alexander Voishvillo (JBL Professional, 8400 Balboa Blvd., Northridge, CA 91329)

A horn driver is an essential part of sound reinforcement systems. Principles of horn driver operation are associated with generation of acoustical nonlinear distortion. Distortion is generated in a compression chamber by nonlinear compression, and by modulation of air stiffness, mass, and viscous losses, and a nonlinear relationship between particle

velocity and sound pressure. Distortion is also generated in the phasing plug and horn by propagation of high amplitude waves. The effect of compression chamber air mass and viscous losses modulation and nonlinearity has not been researched previously. The analysis of modulation and nonlinear effects in the compression chamber is based on analysis of variation of the compression chamber's air moving mass, viscous losses, and compliance as a function of the diaphragm's instantaneous position and level of acoustical pressure. The pressure and diaphragm position-dependent parameters are incorporated into the system of nonlinear differential equations governing movement of the diaphragm and oscillation of air in compression chamber. Analysis of nonlinear propagation effects in the phasing plug and horn is based on numerical solution of an implicit nonlinear equation for propagation of finite amplitude sound waves. The work results in the methodology of simulation dynamic performance of horn drivers and provides comparative analysis of different acoustical nonlinear and parametric mechanisms.

2:45

4pEA6. Additional modeling details for the suspension of a moving-coil loudspeaker. Timothy Leishman and Gordon Dix (Acoust. Res. Group, Dept. of Phys. and Astron., Brigham Young Univ., Provo, UT 84602, tim_leishman@byu.edu)

The suspension of a typical moving-coil loudspeaker incorporates both a surround and a spider of dissimilar construction. Nevertheless, classical lumped-element models simplify this composite system by representing it with a single mechanical compliance and a single mechanical resistance. This paper presents a more detailed lumped-element model of the suspension to account for individual contributions of the surround and spider, their effective masses, and their primary resonances. It subsequently proposes an expanded equivalent circuit for a loudspeaker and discusses experimental methods to characterize new circuit elements.

THURSDAY AFTERNOON, 18 NOVEMBER 2004

PACIFIC SALON 3, 1:00 TO 4:30 P.M.

Session 4pMUa

Musical Acoustics: Musical Instruments of the Asia-Pacific Region

James P. Cottingham, Chair

Department of Physics, Coe College, Cedar Rapids, Iowa 52402

Chair's Introduction—1:00

Invited Papers

1:05

4pMUa1. The acoustics of the khaen, bawu, and gourd pipe. James P. Cottingham (Phys. Dept., Coe College, Cedar Rapids, IA 52402, jcotting@coe.edu)

Mouth-blown instruments using a free reed coupled to a pipe resonator have a long history in China, Japan, and throughout Southeast Asia. The khaen and the gourd pipe are unusual wind instruments, employing a free reed mounted in the side of a cylindrical tube open at both ends, although in the gourd pipe the player has the option of closing the bottom of the pipe so that the reed is near the closed end. The bawu is a closed cylindrical pipe with a free reed at one end, in which the effective acoustical length is varied by the use of tone holes. The playing frequency of each pipe of the khaen or the gourd pipe is typically slightly above both the resonant frequency of the pipe and the natural frequency of the reed. In the bawu, on the other hand, the pipe resonance pulls the sounding frequency well above the natural reed frequency, resulting in a striking difference in tone quality. Acoustical measurements made on these instruments include studies of reed vibration and impedance measurements of the pipes, with particular attention to the coupling of the reed vibration with the pipe resonator.

1:25

4pMUa2. Expressive modeling of Chinese folk wind instruments. Lydia Ayers and Andrew Horner (Dept. of Computer Sci., Hong Kong Univ. of Sci. and Technol., Clear Water Bay, Kowloon, Hong Kong, layers@cs.ust.hk, horner@cs.ust.hk)

Chinese folk wind instruments, including the *paixiao*, *xun*, *koudi*, *bawu*, *hulusi*, and *lusheng*, have distinctive timbres but are less well known than the other Chinese wind instruments. The *paixiao* (a panpipe) and the *xun* (ocarina) are ancient instruments. The *koudi* is a small bamboo whistle/flute, which can play exciting birdlike melodies. The *bawu*, *hulusi*, and *lusheng* have different shapes and playing techniques but have similar embedded metal plate reeds that produce their sounds. This talk will give brief descriptions of the musical and spectral characteristics of these instruments, and then describe the additive synthesis design template used to capture their subtle timbral characteristics using score data in Csound. In addition, the design is expressive and successfully uses spectral interpolation for the idiomatic ornamental phrasing for each instrument. We use a special amplitude/frequency function modulation method to simulate trills, which are also tied to varying numbers of grace notes, as tested on traditional Chinese scores. [Work supported by RGC Grant No. HKUST 6020/02H, with special thanks to Chan Chun Nin for recording instrument samples and consultation on playing the instruments, and to Alfred Wong and Jenny Lim for preparing examples.]

4p THU. PM

4pMUa3. The timbre space of the Chinese membrane flute (dizi): Physical basis and psychoacoustical effects. Chen-Gia Tsai (Grad. School of Folk Culture and Arts, Taipei Natl. Univ. of Arts, 1 Hsuen-Yuan Rd., Peitou, Taipei 112, Taiwan, tsai_chen_gia@yahoo.com.tw)

The Chinese membrane flute (dizi) has a membrane covering a hole in the wall of the instrument between the mouth hole and the uppermost finger-hole. The membrane is driven by the acoustic pressure in the pipe, producing rich harmonics through its nonlinearity. The dizi timbre is multidimensional and varies with pitch. Inhomogeneity of the dizi timbre is well known by musicians. It reflects a general aesthetic tendency in traditional Chinese music that a skillful musician should produce a variety of timbre with a solo instrument. The major spectral features of dizi tones, i.e., (1) formants, (2) the predominance of odd harmonics, and (3) harmonic-to-subharmonics ratio at high frequencies, can be explained by a Duffing oscillator model of the membrane. They constitute the three dimensions of the dizi timbre space, where the “trajectory” of the changing timbre of a dizi melody is embedded. Psychoacoustical effects associated with them are discussed, including (1) spatial effects due to abrupt brightness changes, (2) multi-pitch effect due to the predominance of upper odd harmonics, and (3) roughness induced by subharmonics.

4pMUa4. Investigation on tonal and playability qualities of didgeridoos: Correlations with physical properties. Rene E. Causse (Ircam CNRS UMR 9912, 1 place Igor Stravinsky, 75004 Paris, France)

The quality of an instrument, perceived by a player, is usually separated into two fields, tonal and given possibilities by the instrument in terms of playability. For the didgeridoo, tonal quality is first linked to the spectral content of the emitted notes, a result of the oscillation of a complex system composed of the instrument, the lips and the player's vocal tract. It is also related to the character of spatialization of the sounds which one sometimes associates with the semantic attribute of sound width or sound envelopment. The playability is attached to the playing facility but also to the various aptitudes of the instrument. Among those one finds the aptitude for the modulations, for the rhythmic effects, for the overblowing, for vocalization, the cries, etc. The air column resistance, the sound efficiency, and the sound dynamics are also linked to the playability. The study is carried out on a series of eight instruments chosen in a corpus of around 50 instruments. The study includes tests with instrumentalists, acoustical and geometrical measurements and analysis of the emitted sounds. Some correlations between the physical properties of the same instrument, the nature of the emitted spectrum and the quality will be presented.

4pMUa5. Conch-shell and bamboo flute: Spiritual and musical expressions of acoustics in Vedic Hinduism. M. G. Prasad (Mech. Eng. Dept., Stevens Inst. of Technol., Hoboken, NJ 07030, mprasad@stevens.edu)

In Vedic Hinduism rooted in India, the field of acoustics plays an important role. The multi-faceted expressions of acoustics include Vedic mantras (chants), music, yoga, philosophical aspects of language, spiritual aspects of sound, etc. A large number of musical instruments such as woodwind, percussion, string, etc. are used in Hindu culture and religion. Conch-shell and bamboo flute are among the earliest musical instruments. A conch-shell is used to evoke the spiritual effects in the environment during worship rituals at temples and homes. It has a high quality factor. The music played through a transverse bamboo flute is very melodious. It is a major instrument in orchestra music and dance performance. Concerts with bamboo flute as the lead instrument are also very common. This paper deals with spectral characteristics and tonal effects of these two important instruments. Also, the presentation includes a discussion of the importance of acoustics in life as viewed in Vedic Hinduism.

4pMUa6. Model experiments on the “sawari” mechanism: Vibrational behavior of a string with a moving boundary. Akiko Fujise and Shigeru Yoshikawa (Dept. of Acoust. Design, Grad. School of Kyushu Univ., Fukuoka, 815-8540, Japan)

The “*sawari*” mechanism, seen in some plucked string instruments like Japanese *biwa* and *shamisen*, produces the vibration of the string with a moving boundary and results in its unique sustaining tone, whose elaborate tuning is essential in Japanese traditional music. Since a slight change of the *sawari* surface's shape strongly affects the tone quality, it is important to examine the relationship between the boundary shape and the consequent vibration in reality. In this presentation, some results of model experiments on a string with the *sawari* mechanism are reported and discussed. By using a high-speed digital video camera and a piezo-electric pickup, the overall string motion and the local interaction between the string and the boundary surface are experimentally measured under several *sawari* conditions. The results are analyzed by applying image processing to video data and also time-frequency analysis to acceleration signals, then compared between basic shapes (plain and curved), widths, and shaven depths of the constraint.

4pMUa7. Acoustics of Chinese, Japanese, and Korean bells. Thomas D. Rossing (Phys. Dept., Northern Illinois Univ., DeKalb, IL 60115, rossing@physics.niu.edu)

The earliest known Chinese bells were small clapper bells of clay and copper from the third millennium BC. Beginning with the early Bronze Age (ca. 2000 BC), however, bronze became the material of choice. We discuss early Chinese two-tone bells including *nao*, *yongzhong*, *niuzhong*, and *bo* and illustrate their vibrational behavior using holographic interferometry. We discuss the effect of various physical parameters on the spacing between the two tones. We compare temple bells from China, Japan, and Korea, including such famous bells as the the Yong-le bell in Beijing and the King Songdok bell in Korea.

4pMUa8. Vibrational modes of pyen-gyoung (Korean chime stone) and pyeon-jong (Korean chime bell). Junehee Yoo (Dept. of Phys. Education, Seoul Natl. Univ., Seoul 151-742, Republic of Korea) and Thomas D. Rossing (Northern Illinois Univ., DeKalb, IL 60115)

Pyeon-gyoung, chime stone, and Pyeon-jong, chime bell, have been allocated as a set of instruments in Korean traditional court music and Confucian ceremony. Pyeon-gyoung is a set of 16 L-shaped chime stones covering one and one-third octaves and pyeon-jong is a set of 16 almond-shaped bronze chime bells, one octave lower than pyeon-gyoung. The nominal frequencies increase with the thickness of the stones and bells. The vibrational mode frequencies in a pyeon-gyoung and pyeon-jong have been studied. The frequency ratios of the modes in stone chimes are smaller than those of marimbas. The modal shapes of the lowest and highest stones, mapped by TV holography, by scanning with an accelerometer and animated by STAR, are shown.

4pMUa9. Acoustic matching of the bars and resonators of the Indonesian gender. Paul Wheeler (Utah State Univ., 2140 Old Main Hill, Logan, UT 84322)

The gender is a 14-bar metallophone with bamboo resonators used in the Indonesian gamelan. The 14 bronze bars (with stiffening ridges on top of the bars) are carefully tuned to one of three pentatonic scales. The bars are strung together and spaced above 14 bamboo resonator tubes housed in a wooden frame. The bamboo tubes are roughly the same length and diameter, however the knot in the bamboo pipe is in different locations, creating closed pipes of different lengths. Fine tuning of the tubes is achieved by placing rocks or sand in the bottom of the tubes. This paper describes an acoustic study of the gender barung. Tuning of the bars across the octaves was within 15 cents (with one exception), showing great care in the construction and tuning of the bars. Tuning of the resonator tubes, however, was much more divergent, showing less consideration for the actual frequencies of the resonators. Tuning of the resonators to match the bars is left for the player by adding materials to the resonator tube.

Contributed Paper

4:15

4pMUa10. Additional modes in transients of a Balinese Gender dasa plate. Bader Rolf (Musikwissenschaftliches Institut, Neue Rabenstr. 13, 20354 Hamburg, Germany, r_bader@t-online.de)

The initial transient of a Balinese bronze plate of a Gender dasa (metallophone) struck by the instrument's hammer is examined in detail. The spectrum of the radiated sound as well as that of measurements by an accelerator consist of many more eigenmodes compared to the theoretical values for the bending, longitudinal, axial and torsional wave modes. It shows up that the trapezoid shape of the plate leads to these additional

frequencies in the transient spectrum, which disappear in the quasi-steady state. As to each bending mode, there exist at least two additional modes. There seems to be a scattering of the initial impulse at the geometrically unsteady boundaries of the trapezoid, which define new internal borders, where new modes can arise from. The low energy of these modes results in a fast decay, while the fundamental mode is still sustained by the bamboo resonator of the Gender. Computer models show the reasoning for such additional spectra modes in principle using transient impulse simulations. These modes can occur just in the case of a hard struck as the typical playing mode of Bali music, which is known to be more lively than Japanese music and corresponds to the initial transient found here.

THURSDAY AFTERNOON, 18 NOVEMBER 2004

SAN DIEGO ROOM, 5:00 TO 6:00 P.M.

Session 4pMUb

Musical Acoustics: Concert by Christopher Adler: New and Traditional Music for Khaen, the Free-Reed Mouth Organ of Laos and Northeast Thailand

James P. Cottingham, Chair

Department of Physics, Coe College, Cedar Rapids, Iowa 52402

Chair's Introduction—5:00

The *khaen* is a free-reed bamboo mouth organ prominent among peoples of Lao ethnicity in Laos and northeast Thailand, and likely an ancestor to more well-known East Asian mouth organs as well as to Western free-reed instruments. Dr. Christopher Adler has been researching and performing the traditional music of the *khaen* for a decade and composing and performing new compositions for *khaen* almost as long. This performance will include five traditional improvisations in each of the five melodic modes available on the *khaen* as well as two original compositions by Dr. Adler which make extensive use of non-traditional techniques. The performance will also include demonstrations of two single-pipe free-reed instruments, the *saneng keo* and the *pii phuthai*, and a duo performance of a classical Thai composition with Supeena Insee Adler playing the three-string Thai zither *jakay*.

Christopher Adler is a composer, improviser and performer living in San Diego, California. In addition to working with the traditional musics of Thailand and Laos, his compositions draw upon his background in mathematics and computer modeling. He received Ph.D. and Master's degrees in composition from Duke University and Bachelor's degrees in music composition and in mathematics from the Massachusetts Institute of Technology. He is currently Director of the Music Program and an Assistant Professor at the University of San Diego. His work may be heard on 9 Winds Records, Artship Recordings, and WGBH's Art of the States, and three of his compositions for *khaen* may be heard on *Epilogue for a Dark Day*, a newly released solo CD on Tzadik Records. (www.christopheradler.com).

Session 4pNS

Noise: Test Chambers, Sound Sources, Noise Reduction

Brian Anderson, Chair

Graduate Program in Acoustics, Pennsylvania State University, State College, Pennsylvania 16802

Contributed Papers

1:30

4pNS1. Development and measurement of a variable acoustic test chamber. Colin Barnhill, Robert Ng, James E. West, and Ilene J. Busch-Vishniac (Elec. and Computer Eng. and Mech. Eng. Depts., Johns Hopkins Univ., 3400 N. Charles St., Baltimore, MD 21218)

We have designed and constructed an acoustic test facility capable of varying from largely anechoic to largely reverberant. The chamber measures $18.25 \times 13.5 \times 8$ ft., so it has a total volume of 1971 ft^3 and a total surface area of 1000.75 ft^2 . The walls are lined with removable fiberglass panels covered in fabric; a hard ceiling has a removable drop ceiling of tectum panels located below it; and the linoleum floor on slab is covered by removable thick carpeting. The reverberation time in the chamber can be varied by over an order of magnitude (i.e., from 150 ms to 1.5+ s) for frequencies above 200 Hz as measured by traditional means and with a special directional microphone system. All measurement tools excluding sound sources and microphones are located outside of the room so as not to influence the sound characteristics.

1:45

4pNS2. Towards aural opacity—across a visually transparent medium. Michael Ermann, Matt Lutz, Chris Jackson, Brooke Karius (Virginia Tech School of Architecture + Design, 201 Cowgill Hall (0205), Blacksburg, VA 24061, mermann@vt.edu), David Kilper, Indhava Kunjara Na Ayudhya, Thomas Pettin, Tom Pritts, Damian Seitz, Rebecca Stuecker, and Joe McCoy (Virginia Tech School of Architecture + Design)

Virginia Tech students undertook a study and participated in a design competition to provide the campus physical plant building with a wall that allows for natural light and views while maintaining a suitable aural barrier. The wall, which replaces a large garage door, will be constructed as part of an effort to renovate a wood/metal shop into an office. It must provide for adequate transmission loss to account for the noise generated by adjacent utility trucks, shop activities and an *ad hoc* basketball court. Unable to obtain published data for glass vestibules, students measured the transmission loss across another campus building's glass entry vestibule. Currently there are plans for the winning students design entry to be constructed as part of the renovation.

2:00

4pNS3. S-shaped glass also stands for soundless. Sound insulation measurements and implication for building practice. Beatriz M. Pinto (AFassociados, Projectos de Engenharia S.A., Dept. of Acoust., Cais do Luga 224 P-4400-492 V. N. Gaia, Portugal, beatriz@beatrizpinto.com), Laurentius C. J. van Luxemburg (DHV-Dorsserblesgraaf, 5616 vb Eindhoven, Netherlands), Antonio P. O. Carvalho (U. of Porto, P-4200-465 Porto, Portugal), and Heiko J. Martin (Eindhoven U. of Technol., 5600 mb, The Netherlands)

This paper brings to discussion the sound insulation of a special type of glazing by studying a S(inusoidal)-shaped curved glass. This is been used in a concert hall (under construction in Portugal) with a forward-thinking Rem Koolhaas's architecture requiring nonstandard solutions: glazing with very large dimensions (22 m by 15 m) and each "S-glass panel" with a wavelength of about 1.0 m and amplitude of about 0.35 m

will be applied as a double facade. The restrictions in ISO 104-3 laboratory measurements (standard test opening for glazing 1.25 by 1.50 m and niche depth of 0.45 m) were not appropriate and not completely fulfilled due to differences in dimensions between small test sample, and the real required use on site as well as the silicone joints that connect the glass elements. For these reasons, the standard test opening of 10 m^2 was used. These aspects are described and analyzed. The tests results (showing satisfactory sound insulation) were essential to understand the effect of the shape and the efficiency of the glass. This study aspires to be helpful to building designers who can, in this way, lose dependency on production process constraints and bring into being all kinds of shapes.

2:15

4pNS4. Visualizing automobile squeals and their vibration modes. Manmohan Moondra and Sean Wu (Dept. of Mech. Eng., Wayne State Univ., Detroit, MI 48202)

Automobile brake squeals have always been a major problem for the auto industry because of the high pitch and intensity level. Currently, the vehicle brake squeal diagnostics is carried out by using a scanning laser synchronized with squeals. If squeals occur at multiple frequencies, then filtering must be used so that the scanned out-of-plane vibration corresponds to a single frequency. This process is very time consuming, especially when squeals contain multiple frequencies. Moreover, the results thus obtained are valid at measurement locations only. In this paper, we use Helmholtz equation least-squares (HELS)-based near-field acoustical holography to reconstruct squeal sounds via a custom-designed 48-microphone array conformal to the shape of a brake assembly. This microphone array allows for taking a snapshot of 3D acoustic image on the surface of the brake assembly and produces 3D mappings of all acoustic quantities. In particular, we can identify the out-of-plane vibration modes that are responsible for squeals and reveal the locations where squeals originate from. There is no need for synchronization and filtering. The whole measurement can be done in seconds, and the reconstructed results are valid not only on source surface, but also in the field. [Work supported by NSF.]

2:30

4pNS5. Sound, as we see it—A better way to diagnose noise sources. Ravinder Beniwal (221 Lewiston Rd., Grosse Pointe Farms, MI 48236) and Sean Wu (Wayne State Univ., Detroit, MI 48202)

In the noise control community, it is often required to diagnose noise sources and their transmission paths. Usually, noise diagnosis consists of taking sound pressure, power, and intensity measurements around suspected sources to learn noise characteristics. If these are not enough to pinpoint the sources, then source ranking and identification of contributions from individual components are carried out. Naturally, success in this type of diagnostics requires many years of practical experiences. Even so, these tests are time consuming and *ad hoc* in nature, and the results thus obtained are valid at measurement locations only. In this paper, a new approach to noise diagnostics is presented, which can significantly reduce the time and complexities involved in diagnostics. This approach is based on the Helmholtz equation least-squares method to visualize the entire sound field generated by a sound source. It allows for 3D conformal measurements of the acoustic pressure and produces an accurate mapping of

the acoustic field on 3D surfaces and surrounding medium. These results enable one to trace sound back to its source and separate airborne from structure-borne sounds. The process can be completed in minutes and the acoustic images thus obtained speak for themselves. [Work supported by NSF.]

2:45

4pNS6. New development of two-dimensional sound index for the design of brand sound in the passenger cars. Sang-Kwon Lee, Byung-Og Jo (Dept. of Mech. Eng., Inha Univ. 253, Young Dong, Nam Gu, Incheon, Korea), and Byung-Soo Kim (Hyundai Motor Co., Whasung City, Korea)

In automotive engineering, the production sound quality becomes an important design requirement. Especially, the brand sound is one of important advantage strategy in a car company. In the paper, the subjective evaluation for booming sound and rumbling sound, which are professional words used by NVH engineers in automotive technology, are adopted as the descriptive word for a car sound. In the stage of development of a car, the trained NVH engineers evaluate the booming sound and rumbling sound subjectively in car companies. The two sound indexes, which are objective evaluation for booming sound and rumbling sound, are developed in the paper. These indexes are used for the objective evaluation of sound quality for 14 passenger cars, which are samples of famous cars in the world. The results of objective evaluation are expressed in the two-dimensional sound indexes. The brand sounds of famous cars in the world are very well represented in two-dimensional indexes. In the paper, a production car is modified using these two-dimensional indexes and its sound quality is improved. In future, the two-dimensional sound indexes are very useful for the development of brand sound. [Work supported by BK21 projects and partly supported by Hyundai Motor in Korea.]

3:00

4pNS7. Validation of the cylindrical microphone array for measuring sound power levels of noise sources. Matthew A. Nobile (IBM Hudson Valley Acoust. Lab., M/S P226, Bldg. 704, Boardman Rd. Site, 2455 South Rd., Poughkeepsie, NY 12601)

Recent papers have (i) introduced the concept of the cylindrical microphone array for the measurement of sound power levels of noise sources as an alternative to the parallelepiped and hemispherical arrays [M. A. Nobile and J. A. Shaw, Proc. Inter-Noise 99], (ii) set forth a detailed procedure for using the cylindrical array that could readily be incorporated into international standards [Nobile *et al.*, Proc. Noise-Con 2000], and (iii) presented the results of measurements using that procedure on a 1.75-m-high noise source using a variety of microphone positions to draw conclusions regarding the number of microphones, equal versus unequal areas, and traversing versus fixed-point implementations [Nobile *et al.*, Proc. Inter-Noise 2002]. The cylindrical array has since been incorporated into ISO 7779 and is being proposed for inclusion in the

upcoming revision of ISO 3744. In an effort to gain experience with this new array and to help validate its use, a prototype rotating cylindrical array was constructed, and measurements were taken on a variety of noise sources. The results are compared to those taken with the well-established hemispherical microphone array, using 40 coaxial circular paths.

3:15

4pNS8. Optimal configurations for active control of axial fans. Brian B. Monson and Scott D. Sommerfeldt (Brigham Young Univ., N283 ESC, Provo, UT 84602, brianbmonson@byu.edu)

The minimum power radiated from a single monopole noise source surrounded by four symmetrically spaced control sources has been shown analytically. A similar analysis has also been completed for a monopole noise source surrounded by three symmetrically spaced control sources. At Brigham Young University, a multichannel active noise control system has been developed for a small axial cooling fan employing four symmetrically spaced miniature loudspeakers as control actuators. This arrangement of control loudspeakers has exhibited good global control of the fan's tonal noise, achieving close to the predicted ideal for some harmonics of the blade passage frequency. According to theory, a three-source control configuration should be able to achieve nearly the same power reduction as a four-source configuration. While this is so, increasing the number of control sources to greater than four gives little to no increase in predicted power attenuation. A three-source control implementation, then, appears to be a comparable solution for the fan noise control problem. The experimental results of a three-source control and fan configuration will be discussed. An assessment will be made of the three- and four-source arrangements, including also a comparison of the experimental results versus the theoretical ideal.

3:30

4pNS9. Protection against noise produced by open lines of the underground on the example of residential area Butovo in Moscow. Igor L. Shubin, Irina L. Sorokina, and Sergey I. Kryshov (NIISF 21 Lokomotivnaya line, Moscow, 127238, Russia, niisf@ipc.ru)

In Moscow (Russia) there is actively conducted construction of new lines of the underground. Nowadays we use the model of the easy underground with an open system of lining for the economy of means for construction in designs of the underground. Till now in Moscow only 7% of the length from the general extent of the underground (276 km) was open. There are plans for construction of the underground with open lining for more than 100 km. One of the most difficult problems of designing lines of the underground with easy lining is the decision on the problem of protection against noise and vibration of adjoining residential areas. In the present work there is written the accepted decisions on protection against noise and vibration of a line of the underground constructed in the micro-district of Butovo in Moscow. The accepted technical and building-acoustic decisions are resulted.

4p THU. PM

Session 4pPP

Psychological and Physiological Acoustics: General Topics in Psychological and Physiological Acoustics

Lori L. Holt, Chair

Department of Psychology, Carnegie Mellon University, 5000 Forbes Avenue, Pittsburgh, Pennsylvania 15213

Chair's Introduction—1:30

Contributed Papers

1:35

4pPP1. Auditory categorization: Cue weighting and dimension bias. Lori L. Holt (Dept. Psych., Carnegie Mellon Univ., 5000 Forbes Ave., Pittsburgh, PA 15213) and Andrew J. Lotto (Boys Town Natl. Res. Hospital, Omaha, NE 68131)

The ability to integrate and weight information across dimensions is central to perception. The present experiments investigate this issue by training participants to categorize sounds drawn from two input distributions in a two-dimensional acoustic space defined by frequency-modulated sine waves center frequency (CF) and modulation frequency (MF). These dimensions were psychophysically matched to be equally discriminable and, in the first experiment, were equally informative for accurate categorization. Nevertheless, listeners' category responses reflected a bias for use of CF. This bias was moderated when the informativeness of CF was decreased by shifting distributions to create more overlap in CF. A reversal of weighting (MF over CF) was obtained when distribution variance was increased for CF. These results demonstrate that even when equally informative and discriminable, acoustic cues are not necessarily equally weighted in categorization; listeners exhibit biases when integrating multiple acoustic dimensions. Moreover, drastic changes in weighting strategies can be affected by changes in input distribution parameters. This methodology provides potential insights into acquisition of speech sound categories, particularly second language categories. One implication is that ineffective cue weighting strategies for phonetic categories may be alleviated by adding variance to noninformative dimensions in training stimuli. [Work supported by NIH.]

1:50

4pPP2. Overriding the signal: Perception of default versus non-default interconsonantal vowels in Japanese. Yuki Hirose (Univ. of Electro-Commun, 1-5-1 Chofugaoka, Chofu, 182-8585, Japan) and Emmanuel Dupoux (CNRS)

Consonant clusters are illegal in Japanese and in producing foreign words with clusters, the vowel [u] is used as a default epenthetic vowel. Our previous study revealed that Japanese speakers perceive a vowel [u] between consonants in VCCV stimuli, even when the vowel originally present was removed, thereby creating illusion. Our experiment 1 further shows the illusion can override the existing acoustic information supplied by other vowels. The duration of word medial vowels in VCuCV and VCiCV nonwords were manipulated in eight stages from stimuli full to 0 (removed). The Japanese subjects identified the word medial sound by choosing from [a], [i], [u], [e], and "None." For the stimuli derived from VCuCV, majority of the Japanese subjects reported [u] even when most of the vowel information was removed. Other vowels were hardly reported. In contrast, for the stimuli made from VCiCV, [u] was still reported in 60% of the cases, [i] less than 10%, and "none" about 30% when the original [i] was largely removed but still existed. Together with the results of experiment 2, an ABX discrimination study, our data suggest that epenthesis is a perceptual phenomenon that could sometimes override existing acoustic information.

2:05

4pPP3. Models of commodulation masking release. Bruce G. Berg and Ying-Yee Kong (Dept. of Cognit. Sci., Univ. of California, Irvine, 3151 Social Sci. Plaza, Irvine, CA 92697-5100)

Three models of commodulation masking release (CMR) are investigated in the context of a task in which masking and flanking bands each consist of five equal-intensity tones spanning 20 Hz. Two are multiple-channel models [J. Hall *et al.*, *J. Acoust. Soc. Am.* **76**, 50–56 (1984); S. Buus, *J. Acoust. Soc. Am.* **78**, 1958–1965 (1985)] and one is a single-channel model [B. G. Berg, *J. Acoust. Soc. Am.* **100**, 1013–1023 (1996)]. Thresholds do not provide an adequate test of the models because all three outperform listeners, thus reducing the solution to arbitrary assumptions about internal noise. However, the models can be discerned with a molecular psychophysical technique. A small intensity perturbation is added to each component of the stimulus. Correlating the trial-by-trial, simulated responses to the perturbations of each component yields a set of spectral weights. It is shown that the three models predict highly distinct patterns of spectral weights.

2:20

4pPP4. A linear mechanical-electrical-acoustic coupled physiological model of the cochlea: Response to acoustic and electrical excitation. SriPriya Ramamoorthy and Karl Grosh (Dept. of Mech. Eng., Univ. of Michigan, 3124 GG Brown, 2350 Hayward St., Ann Arbor, MI 48109-2125, grosh@umich.edu)

A linear physiologically based finite element model is developed for analyzing the global mechanical-electrical-acoustic (active) filtering in the mammalian cochlea. The model consists of a two-duct fluid-filled rectangular geometry, the micro-mechanical structural network interacting with the fluid; the electrical circuit equivalent of cells and fluid in every cross-section is connected by longitudinal cables representing the conductivity of the cochlear fluids and includes the mechano-electrical and electro-mechanical transduction at the OHC. The acoustic pressures, structural displacements, and electrical potentials are determined numerically and compared with experiments. Reducing the amount of activity present in the model reduces the gain and lowers the frequency of peak BM velocity response compared to a fully active model, in accordance with experimental data. This model also possesses near-invariance to click induced noise at different gain levels. Using the same model parameters, the predictions of the local BM velocity response to electrical stimulation match available experimental data, providing an independent test of the model capability. Predictions of electrically evoked otoacoustic emissions are found to match experimental results as well. Roughness introduced into BM stiffness is found to result in fine structures in a fully active model and have little effect on a model with reduced activity. [Research funded by NIH–NIDCD.]

4pPP5. Active linear modeling of cochlear biomechanics using Hspice.

Soon-Suck Jarng and You-Jung Kwon (Dept. of Information Control and Instrumentation, Chosun Univ., 375 Seoseok-Dong, Dong-Ku, Gwang-Ju, South Korea)

This paper shows one and two dimensional active linear modeling of cochlear biomechanics using Hspice. The cochlear biomechanical modeling is based on the deductive approach of others but is processed using Hspice which is the most popular analog electric circuit simulator among electrical and electronic engineers. Because Hspice is so generally used for active filter design, the shortage of the present cochlear model may be supplemented with analog active filters added to the present model. This extra filtering may be arbitrary so as to be called second filters. The advantage of the Hspice modeling is that the cochlear biomechanics may be implemented into an analog IC chip. This paper explains in detail how to transform the physical cochlear biomechanics to the electrical circuit model and how to represent the circuit in Hspice code. The comparison between one and two dimensional models is figured.

2:50–3:05 Break

3:05

4pPP6. Extracting the frequencies of the pinna spectral notches from measured head-related impulse responses.

Vikas C. Raykar, Ramani Duraiswami (Perceptual Interfaces & Reality Lab., Dept. of Computer Sci. and Inst. for Adv. Comput. Studies, Univ. of Maryland, College Park, MD 20742), and B. Yegnanarayana (Indian Inst. of Technol. Madras, Chennai-600036, Tamilnadu, India)

Experimental data are available for the Head-related impulse response (HRIR) for several azimuth and elevation angles, and for several subjects. The measured HRIR depends on several factors such as reflections from body parts (torso, shoulder, and knees), head diffraction, and reflection/diffraction effects due to the external ear (pinna). Due to the combined influence of these factors on the HRIR, it is difficult to isolate features thought to be perceptually important (such as the frequencies of pinna spectral notches) using standard signal-processing techniques. Signal-processing methods to extract the frequencies of the pinna spectral notches from the HRIR are presented. The techniques are applied to extracting the frequencies of the pinna spectral notches from the publicly available HRIR databases. A brief discussion relating the notch frequencies to the physical dimensions and the shape of the pinna is given. [The support of NSF Award ITR-0086075 is gratefully acknowledged.]

3:20

4pPP7. A study of the pinna anthropometry and the spectral notch frequencies.

Vikas C. Raykar, Ramani Duraiswami (Perceptual Interfaces & Reality Lab., Dept. of Computer Sci. and Inst. for Adv. Comput. Studies, Univ. of Maryland, College Park, MD 20742), and B. Yegnanarayana (Indian Inst. of Technol. Madras, Chennai-600036, Tamilnadu, India)

The head-related transfer function (HRTF) varies significantly between individuals due to the different sizes and shapes of different anatomical parts like the pinnae, head, and torso. Applications in the creation of virtual auditory displays require individual HRTFs for perceptual fidelity. A generic HRTF would not work satisfactorily, since it has been shown that nonindividual HRTFs result in poor elevation perception. While good geometrical models exist for the effects of head, torso, and shoulders, a simple model for the pinna that connects the anthropometry to the features in the HRTF does not exist. One of the prominent cues contributed by the pinna are the sharp spectral notches, which encode the elevation of the sound source. Recently, we proposed a method to extract the frequencies of the pinna spectral notches from the HRTF. In this paper we study the relationship between the extracted notch frequency and pinna anthropom-

etry. Based on these observations new approaches for HRTF customization can be developed. [The support of NSF Award ITR-0086075 is gratefully acknowledged.]

3:35

4pPP8. Comparison of auditory localization performance with various loudspeaker rendering systems.

Paul D. Henderson, Ioana N. Pieleanu, Evelyn Way, and Rendell R. Torres (Program in Architectural Acoust., Rensselaer Polytechnic Inst., 110 8th St., Troy, NY 12180, hendep2@rpi.edu)

Sound-field rendering systems based on multiple loudspeakers provide significant benefits as auditory displays for many applications in virtual audio, multimedia, teleconferencing, and auralization. In contrast to headphone-based binaural reproduction, the performance of loudspeaker rendering systems is independent of the HRTFs of the listener, and the listener is physically decoupled from the rendering system. In addition, many rendering techniques such as wave-field synthesis (WFS) are capable of simultaneously presenting an accurate perceptual experience to multiple listeners. However, the sound-field rendering technique, loudspeaker configuration, and number of sources play an important role in both the physical behavior of the rendering system and the associated localization performance of listeners. In this study, we examine the *in situ* localization performance of listeners exposed to various loudspeaker rendering systems using virtual sources. Specific reproduction methods investigated are WFS, first- and second-order Ambisonics (including a periphonic implementation), and stereophonic rendering techniques. In addition to virtual sources in the simulated free field, the effect of adding spatialized reverberation is also reviewed. The results are compared with a study of baseline auditory-system localization performance with real sources. [Work supported by the National Science Foundation and Rensselaer Polytechnic Institute.]

3:50

4pPP9. Directivity pattern simulation of the ears with two pairs of hearing aid microphone arrays by BEM.

Soon-Suck Jarng and You-Jung Kwon (Dept. of Information Control & Instrumentation, Chosun Univ., 375 Seoseok-Dong, Dong-Ku, Gwang-Ju, South Korea)

The noise reduction of the ITE (in-the-ear) hearing aid (ha) can be achieved by an array of microphones. Each of the right and the left ears is assumed to be fixed with two HA microphones. This two HA microphone array produces particular patterns of directivity by some time delay between two microphones. The directivity pattern geometrically increases the S/N ratio. The boundary element method (BEM) was used for the three dimensional simulation of the HA directivity pattern with the two pairs microphone arrays. This numerical analysis may then be applied for the calculation of the time delay parameter of the digital hearing aid DSP chip. The separation between two microphones was fixed to 10 mm. The time delay between the two microphones was changed to produce the most narrow directivity pattern in the forefront of the head. The variation of the time delay was examined in accordance with input frequencies.

4:05

4pPP10. Skull simulator for design of hearing protection systems for bone-conducted sound.

B. Scott May, Anthony J. Dietz, Odile Clavier, and Steve Bilski (Creare, Inc., P.O. Box 71, Hanover, NH 03755)

A skull simulator was developed for an investigation of bone-conducted sound transmission, which can be a significant source of sound in extreme noise environments. The simulator was built from a human skull, with silicon gel used to model internal organs and silicon and latex used to model the skin. Accelerometers attached to the skull bones were used to measure the skull vibration response. Mechanical point impedance measurements on the simulator were compared to results reported in the literature for humans, cadavers, and skulls. Reasonable agreement with these results served to validate the simulation. The link between skull

vibration measured on the skull simulator and bone-conducted sound at the cochlea of a human subject was determined by comparing the response at the mastoid measured on the skull with similar measurements on a human subject. The response of the skull simulator in a sound field was then measured. The frequency response peaked from 1 to 3 kHz, which is

where the bone-conduction limit to hearing protection attenuation is a minimum. The simulator was used to determine the effectiveness of various hearing protection components in reducing skull vibration. The results are compared to attenuation measurements of these components on human subjects.

THURSDAY AFTERNOON, 18 NOVEMBER 2004

ROYAL PALM SALON 5, 2:00 TO 3:45 P.M.

Session 4pSA

Structural Acoustics and Vibration: Measurement Methods

Scott D. Sommerfeldt, Chair

Department of Physics, Brigham Young University, Provo, Utah 84602

Contributed Papers

2:00

4pSA1. On the accuracy of modal parameters identified from exponentially windowed, noise contaminated impulse responses for a system with a large range of decay constants. Matt S. Allen and Jerry H. Ginsberg (Georgia Inst. of Technol., Atlanta, GA 30332-0405)

Multi-degree of freedom dynamic systems may contain dynamic modes with a wide range of decay rates. In such a case, collecting data over a time span long enough to capture slowly decaying modes serves to magnify the effect of noise on quickly decaying modes because their response at later times is dominated by noise. An exponential window may be applied to the data to reduce the effect of noise on quickly decaying modes or minimize leakage errors. Some practitioners may instead use a Hanning or similar window for the same purpose. This study investigates the effect of windowing on the accuracy of modal parameters identified using an experimental modal analysis technique. The data for this evaluation are noise contaminated analytical impulse response samples from which frequency response functions are obtained via FFTs. It is seen that accurate results may be obtained for the slowly decaying modes of the system even when their decay constants are many times smaller than the decay constant of the exponential window. The role of exponential windowing in reducing the effect of noise on quickly decaying modes is also demonstrated. In contrast, other windows bias the results. [Work supported by a National Science Foundation Graduate Research Fellowship.]

2:15

4pSA2. Development of a near-field acoustic holography system based on pressure and velocity measurements. Michael C. Harris, Jonathan D. Blotter (Dept. of Mech. Eng., Brigham Young Univ., Provo, UT 84602), and Scott D. Sommerfeldt (Brigham Young Univ., Provo, UT 84602)

Since its conception in 1980, NAH has become an accepted analysis tool with a number of commercial packages now available. One limitation of NAH is that the acoustic field is reconstructed through a spatial sampling. Therefore, this technique becomes less efficient at higher frequencies where microphone array spacing must be reduced. Analytical models have indicated that a pressure and velocity-based 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 at high frequencies. The pressure and velocity-based reconstruction method will be introduced. Experimental work towards validation of the analytical models will be presented.

2:30

4pSA3. Reconstructing transient acoustic field resulting from an explosion. Huancai Lu and Sean Wu (Dept. of Mech. Eng., Wayne State Univ., Detroit, MI 48202)

Generalized Helmholtz equation least squares (HELs) formulations for reconstructing transient acoustic radiation from an arbitrary object subject to arbitrarily time-dependent excitations are developed. Laplace transform is employed to build a transfer matrix for a multi-input multi-output system and reconstructed signals are obtained using convolution with respect to the input data in the time domain directly. Stability of solutions is realized by controlling the pole-residue locations in the Laplace domain. To facilitate calculations of the poles and residues, ztransform is utilized. To validate this general transient HELs formulation, the reconstructed pressure histories are compared with those measured by pressure sensors at the same locations. Space-time tomography resulting from different charges is produced, and the acoustic pressure distribution over the entire time record is reconstructed. It is interesting to notice that this transient HELs formulation enables one to trace time histories of blast pressures back to the explosion location. In other words, it is possible to reconstruct an explosion back in space and time based on the time-domain signals measured at a later time. [Work supported by NSF.]

2:45

4pSA4. Method of superposition applied to patch holography. Angie Sarkissian (Naval Res. Lab., Washington, DC 20375)

The method of superposition is applied to reconstruct the field on a patch on the surface of a radiating structure from measurements made on a nearby surface. In patch holography the measurement surface need not surround the entire structure. It need only be approximately as large as the patch on the structure surface where the reconstruction is required. The surfaces may be planar or curved. The method of superposition is applied where the field on and near the measurement surface is approximated by the field produced by a large number of sources placed on a nearby surface inside the radiating structure. The source strengths are determined by applying boundary conditions on the measurement surface. Because of the ill-posed nature of the problem, this procedure requires the inversion of an ill-conditioned matrix. Singular value decomposition is applied to invert it. Once the source strengths are determined, the field on the partial structure surface may be approximated by the field produced by the sources. [Work supported by ONR.]

4pSA5. Assessment of accumulated structural damage with the help of linear and nonlinear acoustic techniques. Yulian Kin, Chenn Zhou, Eric Roads, Bernard Parsons (Purdue Univ. Calumet, Hammond, IN 46323), Alexander Sutin (Stevens Inst. of Technol., Hoboken, NJ), Pinakin Shaubal, and Frank Huang (Ispat Inland Steel, East Chicago, IN)

The paper presents applications of linear and nonlinear acoustic techniques for reliable detection of early stages of fracture, fatigue, and other damage, as well as investigation of structure-sensitive acoustical signatures for fundamental material research. The experimental investigations were conducted on steel samples, carbon bricks, and pavers under different types of loading (static and cycling regimes) in laboratory conditions and, in field, to detect progressive changes in a blast furnace wall. A special effort was made to investigate the changes of acoustical response after aging of materials, but before a warning of damage (necking, crack initiation, crazing, disbonding, surface flow, etc.) is physically observed in service life. Measurements were made periodically as structural damage accumulated, but no visible surface crack formation was observed during the tests. The described techniques can be used for the assessment of structure integrity, remaining life, and adjustment of maintenance schedules in applications as diverse as reactor containment wall damage, inspection of damage in buildings, bridges, furnaces, pipelines, aging aircraft, etc. [Support of Indiana 21st Century Fund is greatly appreciated.]

4pSA6. Acoustic inspection of friction-stir weld. John W. Doane (GTRI and Dept. of Mech. Eng., Georgia Inst. of Technol., 771 Ferst Dr., Atlanta, GA 30332, john.doane@gtri.gatech.edu)

Friction stir welding is gaining popularity as a joining technique in many manufacturing arenas. As it makes the transition from prototype evaluation status into mainstream, large-scale usage, there is an increasing need for comprehensive, efficient, nondestructive inspection methods for

qualifying friction stir welded joints. Two of the most common defects in this type of weld are cavities inside the weld, called wormholes, and weld depth excursions. Each of these defects presents a potentially serious reduction in joint strength and must be found soon after part production. Present inspection methods, including penetrant dyes and x-ray imaging, are either ineffective or not practical. In this paper a nondestructive inspection method, using high-frequency acoustic waves, is used to detect these defects by measuring scattered signals using a laser-based acoustic receiver system. The experimental configuration, methodology, and sensitivity of the technique to the defects of interest are discussed as well as its advantages and disadvantages compared to the present methods. [Work supported by Lockheed-Martin Co.]

4pSA7. Active vibration damping for optical tables. Vyacheslav Ryaboy (Newport Corp., 1791 Deere Ave., Irvine, CA 92606, vryaboy@newport.com)

The paper considers theoretical background, analysis and experimental verification of an active vibration damping system for optical tables. Optical table is a generic name for a stiff platform supporting vibration-sensitive equipment. A typical optical table is a sandwich structure consisting of two faceplates and a lightweight (usually honeycomb) core. Due to low weight-to-stiffness ratio, these platforms found wide application in optical research and high-precision manufacturing, usually in conjunction with soft pneumatic vibration isolators. Although excellent isolation from floor vibration can be achieved in these systems, the platform deviates from the ideal rigid-body behavior at natural frequencies of its flexural vibrations. The theoretical analysis and experimental data reported in this paper show that a properly designed active damping system consisting of a small number of collocated sensor-actuator pairs provides robust and universal means of attenuating these unwanted vibrations.

THURSDAY AFTERNOON, 18 NOVEMBER 2004

CALIFORNIA ROOM, 2:00 TO 5:00 P.M.

Session 4pSC

Speech Communication: Sounds, Speaking and the Vocal Tract (Poster Session)

Marianne Pouplier, Chair

Biomedical Sciences, University of Maryland Dental School, 666 West Baltimore Street, Baltimore, Maryland 21201

Contributed Papers

All posters will be on display from 2:00 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 2:00 p.m. to 3:30 p.m. and contributors of even-numbered papers will be at their posters from 3:30 p.m. to 5:00 p.m.

4pSC1. Production and perception of clear speech in Croatian and English. Rajka Smiljanic and Ann Bradlow (Dept. of Linguist., Northwestern Univ., 2016 Sheridan Rd., Evanston, IL 60208)

This paper presents a cross-language comparison of clear speech, a distinct mode of speech production intended to enhance intelligibility. Specifically, it explores whether language specific phonological properties guide the articulatory modifications of clear speech in Croatian and English, and examines whether clear speech enhances intelligibility in both languages. Five Croatian and five English speakers read 20 nonsense sentences in Croatian and English, respectively, in conversational and clear speaking styles. In a sentence-in-noise perception test, listeners recognized

words more accurately in clear than in conversational speech in their native language, establishing that the clear speech intelligibility effect is not English-specific. Acoustic analyses showed that both phonemic (e.g., the vowel length contrast in Croatian) and allophonic (e.g., the tense versus lax vowel duration difference in English) contrasts are enhanced in clear speech. Furthermore, speakers enhance contrasts even when categories seem to be sufficiently distinct and there is little chance of confusability. For example, the vowel space is expanded equally in English (with 11 vowels) and in Croatian (with just five vowel qualities). The results suggest that clear speech production is driven by the phonological properties of the language, but at the same time all segments are hyperarticulated regardless of inventory based considerations.

4pSC2. Talker differences in clear and conversational speech: Preliminary data with monosyllabic words. Sarah Hargus Ferguson (Dept. of Speech-Lang.-Hear., Univ. of Kansas, Dole Ctr., 1000 Sunnyside Ave., Rm. 3001, Lawrence, KS 66045, safergus@ku.edu)

Several studies have shown that when a talker is instructed to speak as though talking to a hearing-impaired person, the resulting clear speech is significantly more intelligible than typical conversational speech. A recent study of 41 talkers [S. H. Ferguson, J. Acoust. Soc. Am., in press] demonstrated that for normal-hearing listeners identifying vowels in noise, the amount of improvement talkers achieve by speaking clearly varies widely. The current pilot experiment begins to extend this line of research to meaningful materials using two talkers from the Ferguson database, one male and one female. While the clear speech vowel intelligibility benefit was similar for these two talkers, overall vowel intelligibility was greater for the female talker than for the male talker. Monosyllabic words produced in clear and conversational speech by these talkers were excised from the sentences in which they were recorded and presented in noise for identification by young, normal-hearing listeners and elderly listeners with hearing impairment. These data will provide important information about the relationship between vowel intelligibility and overall speech intelligibility. Results will be discussed in this light as well as in terms of expected talker variability in the clear speech effect for meaningful materials. [Database development supported by NIHDCD-02229.]

4pSC3. The influence of speaking rates and utterance units on segmental duration of Mandarin speech. Jing-Yi Jing (Dept. of Special Education, Natl. Taiwan Teachers College, Taiwan, jjeng@ipx.ntntc.edu.tw)

The main purpose of this study is to investigate the effect of five different speaking rates and five utterance units on segmental durations of Mandarin. Thirty normal speakers participated in the speech sampling. The speakers read the speech material at five speaking rates, including the fastest, faster, median, slower, and slowest rate. There were five utterance units for each target syllable, including monosyllable, disyllable, trisyllable, four syllable phrases, and sentences. These segments measured include VOT of stops, noise duration of fricatives and affricates, nasals and vowel duration, as well as silence intervals, or pause intervals. Results showed that the consonant and vowel durations were all significantly influenced by speaking rates and utterance units. At five kinds of speaking rates, the durations of vowels changed more than those of consonants. Fricative durations expanded more than the durations of other consonants, while the duration of unaspirated stops kept constant at five speaking rates. Vowels in monosyllable had longer duration than those in other utterance units. In addition to speaking rates and utterance units, the syllable structures and tones also greatly influenced the vowel and syllable durations of Mandarin.

4pSC4. Perception of Thai tones in citation form and connected speech. Elizabeth C. Zsiga and Rattima Nitisoroj (Dept. of Linguist., Georgetown Univ., Washington, DC 20057, zsigae@georgetown.edu)

F_0 contour shapes of the five tones of Thai differ from citation form to connected speech [Potisuk *et al.*, *Phonetica* **54**, 2242 (1997)]. “Falling” tones have a rise-fall contour in citation, but may be realized as a rise in fluent speech. This study investigates whether Thai listeners can reliably distinguish tones in connected speech, and whether any cues to tone identity remain stable across contexts. In four experiments, ten Thai listeners identified naturally produced and digitally altered tones in a forced-choice task. In experiment one, listeners identified naturally produced “falling” tones with 100% accuracy in citation forms and 96% accuracy in sentences, despite differing contours. In experiments 2–4 (replicating and extending Abramson [Lg&Sp **21**, 319325 (1978)]), F_0 onset and offset, peak height, and peak alignment were systematically modified on syllables in citation and sentence contexts. In all contexts, tones identified as high or low had an F_0 peak or valley aligned to the right edge of the syllable. Mid

tones had no F_0 inflection. Tones identified as falling or rising necessarily had a pitch inflection at the syllable midpoint (end of the first mora). These findings support the view that tones are aligned with moras in Thai.

4pSC5. The production of the stop voicing distinction in French at normal and rapid speaking rates. Nassima Abdelli-Beruh (Dept. of Speech-Lang. Pathol. and Audiol., New York Univ., 719 Broadway, Ste. 200, New York, NY 10003 and Dept. of Speech and Hear. Sci., City Univ. of New York, 365 Fifth Ave., New York, NY 10016)

This study examined how monolingual Parisian French speakers produced the stop voicing distinction at two different speaking rates (normal, rapid) in two different phonetic contexts (between vowels, between voiceless-fricatives). Results showed that all the acoustic correlates of the voicing distinction under study (syllable-initial: VOT, closure duration and closure voicing; syllable final stops: preceding vowel duration, closure duration, closure voicing and release) were shorter in the rapid speaking rate condition than in the normal speaking rate condition. Furthermore, voicing-related differences were also smaller in the rapid speaking rate condition than in the normal speaking rate condition to a degree that varied with each acoustic correlate and each phonetic context. Overall, similar patterns of voicing-related duration differences and closure voicing differences were found in the normal and rapid speaking rate conditions, but the perceptual salience of small voicing-conditioned duration differences remains to be investigated.

4pSC6. A cross-language study of the influence of duration and intensity on perception of lexical stress. Cheryl L. Moran, Amy E. Coren, Jessica F. Hay, and Randy L. Diehl (Dept. of Psych., Univ. of Texas, 1 Univ. Station A8000, Austin, TX 78712-0187, cmoran@mail.utexas.edu)

The present study examines perceived lexical stress across five languages which differ in their prosodic characteristics: English, French, German, Japanese, and Turkish. Native speakers from each language were presented with a two-syllable nonsense word, (/bibi/), and were asked to judge whether the first or second syllable was more prominent. The relative durations and intensities of the two syllables were varied orthogonally across ranges, which were determined on the basis of analyses of natural utterances from the five languages. The key issue addressed is whether judgment of syllable prominence is cross-linguistically invariant or dependent on exposure to language-specific lexical stress patterns. [Work supported by NIDCD.]

4pSC7. Envelope offset asynchrony as a cue to voicing in final english consonants. Hanfeng Yuan, Charlotte M. Reed, and Nathaniel I. Durlach (Res. Lab. of Electron., MIT, Cambridge, MA 02139)

Previous research [Yuan *et al.*, J. Acoust. Soc. Am. **113**, 2291 (2003)] has demonstrated that a simply derived, real-time measurement of the envelope-onset asynchrony (EOA) of two different bands of speech serves as a reliable cue to syllable-initial voicing. The current study was undertaken to determine the applicability of a similar type of temporal cue to voicing of final consonants. This cue is based on the time asynchrony between the offsets of amplitude-envelope signals derived from two different bands of speech (a low-pass-filtered band at 350 Hz and a high-pass-filtered band at 3000 Hz). Measurements of the offset-timing difference between the low- and high-frequency envelopes (EOFA) were obtained on a set of 16 final consonants represented through multiple tokens of C1VC2 syllables. The EOFA measurements were fit to Gaussian distributions and signal-detection theory was applied to determine the discriminability of pairs of voiced-voiceless cognates by an ideal observer. These results indicate that the EOFA cue provides a reliable and robust cue for distinguishing voicing in syllable-final position regardless of manner

or place of production. This cue has applications to the design of sensory aids for the deaf as well as to automatic speech recognition. [Research supported by NIDCD Grant R01-DC00126.]

4pSC8. Place of articulation effects on the acoustic correlates of the stop voicing distinction in French. Nassima Abdelli-Beruh (Dept. of Speech-Lang. Pathol. and Audiol., New York Univ., 719 Broadway, Ste. 200, New York, NY 10003 and Dept. of Speech and Hearing Sci., City Univ. of New York, 365 Fifth Ave., New York, NY 10016)

This study examined the influence of place of articulation on some acoustic correlates of the voicing distinction in syllable-initial (VOT, closure duration and closure voicing) and in syllable final stops (preceding vowel duration, closure duration, closure voicing and release) embedded in different French sentence contexts (between vowels, between voiceless-fricatives). Results indicated that, as in English, VOT increases as place of articulation moves back in the mouth, labial closures are longer than alveolar and velar closures and preceding vowel durations are shorter before alveolars than before labials and velars. Finally, /b/'s are significantly more often phonated than /d/'s or /g/'s in the between voiceless-fricatives context only. The effects of place of articulation on closure and preceding vowel durations are mediated by the factors of voicing category, local phonetic context and syllable-position.

4pSC9. Aerodynamic reduction in American English flapping. Sidney Martin and Joaquin Romero (Universitat Rovira i Virgili, Pl Imperial Trraco 1, 43005 Tarragona Spain, smm.predoc@estudiants.urv.es, jrog@fl.urv.es)

Are all tongue tip gestures for /t,d/ the same? Acoustic and articulatory studies have revealed that /t,d/'s show a gradient behavior, in which invariance is nonexistent. These results undermine phonological rules which represent leniting processes, such as flapping, as a rule where /t,d/ is replaced by an invariant flap. Adding to more evidence for gradiency, an aerodynamic experiment was designed to investigate the nature of alveolar tongue tip gestures, specifically in flapping contexts. Three native speakers of American English were recruited and each yielded 200 tokens, which contained medial /t,d/ in different phonetic contexts (stressed-syllable onset, unstressed-syllable onset, n+/t,d/, l+/t,d/ and r+/t,d/) and in two speaking rates. For each target consonant, duration ratio and airflow rate were obtained. In general, both duration ratio and airflow rate showed gradiency across rate and context. Tokens at a normal speech rate yielded higher ratios and airflow rates than tokens at a faster speech rate. However, rate did not affect each context equally. Accordingly, /t,d/'s reveal a gradient behavior which ranges from a canonical stop to complete deletion.

4pSC10. Greek fricatives: Inferring articulation from F2 at vowel onset. Chao-Yang Lee and Georgia Malandraki (School of Hearing, Speech and Lang. Sci., Ohio Univ., Athens, OH 45701)

Fricative consonants are produced with a narrow constriction in the vocal tract. According to the acoustic theory of speech production (Fant, 1960), fricatives are characterized by spectral prominences corresponding to the natural frequencies of the cavity anterior to the constriction. The spectral characteristics may also be modified by acoustic coupling of the noise source to the cavity behind the constriction (Stevens, 1998). In addition to the defining acoustic attributes, other articulatory gestures may be added to enhance the perceptual saliency of the fricatives (Stevens, 1989). This study examines the second formant frequency (F2) at the onset of a vowel following a fricative as an indication of tongue body position, a potential enhancing gesture for fricatives. F2 measurements were obtained from a database of 1950 tokens including all ten Greek fricatives (five places of articulation, two voicing categories) produced in five vowel contexts by 25 speakers. The results indicate highest F2 for palatals, followed by alveolars, dentals, labiodentals, and velars. Despite some vari-

ability by vowel context, the F2 values are reliably distinct among the five places of articulation. These findings are discussed in terms of the acoustic theory and the enhancement account.

4pSC11. Developmental processes in production of Korean /s/ and /s*/. Hyunjoo Chung and Anna M. Schmidt (Kent State Univ., School of Speech Pathol. and Audiol., Kent, OH 44242, hchung@kent.edu)

The purpose of this study was to investigate development of Korean children's productions of tense and lax Korean s. Korean alveolar fricatives are acquired late; /s*/ is generally later than /s/. However, little is known about developmental processes in acquiring the contrast between tense and lax Korean alveolar fricatives. Twenty typically developing monolingual Korean children between 4 and 8 and ten monolingual Korean adults were recruited in Korea. All children fell within normal limits on both a Korean articulation test and a standardized Korean vocabulary test. A picture-naming picture task elicited productions of initial fricatives, alveolar stops, and affricates followed by /a, i, u/ in monosyllabic words. Production error patterns will be analyzed to describe general developmental trends in production. More importantly, the contrast between productions of /s/ and /s*/ will be presented based on acoustic parameters including frication durations, centroid frequency, and harmonic-related amplitude measurements. Finally, similarities and differences in each acoustic parameter will be compared between adults and children. Results will discuss which acoustic parameter(s) effectively describe the distinction between productions of /s/ and /s*/ as well as differences between productions of adults and children between age 4 and 8.

4pSC12. The status of vowels in Jordanian and Moroccan Arabic: Insights from production and perception. Jaleleddin Al-Tamimi, Egidio Marsico (Laboratoire Dynamique du Langage, Institut des Sci. de l'Homme, 14, Av. Berthelot, 69007 Lyon, France, jaleleddin.al-tamimi@etu.univ-lyon2.fr), and Rene Carre (ENST, Paris, France)

Due to the morphological structure of Arabic, vowels only occur in consonantal environment, never in isolation. The issue addressed here is whether static acoustic cues are sufficient to characterize Arabic vowels or dynamic ones are needed. This study is based on Jordanian and Moroccan Arabic (JA & MA), with eight and four vowels, respectively. Ten speakers per dialect were tested on both production and perception. In production, speakers produced vowels in words, syllables, and in isolation (in alveolar context). In perception, they categorized isolated vowels on an F1/F2 plan. Results from production show that, both in JA and MA, (a) vowels in isolation have greater variation than in syllables or words, (b) vowels in a word context are more centralized (than in syllables or in isolation), and that (c) MA vowels are more centralized than JAs. In perception, a high degree of variability within and between dialects is found. A new experiment of vowel perception in syllable context (CV) (i.e., dynamic trajectories) was elaborated to study the effect of contextual information on vowel perception in Arabic. Results on five JA speakers show that the variability has been lessened and that dynamic information seems to be more useful in CV categorization.

4pSC13. Phonetic properties of three Ayuuk (Mixe-Zoquean) dense vowel systems. Avelino Heriberto (Dept. of Linguist., Univ. of California at Berkeley, avelino@berkeley.edu)

Several theories of vowel dispersion have proposed that the vowel systems of languages tend to distribute evenly in the phonetic space to maximize the perceptual distinctness of the vowels (Liljencrants and Lindblom, 1972; Lindblom, 1986; Lindblom and Maddieson, 1988; Schwartz *et al.*, 1997). This study investigates the acoustic properties of three Ayuuk (Mixe) languages spoken in Mexico, possessing dense vocalic systems, which in addition have a series of normal and the corresponding set of long vowels: Santa Mara Ocotepec Ayuuk (SMOA) with nine vowels

/i, I, u, e, E, o, A, a, O/, Tlahuitoltepec Ayuuk (TA) with eight vowels /i, I, e, E, o, A, a, O/, and Ałotepec Ayuuk (AA) with seven vowels /i, u, e, E, o, a, O/. The inventories of the Ayuuk languages are typologically interesting; the SMOA and AA systems are rather uncommon (Maddieson, 1984), in contrast with the system of TA, which conforms to the main trends of vowel systems. Preliminary results lend support to a dispersion/focalization hypothesis in that the vowels are evenly dispersed in the available space and remain focalized without significant overlap with adjacent vowels. The phonetic properties analyzed in this paper suggest that two principles, maximal dispersion and maximal distinctness, shape the overall distribution of the vowel inventory of Ayuuk languages.

4pSC14. Diphthongization as cue for the automatic identification of British English dialects. Emmanuel Ferragne and François Pellegrino (Laboratoire Dynamique Du Langage, ISH, 14 avenue Berthelot, 69007 Lyon, emmanuel.ferragne@univ-lyon2.fr)

The degree to which a given speaker of British English diphthongizes her/his vowels has been known for decades to be a good indicator of this speaker's dialectal origin: for instance, some Northern varieties tend to have variants perceived as monophthongs where Standard Southern English has a perceived diphthong in the word GOAT. A portion of the recently released Accents of the British Isles database (14 dialects recorded throughout the British Isles, 20 speakers per dialect) is investigated to determine whether this view still holds at least acoustically for contemporary dialects. Several metrics (based on formant-frequency estimation) are proposed in an attempt to capture a "degree" of diphthongization. In a preliminary and rather oversimplified study, the standard deviation of $F2$ was measured over the whole duration of the vowel in the word BOAT for all speakers and dialects. A nonparametric one-way analysis of variance was performed to test the effect of "dialect" on diphthongization. The effect turned out to be significant at the 0.01 level. More elaborate procedures (including all first three formants) will be shown in the final presentation.

4pSC15. Nasalization, neutralization, and merger in English front vowels. Jerry Liu (Dept. of Linguist., USC, 3601 Watt Way, GFS 301, Los Angeles, CA 90089-1693, jerryliu@usc.edu)

Before a velar nasal, a tense/lax contrast in the front vowels is generally considered to be conflated to only the set of lax vowels in American English. However, Ladefoged (2001, *A Course in Phonetics*) states that many younger Americans pronounce "sing" with a vowel closer to that in "beat" rather than to that in "bit." This paper examines front vowels produced by nine young Californians before the velar nasal to document whether or not the vowel, in fact, raises. Two outcomes are possible: (1) a front and raised vowel intermediate between its tense and lax counterparts (which we call neutralization) or (2) a lax vowel (which we call merger). Further, the pattern for both high- and midfront vowels is examined. The results demonstrate a consistent pattern of neutralization (i.e., an intermediate vowel) for both the high- and mid-vowels, the latter a previously unattested environment for this pattern of raising before velar nasals.

4pSC16. An ultrasound and tagged cine MRI study of German vowels. Marianne Pouplier (Vocal Tract Visualization Lab, BMS, UMAB Dental School, 666 W Baltimore St, Rm 4-D-20, Baltimore, MD 21201; mpoup001@umaryland.edu), Adam Buchwald (Johns Hopkins, Baltimore, MD 21218), and Maureen Stone (UMAB Dental School, Baltimore, MD 21201)

German vowels contrast along the tense-lax as well as the rounding dimension, rendering a relatively complex (front) vowel system. The difference between tense and lax vowels is often described in terms of more extreme articulator movement and greater muscular tension for tense vowels, manifest especially in tongue root displacement and tongue height. These factors are assumed to correlate in the Germanic languages. This

study uses ultrasound data to investigate global tongue shape and in particular tongue root differences for German monophthongs in stressed position. Preliminary results for one speaker indicate that some vowel pairs are nearly identical in tongue height, but differ in tongue root displacement. Rounded front vowels are more retracted and lower than their unrounded counterparts. The /a: a/ pair, which has been described as contrasting in duration only, exhibits a clear qualitative difference as well. In addition, tagged cine MRI data were collected for a subset of high and mid vowels. This technique allows inferring expansion and contraction of the tongue musculature and thus can inform about differences in muscular tension between tense and lax vowels. [Supported by NIH Grant RO1-DC01758.]

4pSC17. 3D tongue segmentation for a minimal pair sentence. Melissa A. Epstein and Maureen Stone (Biomed. Sci., Rm. 5A12, Univ. of Maryland Dental School, 666 W. Baltimore St., Baltimore, MD 21201)

The tongue appears to have as many as five articulatory control regions. If this is true, each of these regions of the tongue, or segments, could be independently controlled. What does it mean to be independently controlled? Independent control would be manifested as low correlations between segments, because independent segments could move asynchronously in speech. Previous studies for midline data have supported this and have shown that segments can aggregate or deaggregate dependent on the phonemic content of the speech sample. However, the tongue is a 3D structure, and its muscles are more localized lateral-to-medial than anterior-to-posterior. Therefore, additional lateral units of control are possible [Stone, J. Acoust. Soc. Am. (1990)]. This study looks at correlations between front to back segments of midline points, lateral points, and 3D shape data to determine whether lateral segments need to be considered independently of midline segments. We will also consider what features besides functional independence might cause low correlations for midline or lateral segments.

4pSC18. An acoustic and perceptual study of voiced geminates in Japanese. Shigeto Kawahara (Dept. of Linguist., UMass, Amherst, MA)

In Japanese, voicing in geminates is subject to dissimilation while voicing in singletons is not. This paper advances a hypothesis that voicing in geminates is not as perceptually salient as voicing in singletons, and therefore losing voicing in geminates is accepted as an instance of "perceptually tolerated articulatory simplification" (Kohler, 1990), while devoicing singletons is too perceptually conspicuous. This paper reports an acoustic and perceptual experiment that supports this hypothesis. The acoustic study shows that closure voicing and closure duration are significantly weakened in geminates: Voicing ceases relatively early during the closure, and the difference between voiced and voiceless geminate closure length is smaller than the closure difference in singletons. The second experiment is a perceptual study, which is an identification task under a noisy environment. The results directly support the hypothesis that voicing in geminates is hard to hear, while voicing in singletons is reliably perceived. Taken together, the results of the two experiments provide support for the view that voicing in geminates may be prone to neutralization because it is poorly perceived. This conclusion, more generally, lends support to the view that contrasts signaled by weak phonetic cues are prone to phonological neutralization (Kohler, 1990; Steriade, 1997).

4pSC19. Movement duration and movement stiffness in long and short consonants. Anders Lofqvist (Haskins Labs., 270 Crown St., New Haven, CT 06511; lofquist@haskins.yale.edu)

This study examined the relationship between movement duration and movement stiffness in speech. Stiffness was measured as the interval from movement onset to peak velocity. The speech material consisted of long and short Japanese consonants produced by native speakers. A magnetometer was used to track articulatory movements. For the lower lip, the

velocity of vertical closing movement for bilabial consonants was measured, while the speed of the tongue tip was measured for dental stops and fricatives. Results from six subjects show that the lower lip movement duration was longer for the long consonants, which also had a lower stiffness than their short cognates. The results also show higher correlations between duration and stiffness for the short consonants. The results for the tongue tip showed more variability across and within subjects, although the overall trend was similar to the one found for the lower lip. In addition, all correlations between tongue tip movement duration and stiffness were high. These results extend earlier work on the same relationship, where it has mostly been studied across variations in speaking rate. They also suggest potential differences between articulators and between stops and fricatives. [Work supported by NIH.]

4pSC20. Perturbation and compensation in speech acoustics using a jaw-coupled robot. Mark K. Tiede (R.L.E. Speech Comm. Group, MIT, Cambridge, MA, tiede@speech.mit.edu, and Haskins Labs, New Haven, CT), Frank H. Guenther (Boston Univ., Boston, MA), Joseph S. Perkell, Majid Zandipour (MIT, Cambridge, MA), Guillaume Houle, and David J. Ostry (McGill Univ., Montreal, PQ, Canada)

Observations were made in three speakers of compensation in formant trajectories in response to jaw perturbations during utterances with the general form /siyCVd/, as in “see red.” Custom dental prostheses were used to help immobilize the head (upper jaw) and couple a computer-controlled robotic device (lower jaw). A 3-Newton perturbation force was applied to the jaw during one out of every five repetitions, selected at random, with half of the perturbations applied downward and half upward. Perturbations were triggered from jaw opening (for CV) exceeding a threshold relative to clench position. Audio (at 10 kHz) and jaw position (at 1 kHz) were recorded concurrently. Individual tokens were extracted using the perturbation threshold for alignment. Formants computed over these intervals show initial deviation from control trajectories and then compensation that begins 60–90 ms after perturbation. Since jaw position does not recover its unperturbed trajectory, compensation presumably is effected through modified tongue movements. The observed behavior is compatible with the function of the DIVA model of speech motor planning, in which corrective motor commands are computed in response to errors between anticipated and produced sensory (auditory and somatosensory) consequences. [Research supported by NIDCD.]

4pSC21. Effects of jaw perturbation on cortical activity during speech production. Jason A. Tourville, Frank H. Guenther, Satrajit S. Ghosh, and Jason W. Bohland (Dept. of Cognit. and Neural Systems, Boston Univ., 677 Beacon St., Boston, MA 02215, jtour@cns.bu.edu)

Functional magnetic resonance imaging (fMRI) was used to investigate cortical activation caused by introducing a perturbation to the jaw during speech. The DIVA model of speech production [e.g., Guenther and Ghosh (2003), Proceedings of the ICPhS. Barcelona, ICPhS Committee] predicts that jaw perturbation will cause a mismatch between learned target (expected) sensations and actual sensory inputs in higher-order somatosensory cortical areas in the supramarginal gyrus, and that this mismatch will drive compensatory activations of motor cortical areas. To test this hypothesis, we performed an fMRI experiment in which articulation of pseudo-words was perturbed by the rapid inflation of a small, stiff balloon held between the subjects molars. The perturbation condition occurred pseudo-randomly on $\frac{1}{5}$ of the trials. After controlling for brain activation due to normal speech production (no perturbation) and due to the sensation of balloon inflation alone (no production), two regions responded preferentially to the perturbed speech condition: one region along the boundary of the angular gyrus and posterior supramarginal gyrus in inferior parietal cortex, and another in the ventral posterior inferior frontal gyrus extending into the frontal operculum and anterior insula. These results provide evidence for the type of speech motor control processes hypothesized by the DIVA model.

4pSC22. Sound propagation through the large front-cavity volume of American English rhotic sounds. Zhaoyan Zhang, Carol Espy-Wilson (Dept. of Elec. and Computer Eng., University of Maryland, College Park, MD 20742), Suzanne Boyce (Univ. of Cincinnati, Cincinnati, OH 45202), and Mark Tiede (Haskins Labs, New Haven, CT 06511)

Rhotic sounds in American English (AE) are produced with a large front-cavity volume between the palatal constriction and the lip constriction which is the source of the characteristic low F_3 of /r/'s. Depending on its shape, this volume may be modeled as (a) a single tube or (b) a single tube with a side branch. Both models produce a low F_3 , but through different mechanisms that involve differences in wave propagation direction. In this paper, we describe finite element studies of the acoustics of the front cavity, using dimensions derived from MRI studies of different subjects producing AE /r/. In particular, we discuss the case of a front cavity with minimal sublingual space. The sound-pressure distribution in the front cavity and the possible sublingual cavity was calculated at various harmonic excitations. The results show that at low frequencies (up to 2000 Hz, depending on specific dimensions) the front cavity is better modeled as a single tube, and at higher frequencies as a single tube with a side branch. The implications of accounting for wave propagation patterns (and thereby acoustically critical points along the vocal tract) on extraction of dimensional measures from MRI data will be discussed.

4pSC23. A formant-to-area conversion technique based on acoustic sensitivity functions. Brad Story (Univ. of Arizona, Dept. of Speech and Hearing Sci., P.O. 210071, Tucson, AZ 85721-0071)

A technique for transforming formant frequencies to vocal tract area functions is developed by using sum and difference combinations of acoustic sensitivity functions to perturb an initial vocal tract configuration. First, sensitivity functions [e.g., Fant and Pauli, Proc. Speech Comm. Sem., Vol. 74 (1975)] are calculated for a neutral area function, at its specific formant frequencies. The sensitivity functions are then multiplied by scaling coefficients that are determined from the difference between a desired set of formant frequencies and those supported by the current area function. Then, the scaled sensitivity functions are summed together to generate a perturbation of the area function. This produces a new area function whose associated formant frequencies are closer to the desired values than the previous one. This process is repeated iteratively until the coefficients are equal to zero or are below a threshold value. Results show that vowel area functions are generated that are comparable to those measured with imaging techniques. The performance of the technique with time-varying formant frequencies will also be discussed. [Work supported by NIH R01-DC04789.]

4pSC24. Using phonetic constraints to improve inversion. Blaise Potard and Yves Laprie (LORIA, Campus Scientifique - BP 239, 54506 Vandoeuvre-ls-Nancy Cedex, France)

The goal of this work is to recover articulatory information from the speech signal by acoustic-to-articulatory inversion. One of the main difficulties with inversion is that the problem is under-constrained and inversion methods generally offer no guarantee on the phonetical realism of the inverse solutions. A way to address this issue is to use additional constraints: in this work, constraints on phonetically relevant characteristics of the vocal tracts. Knowledge of the phonetic characteristics of most French vowels enables the derivation of reasonable articulatory domains in the space of Maeda parameters: given the formants frequencies (F_1 , F_2 , F_3) of a speech sample, and thus the vowel identity, an “ideal” articulatory domain can be derived. The space of formants frequencies is partitioned into vowels, using either speaker-specific data or generic information on formants. Then, to each articulatory vector can be associated a phonetic score varying with the distance to the “ideal domain” associated with the corresponding vowel. Inversion experiments were conducted on two different speakers (male and female), using different implementations of

these phonetical constraints. The articulatory trajectories were compared with those obtained without using these constraints and to articulatory trajectories measured from x-ray data.

4pSC25. The role of principal component analysis in vowel acquisition research. Krisztina Zajdó (Univ. of Wyoming, Div. of Commun. Disord., Dept. 3311, 1000 E. Univ. Ave., Laramie, WY 82071, kzajdo@uwyo.edu), Jeannette van der Stelt, and Ton G. Wempe (Univ. of Amsterdam, Amsterdam, 1016 CG, The Netherlands)

Traditional formant analysis techniques have yielded questionable results, due to a high fundamental frequency in young children's speech. Therefore, an alternative approach for researching vowel acquisition appears desirable. A band-filtering analysis method was developed that mini-

mizes the dependence of the results on F_0 to measure the spectral envelopes in children's utterances automatically. In PRAAT scripts, first the criteria are set with regard to sound quality (to avoid clipped fragments) and F_0 (to avoid vowel fragments with F_0 above 425 Hz). Secondly, the spectral envelopes covering a range from 0 to 7 kHz, resulting in 40 filter values per spectrum, are calculated. A matrix is produced, representing intensity in each of the 40 filters. Data reduction is achieved via principal component analysis. Previous research (Pols, 1977) suggests that the first two principal components are related to the F_1 and F_2 values. Analyses of unlabeled vowel productions in 2-year-old Hungarian- and Dutch-speaking children show that the first two eigenvectors account for 55% of the variability. By using these two eigenvectors, a reference plane is created. Mapping labeled vowel measures onto the plane suggests that children produce language-specific vowel categories at this young age. [Work supported by UW and UvA/ACLC.]

THURSDAY AFTERNOON, 18 NOVEMBER 2004

ROYAL PALM SALONS 3 & 4, 2:00 TO 4:05 P.M.

Session 4pSP

Signal Processing in Acoustics, Underwater Acoustics, Animal Bioacoustics, Noise, Acoustical Oceanography and Engineering Acoustics: Time Delay Estimation, Localization and Tracking in Acoustics Part II

George E. Ioup, Cochair

Department of Physics, University of New Orleans, New Orleans, Louisiana 70148

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Chair's Introduction—2:00

Contributed Papers

2:05

4pSP1. Video/acoustic-array studies of swarming by echolocating bats. James A. Simmons, Kyler M. Eastman, Gregory Auger (Dept. of Neurosci., Brown Univ., Providence, RI 02912, james_simmons@brown.edu), Michael J. O'Farrell (O'Farrell Biological Consulting, Las Vegas, NV 89131), Alan D. Grinnell (UCLA, Los Angeles, CA 94720), and Donald R. Griffin (Harvard Univ., Bedford, MA)

Echolocating bats (*Pipistrellus*, *Myotis*) sometimes fly in dense swarms for 1–2 h over desert water sources. Swarming is different from streaming at caves or roosts because ~10–100 bats staying within a small volume (~1000 m³) for an extended time rather than transiting. We study swarming using thermal infrared video cameras and a three-microphone acoustic array. Swarming bats emit a steady stream of short (~2 ms) FM sonar sounds with nearly identical sweeps and a collective duty-cycle approaching 100%. Surprisingly, given the tendency to think in terms of bats in close proximity deliberately varying their signals for jamming avoidance, the signals of swarming bats are very similar. Video/acoustic tracks reveal no sign of mutual interference or collisions. It is not obvious whether bats gather to drink or to feed on insects, which also congregate near the water, or whether social behavior predominates. Some drinking occurs throughout the swarming period, but the occurrence of captures during swarming is negligible, at least judging from the absence of feeding buzzes. The bats may use multistatic sonar, in which each bat takes ad-

vantage of the sounds of other bats as helpful additional ensonification rather than interference. [Work supported by NIMH, DoD, Brown University.]

2:20

4pSP2. A generalized Cramer–Rao lower bound for line arrays. E. J. Sullivan (EJS Consultants, 46 Lawton Brook Ln., Portsmouth, RI 02871, paddy priest@aol.com) and G. S. Edelson (BAE SYSTEMS, Nashua, NH 03061-0868)

The Cramer–Rao lower bound (CRLB) on the variance of an estimate is a consequence of the underlying likelihood function. That is to say, the more realistic the likelihood function, the more realistic the bound. It is shown here that by including the forward motion of a line array in the likelihood function for the bearing of a continuous broadband signal, the CRLB on the bearing estimate is lower than that found for the case of the stationary array. This is a consequence of the fact that there is bearing information contained in the Doppler that is not exploited by conventional beamformers. Further, it is shown that this improvement in performance requires that a nuisance parameter in the form of a source frequency be jointly estimated along with the bearing. An example is given which uses a recursive processor and a synthetic broadband signal from a nonmoving source. It is shown that the performance improvement relative to a stationary array of equal physical aperture is a function of the array speed, the integration time, the bearing angle and the direction of motion of the array, i.e., whether the motion incurs “up” or “down” Doppler.

4pSP3. Cramer–Rao lower bounds applied to shallow water geoaoustic parameter estimation using the distribution of the interference invariant. Peter M. Daly and Peter N. Mikhalevsky (Sci. Applications Intl. Corp., 1710 SAIC Dr., MS 1-11-15, McLean, VA 22102, peter.m.daly@saic.com)

One can exploit the broadband acoustic striation patterns produced by loud merchant ships in shallow water to obtain geoaoustic parameters. One measures these patterns as a function of range and frequency, calculates their two dimensional Fourier transform to produce intensity as a function of wavenumber and delay, then performs an inverse Radon transform to obtain the distribution of the interference invariant. With this observable, one can perform a global inversion for parameters of interest, such as sediment and bottom sound speed, density, and attenuation. If the parameter estimate is unbiased and the observable vector Gaussian has high signal-to-noise ratio, then one can obtain the theoretical minimum variance and covariance associated with the parameters by solving for the Cramer–Rao lower bound (CRLB). In this presentation, the parameters of interest are assumed to be deterministic and observed “noise” is a function not only of additive Gaussian noise at the hydrophone, but also due to uncertainty in other model parameters, such as hydrophone depth and range bias. This talk will present numerically calculated CRLB as a method for determining the performance of this inversion method.

4pSP4. Localization and tracking techniques for navigation and guidance of underwater autonomous sensing systems. Hua Lee (Univ. of California, Santa Barbara, CA 93106)

The key to the successful execution of autonomous underwater imaging and sensing is the localization and tracking of the data-acquisition units, which can be achieved by estimating the relative location with respect to a set of underwater base stations. In a three-dimensional underwater acoustic localization, the objective parameters are the range distance and a two-dimensional bearing angle. At the base stations, the transmitters send a sequence of guidance waveforms. One signal is a pair of polarity signals, with a 180° left–right phase offset, for the estimation of the bearing angle in the horizontal direction. Similarly, another one is with a top–bottom polarity for the estimation of the bearing angle in the vertical direction. The received signals represent spatial-temporal samples of the interferences produced by the polarities of the transmitted signals. The range distance can be estimated from the slope of the phase profiles of the spectra. And the bearing angle is directly related to the variation of the complex amplitude of the received polarity signals and can be estimated accordingly. This paper includes system design, signal analysis, development of processing algorithms, and results from full-scale experiments. [Research is supported by the UC MICRO program and Sonatech.]

4pSP5. Source localization in shallow water using artificial neural network. Gee-Pinn James Too and Eshine Lin (Dept. of Syst., and Naval Mechatronic Eng., National Cheng Kung Univ., Tainan, Taiwan 701, z8008070@email.ncku.edu.tw)

This study presents a method of passive localization for shallow water acoustic sources based on ray acoustic theory. ray acoustic theory is used to establish an underwater acoustic channel and to analyze physical quantities (arrival time and arrival angle) received by sensors, and to determine

mutual relationship of rays between sources and receivers by using an artificial neural network for source localization. This ray-acoustic-based method of passive localization features simplicity, less calculation, and robust performance to environment variations. Hence, the method is more practicable and more valuable for source localization. In this study, the ray-acoustic-based method of passive localization is not only used in shallow water, but it is also verified by using a nontrained source to prove its accuracy and reliability. [Work sponsored by National Science Council.]

4pSP6. Subsurface flaw localization based on the spatio-temporal gradient array processing. Kenbu Teramoto (Dept. of Mech. Eng., SAGA Univ., Saga-shi, 840852 Japan, tera@me.sga-u.ac.jp)

Lamb waves are guided ultrasonic waves capable of propagating relatively long distances in plates and laminated structures, such as airframe skins, storage tanks and pressure vessels. Their propagation properties in these media depend on the vibrational frequency as well as on the thickness and material properties of the structure. Structural flaws such as dis-bonds, corrosion and fatigue cracks represent changes in effective thickness and local material properties, and therefore measurement of variations in Lamb wave propagation can be employed to assess the integrity of these structures. In this study, the localization based on the spatio-temporal gradient array processing is derived theoretically. The proposed method has an ability to characterize the phase velocity of the Lamb-wave propagation through the linearity among the 4-dimensional vector which is composed by following components: (1) a vertical (z-directional) displacement, (2) its vertical particle velocity, (3) x-directional and (4) y-directional out-of-plane strains. This paper is organized as follows: the second section presents the process of the localization and the proposed algorithm, the third section evaluates the proposed velocimetry via numerical experiments.

4pSP7. Acoustic system for aircraft detection and tracking based on passive microphones arrays. Gaetano Caronna, Ivan Roselli, Pierluigi Testa (Dept. of Tech. Phys., Faculty of Eng., Univ. of Rome “La Sapienza,” Via Eudossiana 18 - 00184, Rome, Italy), and Andrea Barbagelata (D’Appolonia S.p.A., 16145, Genoa, Italy)

The present study is conducted within a project included in the Sixth Framework Program of the European Union. The project, whose acronym is SAFE-AIRPORT, involves the development of an acoustic system based on two passive phased array microphone antennas, which are to be located in open environments, and capable of detecting and tracking airplanes at a distance in air and on ground. Each antenna array is made up of a series of microphones arranged on a rotating planar support. The Array’s spatial selectivity is discussed, aiming to optimize beamforming design according to spectral properties of common jet aircrafts. In particular, the influence of the number and the geometric distribution of microphones is shown in function of the target’s acoustic frequency and signal-to-noise ratio sensitivity. The system’s range is estimated at different working frequencies and for different positions of the aircraft on the horizon. The effect of wind gradients on the estimation of the aircraft position is also modeled, and computer simulations showing consequent deviation of acoustic waves are performed. Finally, possible developments in antenna design, such as the use of three fixed planar arrays in tetrahedral configuration instead of one rotating planar array, are evaluated showing expected advantages.

Session 4pUW

Underwater Acoustics: Long-Range Acoustic Propagation Part II

Peter F. Worcester, Chair

*Scripps Institution of Oceanography, University of California San Diego, 9500 Gilman Drive,
La Jolla, California 92093-0225**Invited Paper*

1:30

4pUW1. 6+ years of acoustic thermometry in the North Pacific Ocean: A cycle of the Pacific decadal oscillation. Brian D. Dushaw and the NPAL Group^{a)} (Appl. Phys. Lab., 1013 N.E. 40th St., Seattle, WA 98105-6698, dushaw@apl.washington.edu). Acoustic measurements of basin-averaged temperatures have been made in the North Pacific over the last 8 years. The acoustic observations give integral measurements of ocean temperature with the averaging needed to detect weak, gyre-scale signals out of the dominant mesoscale variability. An acoustic source on Pioneer Seamount off California transmitted to receivers distributed throughout the North Pacific from 1996 to 1999 as part of the Acoust. Thermometry of Ocean Climate (ATOC) project. A second source, located near Kauai, began transmitting in 1997. The Kauai acoustic transmissions are part of the North Pacific Acoust. Lab. (NPAL) project. A 6-year time series has been obtained with sampling six times per day every 4 days. The cabled-to-shore source and receivers provide near real-time data. Long-period changes in temperature are apparent in the time series from the Kauai source. The paths from Kauai toward the California coast show steady cooling. A path to the northwest showed warming until early 2003, when the "normal" annual cycle returned. Comparisons of the acoustic data to observations by altimetry and Argo floats are not definitive because of the great differences between data types. ^{a)} (J. Colosi, WHOI; B. Cornuelle, SIO; B. Dushaw, APL-UW; M. Dzieciuch, SIO; B. Howe, APL-UW; J. Mercer, APL-UW; R. Spindel, APL-UW; P. Worcester, SIO)

Contributed Papers

1:55

4pUW2. Raytubes in an ocean acoustic environment. Matthew Dzieciuch and Bruce Cornuelle (Scripps Inst. of Oceanogr./UCSD, 9500 Gilman Dr., La Jolla, CA 92093)

In ocean acoustic tomography, sound propagation is typically modeled by ray propagation, which allows an easy answer to the question, "Where does the sound go?" Sound obeys the wave equation and ray propagation is only valid in the high frequency limit. It is still possible to answer the question by generating a raytube using the method outlined by Bowlin [J. Acoust. Soc. Am. **89**, 2663–2669 (1991)] for environments where full wave modeling is appropriate. It is thus possible to ask how the width of the raytube changes with frequency, bandwidth, and range. Perhaps more importantly, from an acoustical oceanography point of view, how does the environment impact the width? We investigate the effects of the strength of the internal wave field, the depth of the mixed layer, and the presence of ocean spiciness on the raytube width.

2:10

4pUW3. Sensitivity kernels for finite-frequency ocean acoustic observables. Michael Vera (College of Sci. and Technol., Univ. Southern Mississippi, 730 E. Beach Blvd., Long Beach, MS 39560), Emmanuel K. Skarsoulis (Inst. of Appl. and Computational Mathematics, 711 10 Heraklion, Crete, Greece), and Bruce D. Cornuelle (UC San Diego, La Jolla, CA 92093-0230)

Wave theoretic modeling was applied to obtain sensitivity kernels representing the amount by which ocean acoustic observables, such as pressure, intensity, or arrival times, are affected by localized sound-speed variations anywhere in the medium. These kernels can be derived using the first Born approximation, yielding an integral representation of observable perturbations as functionals of sound-speed perturbations. The kernel in each integral represents the spatial sensitivity of the observable to

sound-speed perturbations in the medium. The travel-time sensitivity kernel represents a full-wave generalization of the highly localized ray kernel from geometric optics. This work extended recent wave-theoretic results for the travel-time sensitivity kernel for short-range propagation in simple environments to long-range transmissions in more complicated ocean environments relevant to ongoing propagation and inversion experiments. It was shown that the geometry of the wave-theoretic sensitivity kernels is related to the geometry of Fresnel volumes surrounding eigenrays, provided that the effects of refraction are taken into account for the calculation of the latter. [Work supported by ONR.]

2:25

4pUW4. Spice, internal waves, and sound speed in the upper ocean. Daniel L. Rudnick (Scripps Inst. of Oceanogr., UCSD, Mail Code 0213, La Jolla, CA 92093-0213)

Sound-speed variations in the upper ocean may be caused by spice fluctuations or isopycnal tilt associated with internal waves. Given finely resolved fields of temperature and salinity, the relative effects of spice and tilt on sound-speed can be ascertained. For example, in a 1000-km section in the North Pacific thermocline, root-mean-square sound-speed variability in the horizontal caused by spice is roughly one-half that caused by tilt. Sound-speed variability is almost exclusively caused by spice in the nearly unstratified mixed layer. During the June 2004 North Pacific Acoustics Laboratory (NPAL) Spice04 mooring deployment cruise, fine-scale hydrographic observations were made with the newly developed Underway CTD. Observations on a horizontal scale of 10 km and a vertical scale of 5 m show strong spice-caused sound-speed variability in the remnant of the winter-mixed layer. The largest sound-speed changes were greater than 6 m/s in a horizontal distance of tens of kilometers. The effect of these changes on acoustic propagation is a topic of continuing study.

4pUW5. Abyssal monitoring of upper ocean processes. Walter H. Munk, Matthew A. Dzieciuch, and Peter F. Worcester (Scripps Institution of Oceanogr., Univ. of California at San Diego, La Jolla, CA 92093-0225)

The transition from the early surface-reflected to near-surface-refracted waves is associated with a triplication pattern of arrivals which is very sensitive to upper ocean processes, including mixed layer formation, spicy fronts and internal waves. The pattern is well sampled at surface-conjugate depths (3 to 5 km). Deep and relatively short vertical arrays operating in the still, uniform abyssal environment are an attractive alternative to shallow *in situ* measurements.

4pUW6. Underwater sound source with tunable resonator for ocean acoustic tomography. Andrey K. Morozov and Douglas C. Webb (Webb Res. Corp., 82 Technol. Park Dr., East Falmouth, MA 02536, moro@webbresearch.com)

A high Q tunable resonator makes possible the construction of an underwater sound source with a very high efficiency. Tuning this narrow-band resonator to match the instantaneous frequency of a reference signal enables control of sound projection over a wide spectrum width. At any given time the source has one fundamental resonant frequency equal to the instant frequency of the frequency modulating signal and the sound radiates with maximum efficiency. Such sound sources find application in ocean acoustic tomography and deep-penetration seismic profiling. The analysis of the simplest known Helmholtz resonator shows its ability to instantaneously switch its frequency or to quickly adjust to a changing reference frequency. From this simplest lumped resonant source the consideration then turns to a complicated distributed system, such as a tunable organ pipe. The analysis shows that the organ pipe can also be quickly swept and instantly switch the radiated signal frequency. The 200–300-Hz swept frequency sound source was designed for ocean acoustic tomography. Test results for this sound source are presented showing that it has a high efficiency, up to 50%, 3 dBi directivity gain, and depth independent parameters.

4pUW7. A simple transformation of the environment eliminates standard PE phase errors. Irina I. Rypina, Ilya A. Udovydchenkov, Michael G. Brown, and Francisco J. Beron-Vera (RSMAS/AMP, Univ. of Miami, 4600 Rickenbacker Cswy., Miami, FL 33149, irypina@rsmas.miami.edu)

It is shown that, in range-independent sound channels with a single sound speed minimum, the environment can be transformed in such a way that solutions to a standard parabolic wave equation in the transformed environment are asymptotically equivalent to solutions to the Helmholtz equation in the original environment. This equivalence is demonstrated using both ray- and mode-based wavefield expansions. Reasons that equivalence between solutions to the Helmholtz equation and the parabolic wave equation can only be asymptotic will be briefly reviewed. [Work supported by ONR and NSF.]

4pUW8. A very simple transformation of the environment significantly reduces Thomson–Chapman PE phase errors. Ilya A. Udovydchenkov, Irina I. Rypina, Michael G. Brown, and Francisco J. Beron-Vera (RSMAS/AMP, Univ. of Miami, 4600 Rickenbacker Cswy., Miami, FL 33149, ilya@rsmas.miami.edu)

In a range-independent environment with a single sound speed minimum, a transformation of the environment is available which guarantees that solutions of the standard parabolic wave equation in the transformed environment are asymptotically equivalent to solutions of the Helmholtz equation. In this paper an approximate transformation, to be used in conjunction with the Thomson–Chapman parabolic approximation, is described. Advantages of the approximate transformation over the exact transformation are its simplicity and the fact that it can be used in environments with multiple sound speed minima. Travel time errors, relative to the corresponding Helmholtz equation rays, associated with this approximation in typical deep ocean environments grow approximately linearly in range at a rate of approximately 0.1 ms/Mm. [Work supported by ONR and NSF.]

4pUW9. LOAPEX: The long-range ocean acoustic propagation experiment. James A. Mercer (Appl. Phys. Lab., Univ. of Washington, Seattle, WA 98105)

This paper describes the recent long-range ocean acoustic propagation experiment (LOAPEX) and presents initial results. LOAPEX was conducted between 10 September and 10 October 2004, and was closely coordinated with two other experiments, BASSEX and SPICEX. LOAPEX provided low-frequency broadband acoustic transmissions to vertical hydrophone arrays covering most of the deep ocean water column, and to horizontal hydrophone arrays surrounding the Eastern North Pacific Ocean. Two of the primary objectives are to explain the strong acoustic arrivals that are received in the deep ocean shadow zone, and to better understand the roles of scattering and diffraction in general. Transmission paths to the vertical hydrophone arrays varied from 50 to 3200 km. [Work supported by the Office of Naval Research, Code 3210A.]

4pUW10. Prescription frequency-modulated signal design for the long-range ocean acoustic propagation experiment (LOAPEX). Rex K. Andrew, James A. Mercer, and Bruce M. Howe (Appl. Phys. Lab., 1013 NE 40th St., Seattle, WA 98105)

A “prescription frequency-modulated” signal is a swept FM signal whose sweep rate and possibly envelope are adjusted according to some *a priori* prescription. An idea advanced by Birdsall and Metzger in 1994 for the ATOC long-range tomography experiment, the concept is being revisited for LOAPEX as an alternative to “m-sequences.” The prescription can be used to tailor the output spectrum to a preferred shape: the sweep rate is slower at frequencies where more spectral energy is needed, and faster where less is needed. The prescription can also compensate for the transducer response. In addition, the prescription can drive the envelope to predefined maximum levels at every frequency, producing a signal which is “optimal” with respect to drive level. This signal is in general far less stressful on transducers than standard m-sequences. Examples of prescription FMs used during the LOAPEX experiment will be presented. [Work supported by ONR.]

4pUW11. The long-range ocean acoustic propagation experiment (LOAPEX) observable: Modal content as a function of arrival time.

Part A: Relevance. Frank S. Henyey and Michael A. Wolfson (Appl. Phys. Lab., Univ of Washington, 1013 NE 40th St., Seattle, WA 92105)

It is well known that for gradual enough changes in the sound speed structure, the mode number is nearly constant. Thus, changes in mode number are the most direct measure of scattering by smaller structures, such as internal waves. The mode number changes can be determined from data by separating the signal by arrival time. It is shown how the separation happens in an ideal case without scattering, allowing an optimum choice of the time window. The modal content in a small time window can also be estimated from the depth dependence of the intensity in the region of the turning depths of the modes. Each mode has a rapid falloff near its turning depth, and the extension beyond that depth measures the higher mode content. The deep vertical line array at LOAPEX measured this depth dependence. In addition, for certain times, all of the

arrival is in the deep array, allowing unambiguous projections onto modes, especially if the source is above the sound axis. [Work supported by the Office of Naval Research.]

4pUW12. The long-range ocean acoustic propagation experiment (LOAPEX) observable: Modal content as a function of arrival time.

Part B. Simulation results. Michael Wolfson and Frank Henyey (APL/UW, Seattle, WA 98105)

Following the approach described in Part A, we present the acoustic mode content for fixed frequency, range, travel time, and time window size by performing broadband full-wave simulations using a wide-angle, parabolic equation model. We use a realistic stratification and internal wave spectrum to predict the acoustic arrival structure. The modal content is extracted using the same technique as we intend to use on the LOAPEX vertical line arrays (especially the deep array). Comparing simulations with and without internal waves reveals the additional modal spreading due to scattering.