Session 3aAA

Architectural Acoustics and Musical Acoustics: Variable Acoustics on Concert Stages I

Bill Dohn, Cochair
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Michelle C. Vigeant, Cochair
Mechanical Engineering, Univ. of Hartford, 200 Bloomfield Ave., West Hartford, CT 06117-1599

Chair's Introduction—8:00

Invited Papers

8:05

3aAA1. Orchestra canopies. Magne Skalevik (AKUTEK/BrekkeStrand, Bolstadtunet 7, 3430 Spikkestad, Norway, magne.skalevik@brekkestrand.no)

Looking around, one might get the impression that a canopy has become a "must have" in every new concert hall and a magic recipe in any refurbishment project. There are, however, cases where canopies would be unwanted, unnecessary, insufficient, inadequate, or even detrimental to the acoustics on stage or in the rest of the hall. Some of the highest rated concert halls among musicians and audience, Musikvereinsaal in Vienna, Concertgebouw in Amsterdam, and Boston Symphony Hall, all fail to have a canopy. Still, there are many halls where the use of a canopy is justified. This paper points at the importance of starting by defining the requirements to the acoustics on stage (and in the rest of the hall), and then the question how these requirements could be met. A canopy may be the answer, maybe not. Significant features, acoustic properties, design issues, and parameters for prediction and measurements of canopy performance will be discussed. In short: Orchestra canopies—what difference can they make?

8:25

3aAA2. A new active acoustics system for enhancing musician's environment on concert and recording stages. Wieslaw Woszczyk (McGill Univ., CIRMMT and Schulich School of Music, 555 Sherbrooke St., West, Montreal, QC H3A 1E3, Canada, wieslaw@music.mcgill.ca), Doyuen Ko, and Jonathan Hong (McGill Univ., Montreal, QC H3A 1E3, Canada)

The necessity of variable acoustics in modern performance spaces is widely acknowledged; particularly, there has been a growing interest in active acoustics enhancement systems. Many different systems have been developed and installed in concert halls, public venues, and research facilities worldwide. However, further improvements are required to provide musicians with a desirable acoustic on-stage support equal in quality to the best rooms. An innovative electro-acoustic enhancement system, based on measured high-resolution impulse responses, is developed at the Virtual Acoustics Technology (VAT) lab, in McGill University. The first model is installed in Multi-Media Room (MMR), a large rectangular space (80 ft. $\times$ 60 ft. $\times$ 50 ft.) designed as a film scoring stage. The system uses an array of omni-directional radiators and its signal processing includes 24-channel low-latency convolution at 24 bit/96 KHz, three separate stages of matrix mixing, equalization, and time variation. The objective measurement of the system adjusted to improve on-stage acoustics verifies the enhancement of parameters such as inter-aural cross-correlation coefficient (IACC), early stage support (ST1), and early decay time (EDT) without excessively increasing the overall acoustic energy in the room. The subjective evaluations collected from multiple recording sessions with professional musicians using the system will be presented. [Work supported by NSERC, JMP.]

8:45

3aAA3. Orchestra shell innovation: Vented shell case study. Carl Giegold, Scott Pfeiffer, Dawn Schuette, and Molly Norris (Threshold Acoust., 53 W. Jackson Blvd., Ste. 815 Chicago, IL 60604)

The College Community School District music programs required a venue larger and more conducive to band and choral performance than their 1970s 400-seat theater. The new venue, located in Cedar Rapids, Iowa, seats 1000 in a space designed primarily for music but with theatrical capability as well. An adjustable orchestra shell and variable absorption tailor the room's acoustic response to speech, theater, jazz, symphonic band, choral, and orchestra. Though it must accommodate bands of up to 140 players for audiences of up to a thousand, the hall must also be responsive to small ensembles and intimate to audiences of only a few hundred. To control loudness for the largest bands, the orchestra shell can be vented to th flytower. The design will be presented, as well as measurement data and anecdotal impressions of the shell.

9:05

3aAA4. Venting full-stage portable acoustical shells: The acoustic impact on the stage and audience environments. Ron Freiheit (Wenger Corp., 555 Park Dr., Owatonna, MN 55060, ron.freiheit@wengercorp.com)

The effectiveness of acoustical shells in the stage acoustic environment has long been a topic of discussion. Generally, their impact is positive with increased gain (G) both on stage and to the audience area. However, there has also been speculation that containing too much energy on stage negatively impacts clarity (C80). A study was conducted of an installed portable shell that provided an option to vent energy to the backstage area, thus providing the potential to reduce the overall energy in the stage area. Impulse responses were measured in a number of areas on stage and in the audience area, with vents both open and closed. The results of those metrics will be presented. Binaural recordings were also made at the same locations, providing perceptual information as well.

9:25

3aAA5. Fine-tuning onstage acoustical elements to satisfy... Whom? J. Christopher Jaffe, Jonah Sacks, Benjamin E. Markham, and Robert S. Berens (Acentech, Inc., 33 Moulton St., Cambridge, MA 02138)

Movable onstage features such as over-stage reflector panels, acoustical shell towers, reverberant chamber doors, seating risers, etc. offer a high degree of variability and control over ensemble balance, loudness, and other parameters. Finding the optimal settings for such features is far from trivial and combines many different considerations: quality of sound to the various audience seating sections, balance and intelligibility of the ensemble to the conductor, self and ensemble hearing for the various instrumental sections, and overall loudness onstage, to name a few. On the occasion of welcoming Maestro Christoph Eschenbach as its new Music Director, the National Symphony Orchestra scheduled a special acoustical rehearsal for the purpose of exploring the range of available settings of the adjustable over-stage reflector panels at the Kennedy Center Concert Hall. The presentation will include a detailed description of the rehearsal and the adjustments made to the reflectors, the authors' observations and findings, and various conclusions that have been drawn. Particular attention will be paid to the range of reactions and feedback from various participants, including the Maestro, musicians in different sections on stage, symphony administrators and musicians listening from the audience, and the authors.

9:45

3aAA6. Active acoustics for concert stages. Roger W. Schwenke (Meyer Sound Labs., 2832 San Pablo Ave., Berkeley, CA 94702)

Orchestra shells are often loud places. So much so that some orchestras have instituted noise monitoring and select their repertoire to limit noise exposure. Orchestra shells are loud places because they are surrounded on five out of six sides by nearby hard reflecting surfaces. This also means that the orchestra only experiences reverberation coming from the direction of the house. Active acoustics offers an alternative to physical orchestra shells, which is more light-weight, and offers more flexible acoustical performance including enveloping reverberation for performers.

10:05–10:20 Break

10:20

3aAA7. A survey of selected 800 to 1800-seat multipurpose hall orchestra shells and eyebrows by McKay Conant Hoover Inc. David A. Conant and William Chu (McKay Conant Hoover Inc., 5655 Lindero Canyon Rd., Westlake Village, CA 91362, DConant@MCHinc.com)

New and renovated multipurpose halls by MCH Inc. are examined with focus on the room-acoustical design of sound to and among on-stage performers as well as, necessarily, acoustical considerations associated with direction of that sound to audience. Specifically, orchestra shell/recital screen design as well as integrated eyebrow/canopy design is addressed. Halls discussed include Kavli Theater (Thousand Oaks Civic Arts Plaza), Jackson Hall (Mondavi Center for Performing Arts), Ikeda Theater (Mesa Arts Center), Granada Theatre Restoration (Santa Barbara), Balboa Theatre Restoration (San Diego), Scottsdale Center for Performing Arts and the new Valley Performing Arts Center (Los Angeles).

10:40

3aAA8. The use of multi-channel microphone and loudspeaker arrays to evaluate room acoustics. Samuel W. Clapp, Anne E. Guthrie, Jonas Braasch, and Ning Xiang (GraduateProgram in Architectural Acoust., Rensselaer Polytechnic Inst., Troy, NY 12180, clapps@rpi.edu)

Most room acoustic parameters are calculated with data from omni-directional or figure-of-eight microphones. Using a spherical microphone array to record room impulse responses can yield more information about the spatial characteristics of the sound field, including spatial uniformity and the directions of individual reflections. In this research a spherical array was use to measure room impulse responses in a wide variety of concert halls throughout New York State, both on stage and in the audience, with both the microphone array and a dummy head. The results were analyzed using beamforming techniques to determine spatial information about the sound field, and compared to the results of geometrical acoustics and binaural localization models. Of particular interest was how the spatial data can help to differentiate between different spaces or listener positions that exhibit similar values for conventional metrics. Auralizations were created using both headphone playback and second-order ambisonic playback via a loudspeaker array. These systems were evaluated objectively to compare the reproduction systems with the measured data. Subjects were recruited for listening tests using each reproduction method, and asked to evaluate the halls on both objective measures and subjective preference, and the results of binaural and ambisonic playback were compared.

11:00

3aAA9. Using a spherical microphone array to analyze concert stage acoustics. Terence Caulkins (Arup, 155 Ave. of the Americas, New York, NY 10013, terence.caulkins@arup.com), Anne Guthrie (Arup Rensselaer Polytechnic Inst., NY 12180), Sam Clapp, Jonas Braasch, and Ning Xiang (Rensselaer Polytechnic Inst., NY 12180)

A 16 channel spherical microphone array has been designed and constructed to measure high order Ambisonic sound fields, based on equations for scattering off of a rigid sphere. This microphone has been used to capture and characterize the acoustics of nine different concert hall stages around the state of New York. Musicians have been invited to perform in real-time in auralizations of each hall and participate in a subjective preference test based on their experience. The preference tests have been analyzed using multi-dimensional scaling methods and compared with acoustical data derived from a beamforming analysis of each stage. Additional comparisons with A.C. Gade's omnidirectional parameters (stage support, early ensemble level) and geometric parameters described by J. Dammerud are performed.

11:20

3aAA10. Investigations of stage acoustics at the Sydney Opera House Concert Hall. Timothy E. Gulsrud (Kirkegaard Assoc., 954 Pearl St., Boulder, CO, tgulsrud@kirkegaard.com)

Improvements to onstage hearing conditions were made at the Sydney Opera House Concert Hall during a series of acoustics trials conducted during September 2009. During the acoustic trials, physical changes were made to reflective surfaces around the platform and an active acoustics system was demonstrated. Taken together, the temporary changes had a positive influence on hearing conditions during both rehearsals and performances by the Sydney Symphony Orchestra and Sydney Philharmonia. This paper reviews the strategies used to implement the improvements to stage acoustics, and discusses the unique techniques used to evaluate the stage acoustics both subjectively and objectively.

11:40

3aAA11. Acoustical renovation of Hahn Recital Hall at the Music Academy of the West, Montecito, CA. William Chu and David A. Conant (McKay Conant Hoover Inc., 5655 Lindero Canyon Rd., Westlake Village, CA 91362, WChu@MCHinc.com)

A 350-seat fixed-acoustics recital hall believed designed by Vern Knudsen in 1971 was substantially renovated in 2008 to meet the current needs of the revered Music Academy of the West. The preponderance of its program was to serve Marilyn Horne's operatic program as well as chamber music but required as well to provide suitable space for occasional full orchestra rehearsals and performance. The renovation design is described as it developed from initial acoustical measurements to the novel platform acoustics and other adjustments that permit optimizing the large platform for multiple needs. Specifics of the acoustical coupling characteristics of the custom, sound transparent/sound reflective and diffusive recital screen, its overhead reflectors and absorptive drapery are discussed as well as accommodation for Met Live presentations.

WEDNESDAY MORNING, 2 NOVEMBER 2011 PACIFIC SALON 1, 8:00 A.M. TO 12:00 NOON

Session 3aAB

Animal Bioacoustics: Acoustics for Saving Endangered Species I

Jay Barlow, Cochair
NOAA Southwest Fisheries Science Center, 3333 N. Torrey Pines Ct., La Jolla, CA 92037

Sofie M. Van Parijs, Cochair
Northeast Fisheries Science Ctr., 166 Water St., Woods Hole, MA 02543-1026

3a WED. AM

Chair's Introduction—8:00

Invited Papers

8:05

3aAB1. Automatically identifying rare sounds of interest in environments cluttered with biological homophones. Kurt M. Fristrup (Natural Sound and Night Sky Div., Natl. Park Service, 1201 Oakridge Dr., Ste. 100, Fort Collins, CO 80525, kurt_fristrup@nps.gov)

Successful acoustical monitoring for threatened or endangered species must surmount challenges of adequate spatial and temporal coverage in the data collection phase, and efficient and effective processing in the data analysis phase. For terrestrial environments, a diverse array of digital recording options has relaxed the difficulty of obtaining sufficient coverage. However, this capability has amplified the requirement for efficient processing. In many terrestrial environments, the principal processing challenge is distinguishing the sounds of rare species from many other sounds that are similar in time-frequency structure. Some of these biological homophones are generated by species that are much more numerous than the target. Solutions to these problems will be assessed in the context of large scale projects that focused on bird species of special interest.

8:25

3aAB2. Monitoring a small marine cetacean population using passive acoustic techniques: Vaquita, a critically endangered species. Armando Jaramillo-Legorreta and Lorenzo Rojas-Bracho (Instituto Nacional de Ecologia, DGIOECE–Marine Mammals Dept., CICESE, Ensenada, B.C., Mexico)

In 2008, vaquita population was estimated in only 245 individuals. Between 1997 and 2007, a passive acoustic semi-autonomous system was used for monitoring detection rate of this species. The analysis resulted in decline of approximately 58%. Sighting and acoustic data from a 2008 research cruise, compared to the 1997 estimate of abundance, resulted in a decline of approximately 57%. Hence, passive acoustic detection probed reliable to monitor the population. The Mexican Government implemented a recovery plan, which includes monitoring population trends. At current level, the population can grow at maximum rates lower than 4% annually. The severely reduced population level, and scarcity of acoustic detections, made unreliable to continue the use of the methods applied till 2007. An increase of sampling effort was identified as the key to implement a reliable system, only achievable using completely autonomous detectors. Field test of autonomous detectors identified C-POD as very reliable. Using data collected with this equipment it was determined that an annual effort of 5000 C-POD days is needed to detect population increase. A sampling grid of 62 stations was designed. Work to design and test reliable mooring devices was done. Currently, the first of five years of sampling has started.

8:45

3aAB3. Statistical design for acoustic monitoring to detect declines of endangered species and prevent extinction. Jay Barlow (NOAA Southwest Fisheries Sci. Ctr., 3333 N. Torrey Pines Ct., La Jolla, CA 92037), Armando Jaramillo-Legorreta (Instituto Nacional de Ecologia, c/o CICESE, Km. 107 Carretera Ensenada-Tijuana, Ensenada, BC 22860, Mexico), Barbara L. Taylor (NOAA Southwest Fisheries Sci. Ctr., La Jolla, CA 92037), Lorenzo Rojas-Bracho (Instituto Nacional de Ecologia, *c/o CICESE*, Km. 107 Carretera Ensenada-Tijuana, Ensenada, BC 22860, Mexico), and Nicholas J. C. Tregenza (Beach Cottage Ltd., Long Rock, Cornwall, TR20 8JE, United Kingdom)

As populations become smaller and more endangered, the ability to monitor trends in their abundance also decreases. For vocal species, passive acoustic monitoring can provide a powerful, cost-effective method of monitoring relative abundance. However, the monitoring effort should be based on a statistical design that can detect population declines in time to prevent extinction. This is illustrated with an example of the vaquita (Phocoena sinus) an endangered porpoise in the northern Gulf of California, Mexico. A pilot project showed that porpoise detectors (C-PODs) recorded echo-location clicks from approximately one group of porpoises every two days. Based on this, we estimate that 5000 days of C-POD monitoring per year would be needed to obtain a measure of relative vaquita abundance with a coefficient of variation (CV) of 3%. A power analysis shows that five years of monitoring with this CV would give a high probability of detecting a 5% annual increase or decrease in population size. Visual sighting methods could not detect such small changes with any conceivable level of survey effort. This kind of innovative monitoring is a critical tool in the continuing evaluation of conservation measures.

Contributed Papers

9:05

3aAB4. Towed hydrophone surveys for monitoring trends in the abundance and distribution of the critically endangered Gulf of California porpoise, Vaquita. Rene Swift (Sea Mammal Res. Unit, Scottish Oceans Inst., Univ. of St. Andrews, St. Andrews, Fife, KY16 8LB, United Kingdom), Shannon Rankin, Tim Gerrodette, Barbara Taylor (Southwest Fisheries Sci. Ctr., La Jolla, CA 92037), Douglas Gillespie, Jonathan Gordon (Univ. of St. Andrews, St. Andrews, Fife KY16 8LB, United Kingdom), Lorenzo Rojas-Bracho, and Armando Jaramillo-Legorreta (Inst. Nacional de Ecologia/CICESE, Ensenada, B.C. 22860, Mexico)

The vaquita (*Phocoena sinus*) is a critically endangered small cetacean found only in the upper Gulf of California, where fisheries bycatch remains an acute threat. Cost, shallow heavily fished areas, and the vaquitas extreme avoidance of noisy motorized vessels argue against using large vessels typically used for visual line transect surveys. Towed hydrophone surveys, using *Rainbow-Click* semi-automatic detection software, were carried out from a 24' sailing trimaran in autumn 2008. Ultrasonic ($\sim$130 kHz) vaquita echolocation clicks were reliably detected and tracked using classification parameters developed for harbor porpoise. Transects were sailed on 49% of days and 31 groups were detected within the vaquitas known range and in areas not easily surveyed using traditional methods. Although very high levels of ambient noise presented challenges for acoustic monitoring, perpendicular distances were calculated to 30 groups giving an estimated strip half-width of 198 m. The detection algorithm has since been implemented in PAMGUARD software and significantly improved using survey data. Shallow, heavily fished areas remain difficult for estimating and monitoring trends in abundance. Towed arrays proved effective for the former and may remain the only alternative for the latter. Precision is likely to remain low for the quick detection of small rates of increase.

9:20

3aAB5. Acoustic detection of the Ivory-billed Woodpecker (*Campephilus principalis*). Michael D. Collins (P.O. Box 1975, Pearl River, LA 70452, mike@fishcrow.com)

There have recently been independent reports of multiple sightings and auditory detections of the Ivory-billed Woodpecker (*Campephilus principalis*) in remote swamps in Arkansas, Florida, and Louisiana. Putative audio and video recordings have been obtained at each site, but no clear image of this critically endangered (and extremely elusive) species has been obtained in decades. Acoustic detection is possible by "kent" calls, alarm calls, double knocks, and pounding sounds associated with foraging and cavity construction, but each of these clues is problematic. Kents are difficult to hear in the distance—the author did not detect a long series of kents before drifting in a kayak to well within 100 m of the bird. This species does not drum like other woodpeckers, and it is believed to forage relatively quiet (at least part of the time) by using its massive bill to pry bark loose. The Ivory-billed Woodpecker is not one of those species that is easier heard than seen—during 5 yr of fieldwork, the author had ten definite sightings but only two definite auditory detections. These issues and the current status of the fieldwork in the Pearl River basin in Louisiana will be discussed.

9:35

3aAB6. Acoustic risk factors and passive acoustic monitoring of endangered southern resident killer whales (*Orcinus orca*). Marla M. Holt, Candice K. Emmons, M. Bradley Hanson, and Dawn P. Noren (Marine Mammal Ecology Team, NOAA Northwest Fisheries Sci. Ctr., 2725 Montlake Blvd. East, Seattle, WA 98112)

Southern resident killer whales (SRKWs) occur along the coastal and inland waters of the northeast Pacific Ocean. They are currently listed as endangered in both the U.S. and Canada. Risk factors that are potentially affecting population recovery include prey availability and quality and acoustic disturbance. This is because SRKWs specialize on Chinook salmon, of which many stocks are depleted, and there is a well developed whale-watching industry that focuses on viewing SRKWs in their core summer habitat. This review paper will highlight acoustic research conducted to address risk factors, recovery goals, and other conservation considerations of this endangered population. These include investigations on acoustic and behavioral responses to anthropogenic sounds. A previous study demonstrated amplitude compensation as vessel noise increased and an ongoing study is investigating potential acoustic effects on behavior, including foraging, using suction cup attached digital acoustic recorders (DTAGs). This paper will also discuss passive acoustic monitoring efforts involving coastal ship surveys and acoustic recorder deployments which aim to better characterize critical habitat, particularly in areas and during seasons when SRKW occurrence is less well defined. All of these investigations are designed to provide critical data necessary to address and refine recovery goals and management actions for SRKWs.

9:50

3aAB7. Passive acoustic recording to build acoustic catalogs to remotely monitor resident individuals and the health of the dolphin population in the Indian River Lagoon system in Florida. Edmund Gerstein (Dept. of Psych., Charles E. Schmidt College of Sci. Florida Atlantic Univ., 777 Glades Rd., Boca Raton, FL 33431, gerstein2@aol.com), Beth Brady (Nova Southeastern Univ., Dania Beach, FL 33004), Rebecca Weeks (Florida Atlantic Univ., Dania Beach, FL 33004), Gregory Bossart (Georgia Aquarium, Atlanta, GA 30313), Juli Goldstein, and Stephen McCulloch (Florida Atlantic Univ., Ft. Pierce, FL 34946)

Harbor Branch Oceanographic Institute at Florida Atlantic University together with, NOAA's National Ocean Service/Center for Coastal Environmental Health & Biomolecular Research initiated a research program designed to assess environmental and anthropogenic stressors that may affect the health and long-term viability of bottlenose dolphin populations inhabiting coastal regions of Florida and South Carolina This collaborative program is known as the Health and Environmental Risk Assessment Project. The project involves the capture, sampling, and release of selected wild dolphin stocks to allow comprehensive health screenings by collecting and analyzing a variety of biomedical samples and associated data. During Indian River Lagoon (IRL) dolphin population assessment, the acoustic behavior of 33 individual dolphins was recorded using synchronized DARP buoys configured with directional and omnidirectional hydrophones. Vocalizations were recorded during the capture, holding and health sampling phases. Acoustic levels

and behavior during ABR hearing measurements were also monitored. The whistle contours of 23 individuals have been identified and will be added to the photo-identification and genetic catalog maintained for the IRL population. These signature whistles will be used to train a network of remote acoustic sensors to monitor the distribution, social interactions, and habitat utilization of cataloged dolphins in the IRL. Authorized NMFS permit 14352-01.

10:05–10:25 Break

Invited Paper

10:25

3aAB8. Passive acoustic methods for surveying visually cryptic frog species. David L. Borchers (Dept. Maths & Stats., Univ. of St. Andrews, CREEM, The Observatory, Buchannan Gdns, St. Andrews, KY16 9LZ, Scotland, dlb@st-andrews.ac.uk), John G. Measey, and Res Altwegg (South African Natl. Biodiversity Inst., Cape Town, South Africa)

Passive acoustic survey methods have great potential for assessing cryptic species that are vocally active. The tiny Moss Frogs (genus *Arthroleptella*) that inhabit seepages on remote mountain tops in the Western Cape of South Africa are a case in point. Many species are restricted to individual mountains and most are on the IUCN Red List. Surveys are prohibitively expensive because each individual capture involves three to four person-hours of searching. However, males can be heard calling throughout the winter. Using a portable recorder, we gathered acoustic data from which abundance can be estimated and populations monitored. This involves estimation of both the spatial "footprint" of the acoustic array (i.e., what area it effectively surveys) and animal vocalization rates. Ideally both should be estimated as an integral part of the survey, rather than obtained from outside the survey. By combining recently developed mark-recapture methods that take explicit account of the spatial location of microphones, with data on acoustic signal strength and/or time-difference-of-arrival at different microphones, it is possible to do this. We describe the survey and estimation methods, which have many other potential applications. In the case of these frogs, acoustic survey is hundreds of times more efficient than methods involving physical capture.

Contributed Papers

10:45

3aAB9. Interannual temporal and spatial distribution of bowhead whales in the Western Alaskan Beaufort Sea; 2007–2010. Stephanie L. Grassia, Catherine L. Berchok (NOAA/Natl. Marine Mammal Lab., 7600 Sand Point Way NE, Seattle, WA 98115), and Dana L. Wright (School of Marine Sci., Univ. of Maine, 5706 Aubert Hall, Orono, ME 04469)

Passive acoustic monitoring began offshore of Barrow, Alaska, in 2007 as part of the interdisciplinary BOWFEST (Bowhead Feeding Ecology Study) project. This study area spans from the northern coast of Alaska out to 72°N and between 152° and 157°W. To better understand how bowhead whales use this area, long-term (year-long) and shorter-term (week- to month-long) autonomous passive acoustic recorders were deployed on subsurface moorings. The long-term recorders were deployed at four locations along the 100 m isobath off Barrow, running on a 30%–45% duty cycle at a sampling rate of 8192 Hz. The shorter-term recorders were deployed inshore (<20 m water depth) and sampled at either 12.5 or 40 kHz on a 80%–90% duty cycle (although one sampled continuously at 8192 Hz). Since 2007 we have obtained 18 820 h of recordings from the long-term and 3780 h of recordings from the short-term moorings. All recordings were reviewed manually, with the long-term data analyzed on a 3-h time interval and short-term data analyzed fully. The majority of the bowhead call detections were between April and November. Very few detections were made during the winter months. Comparisons between the offshore recorders and inshore recorders (which were within the main migration corridor) will be discussed.

11:00

3aAB10. Acoustics as a tool in sub-species and population identification for endangered fin whales, *Balaenoptera physalus*. Shannon Rankin, Jay Barlow, Eric Archer (Southwest Fisheries Sci. Ctr., 3333 N. Torrey Pines Ct., La Jolla, CA 92037, shannon.rankin@noaa.gov), and Benjamin Jones (Tulane Univ., New Orleans LA 70118)

Identification of “stocks” (sub-species and independent populations) is important for understanding and mitigating potential sources of human-caused mortality. This is especially critical for endangered and protected species, such as the large whales. Stock identification for whales has typically been based on ecology, life history, morphology, and genetics. However, for many species, acoustic differences in whale call types may indicate population or sub-species structure. The potential role of acoustics in identifying species and sub-species has been identified in numerous publications; however, this role has yet to be realized for large whales. In an effort to include acoustic data in this process, we are contributing to current efforts to update the status of endangered fin whales, *Balaenoptera physalus*, in the North Pacific. An analysis of North Pacific fin whale populations based on identification of “song” provides hypotheses that can be tested with genetics. Strengths and limitations of acoustic methods will be presented, as will the potential for collaboration on the scale of ocean basins.

11:15

3aAB11. New call type produced by the North Pacific right whale, *Eubalaena japonica*. Jessica L. Crance, Catherine L. Berchok, and Phillip J. Clapham (NOAA/AFSC, Natl. Marine Mammal Lab., 7600 Sand Point Way NE, Seattle, WA 98115)

The North Pacific right whale, (NPRW), is one of the most endangered baleen whales in the world and has been the focus of intensive population monitoring studies. In 2010 during transit through the Bering Sea right whale critical habitat (BSCH), near-24-h acoustic monitoring was conducted using DiFAR-capable sonobuoys. A new call pattern was detected during a focal follow of a NPRW, consisting of a series of pulses with a fundamental frequency at 110 Hz and peak energy at 640 Hz, ending in a 250–150 Hz downsweep. This call pattern was repeated multiple times and detected in the presence of gunshot and upsweep calls. Directional bearings to the call were a perfect match to those of the gunshot call. Although humpback whales were present, bearings to the humpback vocalizations were in the opposite direction (200 deg difference). The only other species detected visually or acoustically in the area were fin whales. This same pattern was detected in October and November on a 2009 long-term recorder in the BSCH, also in the presence of gunshot and upsweep calls. Because directional information from the sonobuoys during the focal follow exactly matched those from the gunshot call, we attribute this call pattern to the NPRW.

11:30

3aAB12. Passive acoustic monitoring of North Atlantic right whales in Cape Cod Bay. Ying-Tsong Lin, Arthur E. Newhall (Appl. Ocean Phys. and Eng. Dept., Woods Hole Oceanogr. Inst., Woods Hole, MA 02543, ytlin@whoi.edu), and Mark F. Baumgartner (Woods Hole Oceanogr. Inst., Woods Hole, MA 02543)

Passive acoustic techniques have been applied extensively to marine mammal monitoring, localization, and tracking. An acoustic fieldwork using hydrophone arrays and free-floating buoys was conducted in Cape Cod Bay in the spring of 2011 for monitoring North Atlantic right whales. Three vertical hydrophone arrays were deployed to form a large triangular network with approximately 12 km on each side, and one horizontal array was mounted on the seafloor at one station. The passive hydrophone arrays operated for 25 days and recorded a vast amount of vocalizations made by humpback, fin, sei, minke, and, most importantly, right whales. These data can be used for localizing these endangered animals and tracking their movements in the bay with a large spatial coverage. Acoustic buoys, the real-time acoustic tracking system, were deployed on two one-day cruises for several hours at a time to

triangulate calling whales over small spatial scales, and can be used to ground truth long-range whale localizations derived from the array network. Results from different passive acoustic techniques will be presented and compared. [Work supported by the Ocean Life Institute and the Marine Mammal Center of the Woods Hole Oceanographic Institution.]

11:45

3aAB13. Passive acoustic and visual monitoring of humpback whales (*Megaptera novaeangliae*) in the Olympic Coast National Marine Sanctuary: Importance of quantifying call type. Amanda J. Cummins (Scripps Inst. of Oceanogr., 9500 Gilman Dr., La Jolla, CA 92093-0205), Erin Oleson (NOAA Fisheries, Honolulu, HI 96814), John Calambokidis, Greg Schorr, Erin Falcone (Cascadia Res. Collective, Olympia, WA 98501), Sean Wiggins, and John A. Hildebrand (Scripps Inst. of Oceanogr., La Jolla, CA 92093-0205)

Humpback whales (*Megaptera novaeangliae*) produce a variety of vocalizations such as social and feeding calls as well as patterned calls that comprise song. Typically, social and feeding vocalizations do not follow the highly structural format of song. High-frequency acoustic recording packages were deployed in the Olympic Coast National Marine Sanctuary July 2004 through June 2009 while visual surveys were conducted throughout the year at approximately monthly intervals. Humpback whales were detected visually and acoustically; however, there was a mismatch in the peak seasonality of these detections. Visual detections occurred in all seasons but peaked in summer and early fall. Acoustic detections were documented primarily in late summer to early winter. Male humpback whales are known to produce long songs primarily during their winter breeding season. To test whether the detection differences between visual and acoustic surveys could be explained by changes in the whales' vocal behavior, we quantified the relative occurrence of song and non-song calling using a variety of metrics and related the occurrence to the visual survey sightings. We show how identification of the type of acoustic detection is an important consideration and can help address biases introduced by seasonal differences in the production rate of reproduction-related calls.

WEDNESDAY MORNING, 2 NOVEMBER 2011 — ROYAL PALM 5/6, 7:30 TO 11:55 A.M.

Session 3aBA

Biomedical Acoustics and Physical Acoustics: Biomedical Applications of Acoustic Radiation Force

Mostafa Fatemi, Chair
Physiology and Biophysics, Mayo Clinic, 200 First St., SW, Rochester, MN 55905

Chair's Introduction—7:30

Invited Papers

7:35

3aBA1. Production of shear waves with novel radiation force beams. James Greenleaf, Shigao Chen, and Matt Urban (Mayo Clinic, Dept. of Physiol. and Bioengineering, 200 First St., S.W., Rochester, MN 55905)

Propagating shear waves in tissue can be measured with high frame rate Doppler or correlation methods. The measured characteristics of the shear waves, such as speed versus frequency, can be used to deduce material properties such as complex viscoelastic modulus using physics models appropriate to the geometry and properties of the tissue. This inverse problem is characterized by calculating the storage and loss modulus as a function of frequency and requires appropriate tissue motion, which in turn requires optimized dynamic radiation force distributions. We will discuss novel radiation force distributions that provide enhanced tissue motions appropriate to dealing with the inverse problem of determining tissue material properties from ultrasonically measured tissue motion.

7:55

3aBA2. Acoustic radiation force for rapid detection of particles in biological liquids. Lev Ostrovsky (Zel Technologies and Univ. of Colorado, 325 Broadway, Boulder, CO 80305), Aba Priev, and Yechezkel Barenholz (Hebrew Univ. of Hadassah Med. School, Israel)

As known, ultrasonic standing waves can be used to concentrate particles and biological cells into separated bands. Acoustical separation based on plane standing waves is limited to particles of few microns and larger. This presentation concerns using acoustic radiation force (ARF) produced by cylindrical standing waves for detection of high-density submicron-size particles (bacterial cells) in pressure nodes and low-density particles (fat globules) in antinodes. Theoretical calculations show that in a cylindrical ultrasonic chamber, ARF near the central node can exceed the force at the chamber periphery by about 20 times. In a cylindrical standing wave, ARF may induce movement of bacteria with a speed of the order of a few millimeter per second at a frequency of 2 MHz and pressure amplitude of 100 kPa, whereas the speed of bacteria in plane standing wave does not exceed 0.2 mm/s under the same conditions. The cylindrical standing wave system performance was tested for the *E. coli* bacteria in water and for a multi-component system containing fat globules and somatic cells in milk. Dilute suspensions of bacteria or fat globules were concentrated by at least 2 orders of magnitude.

8:15

3aBA3. Parameterization of the scattering and radiation forces and torques on spheres in terms of complex partial wave s-functions: Applications and interpretation. Philip L. Marston and Likun Zhang (Phys. and Astronomy Dept., Washington State Univ., Pullman, WA 99164-2814, marston@wsu.edu)

A wide range of physical responses of spheres in acoustic beams, including radiation forces and torques, are closely related to properties of the far field acoustic scattering. See for example the short reviews in [Marston, J. Acoust. Soc. Am. **125**, 3539–3547 (2009)] for radiation forces and [Zhang and Marston, J. Acoust. Soc. Am. **129**, 1679–1680 (2011)] for torques. When evaluating these properties it can be advantageous to retain the notation of resonant scattering theory from prior work for plane-wave illumination. That notation involves the complex partial wave, s-function such that the nth partial wave, the amplitude, is proportional to (s-1) and s is unimodular in the absence of absorption. Some applications of this parameterization to the interpretation and understanding of the scattering by, and radiation forces and torques on, spheres in acoustic beams will be examined. These include examples of forces and torques associated with helicoidal beams and simple methods for modulating the torque. This formulation of the scattering should be helpful for cross-disciplinary applications. Some potentially confusing points from the literature on acoustic plane-wave scattering from the 1970s and early 1980s will be clarified. [Work supported by ONR and by NASA.]

8:35

3aBA4. Vibro-acoustic doppler. Alireza Nabavizadeh, Matthew W. Urban, and Mostafa Fatemi (Dept. of Physiol. and Biomed. Eng., Mayo Clinic College of Medicine, 200 First St., SW, Rochester, MN 55905)

This paper describes the principles and initial experimental results of a new Doppler method called vibro-acoustic Doppler (VAD). VAD uses acoustic response of a moving object to a highly localized dynamic radiation force of ultrasound to measure the velocity of the object. The low frequency radiation force is exerted by two co-focused ultrasound beams with slightly different frequencies. The acoustic response of the object is detected by a hydrophone. A formula that describes the relation between the Doppler frequency shift of the emitted acoustic field and the velocity of the moving object is reported. To verify the theory, experiments are conducted on a moving string and a fluid flow phantom. Results show that the error in velocity measurement is less than 9.1% for either phantom. An advantage of this method over the traditional ultrasound Doppler is that velocity measurement with VAD is almost independent of the angle between the ultrasound beam and motion direction. It is shows that in the worst case, the error is 10.3% for a ± 30 deg angle variation. Potential biomedical applications of VAD will be discussed.

Contributed Papers

8:55

3aBA5. Young's modulus estimation of bovine lens *ex-vivo* using a laser-induced microbubble under impulsive acoustic radiation force. Sangpil Yoon (Dept. of Mech. Eng., Univ. of Texas at Austin, 204 E. Dean Keeton St., Austin, TX 78712, s.yoon@mail.utexas.edu), Salavat Aglyamov, Andrei Karpiouk, and Stanislav Emelianov (Univ. of Texas at Austin, Austin, TX 78712)

According to the most widely accepted theory of presbyopia, the age-related loss of accommodation is attributed to Young's modulus changes in the lens. Previously, we have developed an approach to measure the mechanical properties of viscoelastic medium using microbubbles and acoustic radiation force. In this study, we tested this technique to assess the mechanical properties of bovine lenses ex-vivo. An impulsive acoustic radiation force was applied to laser-induced microbubbles created with nanosecond laser pulses at different locations in the lens. An acoustic radiation force with typical duration less than 150 μs was generated by a 3.5 MHz transducer. For accurate spatio-temporal measurements of the microbubble's displacement, a custom-made ultrasound system consisting of two 25 MHz transducers was built. The first transducer emitted a train of pulses and the other transducer received the train of echoes reflected from the microbubble. The developed system was operating at 400 kHz pulse repetition frequency. The results show good agreement between experimental measurements and the theoretical model of microbubble dynamics. Evaluation of the spatial distribution of elasticity demonstrates that the Young's modulus of the nucleus is higher than that of the cortex for bovine eye lenses of mature animals. [Work supported by NIH Grant EY018081.]

9:10

3aBA6. Measurement of elasticity of thin elastic layers with radiation force and wave propagation methods. Matthew W. Urban, Ivan Z. Nenadic, Miguel Bernal, Bo Qiang, James F. Greenleaf, and Shigao Chen (Dept. of Biomed. Eng., Mayo Clinic College of Medicine, 200 First St. SW, Rochester, MN 55905, urban.matthew@mayo.edu)

In the field of tissue engineering, monitoring the elasticity of developing tissue cultures is important for evaluating their growth and maturation. In this work, we used wave propagation methods to measure the elasticity of thin tissue-mimicking layers to test the feasibility of measuring the elasticity of developing tissue cultures. We developed finite element and analytical models for investigating this problem and found good agreement between the results from both models. We used ultrasound radiation force to induce propagating waves, and the waves are measured using high frame rate ultrasound imaging at 10 kHz. The wave modes were identified and compared with an analytic model for Rayleigh wave propagation in a thin elastic layer bound to a solid substrate. The same gelatin mixture was used to make phantoms 1, 4, and 50 mm thick, where the phantom with 50 mm thickness was used as control and evaluated with shear wave imaging methods. The analytic model was used to fit data from experiments in the large block and thin layers of 1 and 4 mm thick, and the measured shear moduli were 22.7, 21.8, and 22.5 kPa, respectively. These results provide a validation for measurement of elasticity in thin elastic layers.

9:25

3aBA7. Modeling of the radiation force imparted on a spherical kidney stone by a focused beam of an ultrasound array. Oleg Sapozhnikov and Michael Bailey (Appl. Phys. Lab., Ctr. for Industrial and Medical Ultrasound, Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105)

Radiation force imparted on an elastic scatterer by a narrow ultrasound beam is different from that imparted by a plane wave, especially when the beam waist is comparable or smaller than the scatterer's diameter. Such a situation exists when a kidney stone is pushed by an acoustic wave emitted by a megahertz frequency ultrasound array. A spherical stone of several millimeters in diameter with elastic properties similar to calcium oxalate monohydrate kidney stones is considered. An acoustic wave is taken in a form of a continuous-wave 2.5 MHz focused ultrasound beam emitted by Philips HDI C4-2 imaging array. To study radiation force, the transducer field is created as a sum of plane waves of various inclinations, and scattering of each plane waves is modeled based on a known solution. Numerical calculations show that within some frequency range both the backscattering from and the radiation force on a kidney stone exceeds the values for absolutely soft or rigid spheres of the same diameter. A vector component of the radiation force can be created in a direction other than the ultrasound propagation by targeting off the stone center. [Work supported by NIH (DK43881, DK092197, and DK086371), RFBR, and NSBRI through NASA NCC 9-58.]

9:40

3aBA8. Acoustic radiation force applied to the treatment of lower pole kidney stones. Michael R. Bailey, Bryan W. Cunitz, Barbrina Dunmire, Yak-Nam Wang, Wei Lu, Julianna C. Simon, Peter J. Kaczkowski, and Oleg A. Sapozhnikov (Appl. Phys. Lab., Ctr. for Industrial and Medical Ultrasound, Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105)

Lower pole kidney stones have lower rates of clearance after shock wave lithotripsy or ureteroscopy compared to other stone locations. Residual stone fragments, post treatment, often lead to additional morbidity and secondary surgery. We describe the use of acoustic radiation force created by transcutaneous focused ultrasound to manipulate the location of stone fragments within the collecting system in order to facilitate their passage. Artificial and human stones were placed in the lower pole of kidneys of a porcine animal model. An open architecture, software based diagnostic ultrasound system and scanhead were modified to produce roughly 100 μs pulses of up to 16 MPa peak positive pressure in water. Stone motion was observed in real-time with simultaneous imaging through the same scanhead and with fluoroscopy. All stones were seen to move. Stone velocities were on the order of 1 cm/s. Stone displacement distance was up to 3 cm, and operators could generally control the direction of stone movement. No evidence of thermal necrosis or mechanical damage of renal tissue was observed. Thus acoustic radiation force can be used to facilitate lower pole stone fragment clearance. [Work supported by NIH DK43881, DK086371, DK092197, and NSBRI through NASA NCC 9-58.]

9:55

3aBA9. Determination of thresholds for renal injury in a porcine model by focused ultrasound. Julianna C. Simon, Yak-Nam Wang (Appl. Phys. Lab., Ctr. for Industrial and Medical Ultrasound, Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, jcsimon@uw.edu), Andrew P. Evan (Indiana Univ. School of Medicine, Indianapolis, IN 46202), Marla Paun, Frank L. Starr, Lawrence A. Crum, and Michael R. Bailey (Univ. of Washington, Seattle, WA 98105)

Recently, a system that uses focused ultrasound to expel renal stone fragments from the kidney by radiation force was developed by Shah *et al.* [Urol. Res. **38**, 491–495 (2010)]. A worst-case treatment protocol using this system would require a total exposure time of 10 min, with a spatial peak pulse average intensity (I_SPPA) of 3600 W/cm^2 in water (16 MPa peak positive pressure) and a 3% duty cycle. As the system operates above the FDA limits for diagnostic ultrasound, our goal is to verify the safety of the system by determining the threshold for renal injury. A 2-MHz annular array generating I_SPPA up to 29 000 W/cm^2 in water was placed on the surface of *in vivo* porcine kidneys and focused in the proximal parenchyma. Exposures of 10 min duration with varying I_SPPAs and duty cycles were repeated at least 6 times. Mechanical tissue damage and cell viability were evaluated histologically using H&E, PAS, and NADH diaphorase stains. The proportion of samples showing injury was plotted versus duty cycle and I_SPPA. The results indicate that the system to expel renal stones operates below the threshold for kidney injury. [Work supported by NIH DK043881, DK086371, DK092197, and NSBRI through NASA NCC 9–58.]

10:10–10:25 Break

10:25

3aBA10. On the numerical computation of the acoustic radiation force generated by a modulated sound field. Egor Dontsov, Bojan Guzina (Dept. of Civil Eng., Univ. Of Minnesota, 500 Pillsbury Dr. SE, Minneapolis, MN 55455, donts002@umn.edu), Shigao Chen, and Mostafa Fatemi (Mayo Clinic College of Medicine, MN 55905)

The acoustic radiation force (ARF), which signifies the average transfer of momentum from the sound wave to the propagating medium, is nowadays frequently used to compute the mean displacement field in soft tissues due to the action of a focused ultrasound beam. In situations when the sound field used to generate the ARF is modulated, the latter depends on both spatial and temporal derivatives of the acoustic intensity. Unfortunately, the numerical computation of the nonlinear acoustic solution due to modulated focused ultrasound beam is complicated by the presence of two disparate time scales in the formulation. To deal with the problem, the KZK-type spatial scaling is complemented by its dual-time-scale companion, where the temporal coordinate is split into its "fast" and "slow" components, allowing one to track the ultrasound-scale oscillations and modulation-scale variations separately in the solution. In this case, the nonlinear acoustic solution (and consequently the ARF) can be effectively computed in the "fast" frequency domain and "slow" time domain for any given modulation envelope, transient, or steady-state. The proposed developments are both validated by the experimental results and illustrated via numerical examples that show the effectiveness of the computational scheme.

10:40

3aBA11. The role of constitutive nonlinearities and sound modulation in the generation of the acoustic radiation force. Bojan Guzina and Egor Dontsov (Dept. of Civil Eng., Univ. of Minnesota, 500 Pillsbury Dr. SE, Minneapolis, MN 55455, guzina@wave.ce.umn.edu)

This study investigates the acoustic radiation force (ARF) in soft tissues when the generating ultrasound field is modulated with a low-frequency envelope (10^2-10^4 Hz range). On approximating the soft tissue as that of a nonlinear elastic material with heat conduction and viscosity, the system of nonlinear balance equations, governing both the ultrasound-scale oscillatory motion and the ARF-induced mean motion, is formulated explicitly. To deal with the effects of ultrasound modulation, a dual-time-scale approach featuring the "fast" (ultrasound-scale) and "slow" (modulation-scale) temporal coordinates is deployed. In this setting the governing equations for the mean motion, featuring the ARF as the body force term, are extracted by taking the "fast" time average of the nonlinear balance laws. The ARF is shown to consist of two distinct terms, namely (i) the potential term, which is proportional to the gradient of the ultrasound intensity and (ii) the axial term, which contains both an attenuation-driven component and a modulation-driven component. A salient feature of the new solution for the ARF is that its entries feature specific combinations of the elastic nonlinearity coefficients, which may vary depending on tissue type. For completeness, the proposed formula is illustrated by numerical simulations and compared to the existing expressions.

10:55

3aBA12. Modeling of shear waves generated in a soft solid by a piston. Kyle S. Spratt, Yurii A. Ilinskii, Evgenia A. Zabolotskaya, and Mark F. Hamilton (Appl. Res. Labs., Univ. of Texas at Austin, Austin, TX 78713-8029)

This work is motivated by the transient elastography experiments described by Catheline *et al.* [J. Am. Stat. Assoc. **105**, 2941–2950 (1999)], the purpose of which was to measure the phase speed of shear waves propagating through tissue phantoms. In that work, a small circular piston was used to generate the shear waves, the motion of which was perpendicular to the bounding surface of the sample. The current work is a theoretical investigation of this type of source condition based on the assumption of a linear elastic medium and using an angular spectrum approach to solve for the entire elastic field, both compressional and shear waves, that results from a given velocity distribution at the source plane. Special attention is paid to the velocity field near the source, in particular how the near incompressibility of the tissue-like medium is conducive to the generation of shear waves from such a compressive piston source. For high-frequency excitation, in which the resulting shear wave disturbances are beam-like, the validity of using a parabolic approximation to describe diffraction of the transverse motion of the field in the paraxial region is investigated. [Work supported by the ARL:UT McKinney Fellowship in Acoustics.]

11:10

3aBA13. Two-dimensional shear elasticity imaging using comb-push acoustic radiation force and algebraic direct inversion of the motion differential equation. Pengfei Song, Armando Manduca, Zhoubo Li, Heng Zhao, Matthew W. Urban, James F. Greenleaf, and Shigao Chen (Dept. of Physio. and Biomed. Eng., Mayo Clinic College of Medicine, 200 First St., SW, Rochester, MN 55905, song.pengfei@mayo.edu)

Tissue mechanical properties can be obtained by algebraic direct inversion (ADI) of the shear-wave motion differential equation, which is insensitive to reflection and geometry of the pushing beam. Shear waves normally generated by a focused ultrasound pushing beam have limited spatial extent in depth and are transient in time, leading to noisy and unstable ADI results. A comb-shape acoustic radiation force distribution can generate shear waves with longer spatial extent and time duration, facilitating more robust ADI. For 400 μs, a linear array transducer simultaneously transmitted four unfocused pushing beams (12 on elements for each pushing beam, 8 off elements between beams) into a tissue-mimicking phantom ($\sim$1.5 kPa) with a cylindrical inclusion ($\sim$4 kPa). The ultrasound system (Verasonics Inc.) then immediately switched to flash imaging mode (frame rate = 2 kHz, spatial resolution = 0.31 mm) to measure shear-wave displacements in a 38 mm by 30 mm domain that was used for ADI. The reconstructed 2D shear elasticity map provides accurate shear elasticity estimates (background = 1.5 kPa with 7% variance; inclusion = 3.9 kPa with 15% variance) and excellent contrast between the background and inclusion.

11:25

3aBA14. A mechanism of tissue emulsification by high intensity focused ultrasound. Julianna C. Simon, Oleg A. Sapozhnikov, Vera A. Khokhlova, Yak-Nam Wang, Tatiana D. Khokhlova, Lawrence A. Crum, and Michael R. Bailey (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, jcsimon@uw.edu)

High intensity focused ultrasound (HIFU) has been shown to emulsify tissue in histotripsy created by cavitation clouds and millisecond boiling produced by shock wave heating; however, the mechanism by which millimeter-sized boiling bubbles or cavitation bubble clouds emulsify tissue into submicron-sized pieces is not well understood. Here, we experimentally test the hypothesis that acoustic atomization occurs and a miniature acoustic fountain forms (due to radiation force) at the edge of the millimeter-sized boiling bubble. Through high-speed photography, we observed the violent removal of bovine/porcine liver fragments upon focusing a 2-MHz transducer of 70 MPa peak positive and 15 MPa peak negative pressures at the tissue–air interface. Velocity of the fountain projectiles ranged from 5 to 15 m/s. When the focal amplitudes were reduced to 15 MPa peak positive and 7 MPa peak negative pressures, atomization occurred intermittently. The fountain projectiles were collected and examined histologically using H&E and NADH–diaphorase stains. Results show fragments ranging from tens of microns down to submicron in size. The larger fragments contain distinct tissue aggregates with intact cells and the small particles are cell fragments with few to no intact nuclei. [Work supported by NIH DK43881, DK070618, EB007643, DK007742, and NSBRI through NASA NCC 9–58.]

11:40

3aBA15. Vibro-acoustography beam formation with reconfigurable arrays. Matthew W. Urban (Dept. of Physio. and Biomed. Eng., 200 First St., SW, Rochester, MN 55905, urban.matthew@mayo.edu), Hermes A. S. Kamimura, Antonio A. O. Carneiro (Universidade de Sao Paulo, Ribeirao Preto, SP, Brazil), Mostafa Fatemi, and Azra Alizad (200 First St., SW, Rochester, MN 55905)

Vibro-acoustography (VA) is an imaging modality that measures the acoustic response from stimulation produced by the interaction of two ultrasound beams at different frequencies. In this work, we present a numerical study of the use of reconfigurable arrays (RCA) for VA beam formation. A parametric study of the aperture selection, number of channels, number of elements, focal distance, and steering parameters is presented in order to show the feasibility and evaluate the performance of VA imaging based on RCA. Furthermore, an optimization for beam steering based on the channel assignment is proposed for balancing the contribution of the two waves in the steered focus. The point-spread function is calculated based on angular spectrum methods using the Fresnel approximation for rectangular sources. Simulations considering arrays with 50×50 to 200×200 elements with the number of channels varying in the range of 32 to 128 are evaluated to identify the best configuration for VA. We concluded that RCA transducers can produce spatial resolution similar to confocal transducers, steering is possible in elevation and azimuthal planes, and effective setting parameters including number of elements, number of channels, maximum steering, and focal distance are suggested for VA clinical imaging.

WEDNESDAY MORNING, 2 NOVEMBER 2011 PACIFIC SALON 6/7, 8:30 A.M. TO 12:00 NOON

Session 3aEA

Engineering Acoustics and Biomedical Acoustics: Micromachined Acoustic Transducers I

Neal Hall, Cochair
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Won-Kyu Moon, Cochair
Dept. of Mechanical Engineering, Pohang Univ. of Science and Technology, Pohang, 790-784, Korea

Invited Papers

8:30

3aEA1. Capacitive micromachined ultrasonic transducers: A platform technology. Butrus (Pierre) T. Khuri-Yakub (Dept. of Elec. Eng., Stanford Univ., Stanford, CA 94305)

Capacitive micromachined ultrasonic transducers (CMUTs) have been under development as alternative transducers for generating and detecting ultrasound. The advent of silicon micromachining enables the realization of the full potential of these transducers and provides performance that makes CMUTs competitive and superior to piezoelectric transducers. In immersion applications, CMUTs are possible with fractional bandwidth of over 100%, an electromechanical coupling coefficient close to unity; are made in the form of a single element or 1-D or 2-D arrays of tens of thousand of elements, as well as annular arrays. They have been operated in the frequency range of 100 kHz to 50 MHz, and included in systems with a dynamic range of the order of 150 dB/V/Hz. Custom electronics have been developed and integrated with arrays of transducers to form compact imaging catheters. In airborne applications, CMUTs have been used as mass sensors with atto-gram resolution, and in pulse echo systems operation at high temperature (600 °C). This presentation will first review the operation of CMUTs, the technology used to make them, and some airborne and imaging applications.

8:55

3aEA2. Nickel on glass acoustic microsystems. Robert D. White, Zhengxin Zhao, Minchul Shin, Joshua S. Krause, and Shuangqin Liu (Mech. Eng., Tufts Univ., 200 College Ave., Medford, MA 02155, r.white@tufts.edu)

An electroplated nickel-on-glass surface micromachining process has been developed and applied to fabricate condenser MEMS microphone arrays and ultrasound arrays. The major advantage of these arrays over polysilicon surface micromachined sensors is the significant reduction in stray capacitance that can be achieved by using a glass substrate. For 600 μm diameter elements with a 1.5 to 2 μm sense gap, the stray capacitance per element was reduced from approximately 140–1 pF when compared to similar polysilicon

devices fabricated using the PolyMUMPS process. It is important to reduce stray capacitance, since it is the driving factor for two dominant noise sources: preamplifier voltage noise and bias feedthrough noise. In addition, stray capacitance can place a limit on system bandwidth for high frequency applications such as ultrasound. In this paper, we report on cMUT ultrasound arrays for in-air Doppler applications, as well as microphone array-on-a-chip devices for aeroacoustic measurements. Characterization of system performance and comparison to previously reported polysilicon devices [Krause *et al.* and Shin *et al.*, ASA Spring Meeting, 2011] is ongoing. Other properties such as material damping, structure residual stress, temperature sensitivity, and substrate charging are also under investigation. [Work supported by Spirit Aerosystems and Draper Labs.]

9:20

3aEA3. Piezoelectric microspeakers built on various diaphragms. Eun S. Kim (Dept. of Elec. Eng.-Electrophys., Univ. of Southern California, Los Angeles, CA 90089-0271, eskim@usc.edu)

This paper describes and compares four different types of diaphragm-based piezoelectric microspeakers built on (1) a compressively stressed silicon nitride diaphragm, (2) a parylene diaphragm, (3) a dome-shaped silicon nitride diaphragm, and (4) a PZT bimorph diaphragm. The main innovation in the first device is the usage of a wrinkled diaphragm that supports a flat diaphragm, where piezoelectric actuation happens, and allows a large bending displacement. The second device is to exploit the very low elastic modulus of parylene (a polymer material), and is built on a 1.5 μm thick parylene diaphragm with electrodes and piezoelectric ZnO film. Also described in this paper is an acoustic transducer built on a 1.5 μm thick dome-shaped silicon nitride diaphragm (2 mm in radius, with a circular clamped boundary on a silicon substrate) with electrodes and piezoelectric ZnO film. The dome diaphragm is shown to effectively release residual stress through volumetric change of its shape. Finally, this paper describes a microspeaker (composed of 8 mm square PZT bimorph and bulk-micromachined silicon) that shows flat diaphragm displacement from DC to 8 kHz. A bimorph diaphragm is formed by gluing two 127 μm thick PZT sheets and attaching them to a micromachined silicon substrate.

9:45

3aEA4. Piezoelectric thin films for a high frequency ultrasound transducer with integrated electronics. Flavio Griggio, Hyunsoo Kim, Insoo Kim, Thomas N. Jackson, Kyusun Choi, Richard L. Tutwiler, and Susan Trolier-McKinstry (Penn State Univ., 151 Mater. Res. Lab., University Park, PA 16802, STMcKinstry@psu.edu)

Lead zirconate titanate (PZT) piezoelectric films were integrated into prototype one-dimensional array transducers. Linear arrays of diaphragm transducers were prepared using PZT films of 0.5–1.7 μm in thickness and surface micromachining techniques. For this purpose, the PZT and remaining films in the stack were patterned using ion-beam or reactive ion etching and partially released from the underlying silicon substrate by XeF_2 etching. The PZT films were prepared by chemical solution deposition, and have $e_{31,f}$ piezoelectric coefficients of -6 to -12 C/m^2, depending on the crystallographic orientation. Impedance measurements on the fabricated structures showed resonance frequencies between 3 and 70 MHz for fully and partially released structures depending on the transducer dimensions and vibration modes. In-water transmit and receive functionalities have been demonstrated. A bandwidth on receive of 66% has been determined. Because of the small thickness of the piezoelectric element, the elements can be driven at less than 5 V. This enables the ultrasound system to be CMOS compatible, and hence massive miniaturization. A custom designed CMOS chip which enables beamforming on transmit, transducer excitation, amplification, digitization, and data storage was designed and fabricated.

10:10–10:30 Break

Contributed Papers

10:30

3aEA5. Minimizing noise in micromachined piezoelectric microphones. Robert Littrell (Baker-Calling Inc., 1810 14th St., Ste 102, Santa Monica, CA 90404) and Karl Grosh (Univ. of Michigan, Ann Arbor, MI 48109)

Piezoelectric microelectromechanical systems (MEMS) microphones have been researched for over 30 yr because they are relatively easy to build, output a signal without any biasing circuitry, and are relatively linear. The primary impediment to mass utilization of piezoelectric MEMS microphones has been the noise levels of these devices, which have been unacceptably high. The input referred noise of most piezoelectric MEMS microphones is greater than or equal to 55 dB(A) while commercial capacitive MEMS microphones typically have noise floors between 32 and 38 dB(A), roughly ten times lower. In order to achieve competitive noise levels in a piezoelectric MEMS microphone, a systematic approach to mechanical and electrical optimizations must be used. A key microphone metric for this optimization is acoustically generated electrical energy, as opposed to sensitivity. This optimization can be used to determine the minimum achievable noise floor for piezoelectric sensors with relatively few assumptions. Further, this optimization allows for the determination of materials, electrode patterns, layer thicknesses, and mechanical structures that minimize the noise. These models have been used to design and build piezoelectric MEMS microphones with a noise floor of 34 dB(A) and indicate that further improvements can be achieved.

10:45

3aEA6. Ultrasonic sensing using thermal mechanical noise of capacitive micro-machined transducers. Shane Lani, Sarp Satir, Gokce Gurum, Karim G. Sabra, and F. Levent Degertekin (School of Mech. Eng., Georgia Inst. of Technol., 771 Ferst Dr., NW Atlanta, GA 30332-0405)

Monolithic integration of CMUTs and CMOS electronics minimizes interconnect parasitics which allows recording the actual thermal–mechanical component of the ultrasonic noise field. Consequently, an estimate of the pulse-echo response (or Greens function) between two CMUT sensors can be obtained from the cross-correlation of thermal–mechanical noise recorded by these two sensors, as shown by Weaver and Lobkiss [JASA, **113**(5), 2611-21]. This provides a foundation for passive ultrasound imaging using only the thermal–mechanical noise field, without active transmitter elements. We designed and fabricated monolithic a 32 element CMUT-on-CMOS ring array (d = 725 μm) for intravascular imaging with low noise transimpedance amplifiers (TIAs) implemented in 0.35 μm CMOS technology. The bias voltage was set near the collapse value of the CMUT membrane to maximize receiver sensitivity. Demonstration experiments were conducted by immersing the CMUT array in a water bath to sense/image the water–air interface and compact targets from noise signals in the frequency band 14–25 MHz. These experimental results were consistent with the sensor and target locations and were validated using conventional pulse-echo measurements. This totally passive ultrasonic technique could improve

ultrasound imaging of near-field targets in the deadzone created by active transmitters biasing the receivers and may lead to imaging using evanescent waves.

11:00

3aEA7. Recent progress toward a nanostructured piezoelectric microphone. Adam D. Mathias, Jon R. Fox, Jean P. Cortes, Stephen B. Horowitz (Miltec Systems, 678 Discovery Dr. Huntsville, AL 35806), Mohan Sanghadasa, and Paul Ashley (U.S. Army—AMRDEC, Redstone Arsenal, AL 35898)

In an attempt to push the performance limits of piezoelectric MEMS microphones, several variations of micromachined acoustic sensors that contain piezoelectric zinc oxide nanorods embedded in a flexible polymer matrix were designed, fabricated, and characterized. The polymer matrix offers high compliance, high aspect ratio, ultrathin diaphragms, and low residual stress, while the zinc oxide nanorods provide high piezoelectric coupling. The devices, fabricated on a silicon substrate, consist of 100–800 μm diameter circular diaphragms composed of the piezoelectric polymer composite sandwiched between a circular gold bottom electrode and an annular gold top electrode. Electrical, mechanical, and acoustic characterization were performed on the fabricated sensors. Acoustic measurements included frequency response, sensitivity, linearity, and noise floor. Electrical properties such as resistance and capacitance, and piezoelectric properties, such as the effective piezoelectric coefficient, were also measured.

11:15

3aEA8. Electroacoustic parameter extraction of a piezoelectric microelectromechanical systems microphone. Matthew D. Williams, Benjamin A. Griffin, Tiffany N. Reagan, and Mark Sheplak (Dept. of Mech. and Aerosp. Eng., Univ. of Florida, 231 MAE-A, P.O. Box 116250, Gainesville, FL 32611-6250)

A novel suite of parameter extraction experiments were used to assess the accuracy of individual elements of a lumped element model for a microelectromechanical systems based piezoelectric microphone. The MEMS microphone was developed via model-based design utilizing the lumped element model for use in aeroacoustic applications. Laser vibrometer scans of the microphone diaphragm while subjected to electrical or pressure excitation provided experimental predictions for the effective electroacoustic piezoelectric coupling coefficient, diaphragm compliance, and mass. The experimental results were compared with analytical predictions from a piezocomposite diaphragm model for these individual lumped elements. Associated lumped element model predictions were also compared with the results of device characterization experiments. Similar trends in theory and experiments were observed, though comparative error in element values was attributed to uncertainty in model-inputs, most notably thin-film residual stresses in the microphone diaphragm. [This work was sponsored by Boeing Corporation.]

11:30

3aEA9. State space modeling of surface mount silicon microphones. A. Guclu Onaran, Caesar T. Garcia, Alex Liu, Matt Christensen (701 Tillery St., Ste. A-4B, P.O. Box 7, Austin, TX 78702), Michael Kuntzman, Karen Kirk, and Neal A. Hall (Dept. of Elec. and Comput. Eng. ENS 612A 1 University Station CO803 Austin, TX 78712-0240)

Most surface mount microelectromechanical system (MEMS) microphone packages are similar in construction, consisting of a printed circuit board with sound inlet, a MEMS die with a through-wafer etch aligned over the sound inlet, and cap which serves to protect the structure and render an enclosed back volume. From a lumped modeling perspective, this system is a network of acoustical and mechanical elements. Network models (i.e., equivalent circuit models) have proven to be the most common modeling technique for simulating important features of these microphones, including frequency response functions and internal noise floors. While these models have many advantages including their ability to be solved efficiently using modern circuit simulation software, they do not lend themselves well to an understanding of system dynamics as a decomposition of the fundamental mechanical modes of the packaged system. We present a state space model for complete MEMS microphone packages and present frequency response simulations as a superposition of the system's eigenmodes. In addition to offering insight into package behavior, we believe these models are better equipped to address advanced features such as feedback altered dynamics using internal actuation capabilities. Simulations are compared with measurements on a surface mount optical MEMS microphone discussed prior.

11:45

3aEA10. Magnetostrictive micro-loudspeaker. Thorsten S. Albach and Reinhard Lerch (Sensor Technol., Univ. of Erlangen-Nuremberg, Paul-Gordan-Str. 3-5, 91052 Erlangen, Germany, talbach@lse.eei.uni-erlangen.de)

A magnetostrictive MEMS-actuator which performs as a loudspeaker is presented. The design of the device, its theoretical modeling, as well as measurements showing its acoustical performance are outlined. Instead of a closed membrane, two combs of monomorph bending cantilevers are fabricated face to face on a silicon substrate. The small air gaps between the cantilevers lead to an advantageous reduction of the mechanical stiffness. Each cantilever consists of one active (magnetostrictive) layer and further passive layers. The cantilevers deflect simultaneously when a magnetic field is applied. An additional magnetic dc field linearizes the working point. An electro-acoustic lumped element model is developed to calculate the sound pressure level (SPL) generated in a 2 cm^3 ear volume coupler. This model incorporates finite element simulations analyzing the mechanical behavior of the cantilevers. Measurement results validate the model. The effect of the variation of different design parameters on the frequency characteristics is demonstrated. The desired characteristics of a closed membrane for example can be achieved, if the air gaps between the cantilevers are smaller than 3 μm. The prototype micro-loudspeakers have active areas of up to 3.0 mm $\times$ 2.5 mm with air gaps of 10 μm and generate a SPL up to 101 dB at 400 Hz.

Session 3aED

Education in Acoustics and Physical Acoustics: Undergraduate Research Exposition (Poster Session)

Preston S. Wilson, Cochair
Dept. of Mechanical Engineering, Univ. of Texas at Austin, 1 University Station, Austin, TX 78712-0292

Mardi C. Hastings, Cochair
G. W. Woodruff School of Mechanical Engineering, Georgia Inst. of Tech., 126 Love Bldg, 771 Ferst Dr., Atlanta, GA 30332-0405

Contributed Papers

All posters will be on display from 10:00 a.m. to 12:00 noon. All authors will be at their posters for the duration of the session.

3aED1. Ultrasonic analysis of breast tissue for pathology classification. Kristina M. Sorensen and Timothy E. Doyle (Dept. of Phys., Utah State Univ., 4415 Old Main Hill, Logan, UT 84322-4415, Kristina.Sorensen@aggiemail.usu.edu)

The effectiveness of breast conservation surgery (BCS) or lumpectomy relies heavily upon pathology to assure negative or cancer free margins. In a study to develop an intraoperative pathology method, surgical specimens from 17 breast cancer patients were tested with high-frequency (HF) ultrasound (20–80 MHz) to search for pathology sensitive features for the detection of cancer in margins during BCS. Pulse-echo and pitch-catch waveforms were obtained using two single-element 50-MHz transducers. Analysis of time-domain waveforms yielded ultrasonic attenuation and sound speed, whereas fast Fourier transforms of the waveforms produced ultrasonic spectra and cepstra for the evaluation of spectral peak density and cepstrum slope. Spectral peak density indicated significantly higher values for carcinomas and precancerous pathologies than for normal tissue. Cepstrum slope exhibited a substantial distinction between benign and adipose tissues when compared with normal and malignant pathologies. The attenuation coefficients were sensitive to fat necrosis, fibroadenoma, and invasive lobular carcinoma. A multivariate analysis of these parameters was used to further distinguish pathologic classification. Evaluation of ultrasonic attenuation, spectra, and cepstra permits differentiation between normal, adipose, benign, and malignant breast pathologies. These results indicate that HF ultrasound may assist in eliminating invasive re-excision for lumpectomy patients. [Work supported by NIH R21CA131798.]

3aED2. Foreign accent production and perception: An acoustic analysis of non-native Japanese. Lucy Gubbins and Kaori Idemaru (Dept. of East Asian Lang. and Lit., Univ. of Oregon, Eugene, OR 97403, idemaru@uoregon.edu)

What are the characteristics of non-native speech and what contributes to the perception of foreign accent by native speakers? In this study, two experiments are conducted to characterize the acoustic features of non-native Japanese production and to examine how these non-native features influence native Japanese perception of foreign accent. In the production experiment, stop consonants and vowel formants were compared between native Japanese speakers and English speakers with 2 and 4 yr of Japanese instruction. The second experiment examines native listener judgments of foreign accent using a visual analog scale. Preliminary analysis reveals that in both stop consonant and vowel production, learners vary considerably from native speakers, specifically in the production of /k/ and /t/ voice onset time, F1 of /e/, and F2 of /u/ and /i/. These features are correlated with judgment ratings by native Japanese listeners. The findings reveal that even after significant experience in the L2 classroom, speakers still struggle to achieve native-like production of various segments, offering insight into specific problem areas which native English speakers might encounter when learning Japanese pronunciation.

3aED3. The effect of musical training on auditory perception. Irene Kannyo and Caroline M. DeLong (Dept. of Psych., Rochester Inst. of Tech., 18 Lomb Memorial Dr., Rochester, NY 14623)

Previous research has shown that musical training affects the type of cues people use to discriminate between auditory stimuli. The current study investigated whether quantity of musical training and musical area of expertise (voice, percussion instrument, non-percussion instrument) affected musical feature perception. Participants with 0–4 yr of experience (13 non-musicians), 5–7 yr of experience (13 intermediate musicians), and 8 yr or more of experience (13 advanced musicians) were presented with pairs of 2.5 s novel music sequences that were identical (no change trials), differed by one musical feature (pitch change, timber change, or rhythm change), and differed by two musical features (pitch and timber change, pitch and rhythm change, or timber and rhythm change). In 64 trials, participants had to report whether they heard a change, as well as classify the specific type of change. Participants in the advanced group ($M = 91.2\%$) and intermediate groups ($M = 85.0\%$) performed significantly better than non-musicians ($M = 70.0\%$). There was no effect of area of musical expertise (voice or instrument) on musical feature change detection. These results suggest that musical training in any area increases the ability to perceive changes in pitch, timber, and rhythm across unfamiliar auditory sequences.

3aED4. Acoustic correlates of stress in Spanish infant-directed speech. Stephanie Jacobson, Nancy Ward, and Megha Sundara (UCLA Phonet. Lab., Dept. of Linguist., Los Angeles, CA 90095)

In infant speech research, children's input in often examined to determine at what age their production is affected by their language experience. In this study, we examined production of stress in Spanish infant-directed speech. Although the correlates of stress in English infant-directed speech have been examined, the same has not been done for Spanish. We analyzed infant-directed speech from eight native Spanish-speaking adults from Central and South America. The infant-directed speech was acquired by recording half-hour sessions of adults interacting in Spanish with 12-month-old infants. The acoustic measures examined to determine stress were: pitch, duration, and intensity. These measures were selected because they have been shown to be the most consistent correlates of stress in Spanish and English adult-directed speech. These measures were taken over a group of six target words: mama, globo, leche, zapato, mira, and agua. These target words allowed us to examine each of the vowels (/a/, /e/, and /o/) in both stressed and unstressed position, as well as in varying word positions. With these data we hope to discover how stress is instantiated in Spanish infant-directed

speech in order to provide a baseline for future investigations of babbling by Spanish-learning infants.

3aED5. Vibrational assessment of ice hockey goalie sticks. Linda J. Hunt (Dept. of Phys., Kettering Univ., 1700 W. University Ave., Flint, MI 48504) and Daniel A. Russell (Penn State Univ., University Park, PA 16802)

While the vast majority of offensive and defensive ice hockey players prefer composite sticks over wood, a majority of goalies prefer wood sticks over composite. To investigate this goalie preference, experimental modal analysis was performed for one wood and two composite goalie sticks in order to extract mode shapes, frequencies, and damping coefficients. Wood and composite sticks were both shown to exhibit modes with antinodes close to the hand location, and frequencies within the range of maximum sensitivity. The sequence of mode shapes was consistent for wood and composite sticks. The wood stick had lower mode frequencies and higher damping coefficients than the composite sticks. Additional testing was performed on a composite goalie stick with and without the addition of tape at the knob end (required by the NHL). This mass loading lowered the frequencies and increased the damping coefficients and moved the nodes of vibration. These differences in mode shapes, frequencies, and damping along with the effects of tape will be discussed in terms of the influence on the perception of feel for goalies preferring wood sticks.

3aED6. Study of the effects of different endpin materials on cello sound characteristics. Clinton R. Fleming, Cassey R. Stypowany, Robert D. Celmer, and Michelle C. Vigeant (Acoust. Prog. and Lab., Mech. Eng. Dept., Univ. of Hartford, 200 Bloomfield Ave., West Hartford, CT 06117, clntfleming@gmail.com)

The purpose of this study was to investigate the effects of changing a cello's endpin material and boundary conditions on the sound and vibration characteristics of a cello. It was hypothesized that an endpin made of a denser material than stainless steel, which is traditionally used, would improve the tone quality of the cello. In terms of endpin boundary conditions, it was hypothesized that using a shorter endpin with fixed end conditions might also improve the vibration characteristics and sound radiation efficiency of the cello. Objective and subjective tests were conducted to examine the effects of the different endpin materials. Sound power level output and vibration measurements of a cellist playing on different endpins were obtained following ISO 3741. In general, sound power levels and measured vibrations were consistent for all endpins for all notes tested. For the subjective study, volunteer cellists played selected excerpts with the different endpins, not knowing which endpin they were using. This testing showed that the tungsten endpin gives the cello a tone quality with similar warmth to the stock endpin but makes the cello less responsive. The results of both the objective and subjective tests for all endpin materials will be presented.

3aED7. Noise reduction in deployable infrasound arrays through unconventional clover hose layouts. John R. Clay (Dept. of Elec. and Comput. Eng., Univ. of Wyoming, 1000 E. University Ave., Laramie, WY 82071, jclay5@uwyo.edu) and Mihan H. McKenna (US Army ERDC, CEERD-GS-S, Vicksburg, MS 39180)

Infrasound is acoustic energy that lies below the human hearing threshold. This energy is produced from numerous naturally occurring and manmade sources with the ability to propagate hundreds to thousands of miles, depending on the source strength. Infrasound sensors are commonly deployed in networked, multi-sensor arrays for signal localization and characterization. Deployable infrasound sensors, such as Chaparral and IML rely on the use of soaker hoses for noise reduction and long-wavelength spatial averaging. Traditionally, sets of 25–50 ft hoses attach to the ports on each side of the infrasound sensor and are terminated with a cap at the end of the hose. In August of 2010, hose filter geometry testing was conducted using IML ST infrasound sensors designed to minimize the filter footprint by exploring alternatives to the conventional straight hose layouts while maximizing signal-to-noise in desired pass-bands. The optimal arrangement, "Clover Hose Layout," removed the terminator at the end of the hose and looped it to connect to an adjacent port with substantial environmental noise reduction in the 0.5–50 Hz range. Results are presented comparing a suite of hose arrangements including conventional straight hose layouts under calibrated source testing.

3aED8. Acoustical effect of progressive undercutting of percussive aluminum bars. Eric M. Laukkanen and Randy Worland (Dept. of Phys., Univ. of Puget Sound, 1500 N Warner St., Tacoma, WA 98416)

Standard vibraphone bars consist of aluminum beams which are traditionally tuned with an arched undercut, for the purpose of aligning the musical overtones harmonically. The acoustical effect of various progressions of undercuts on aluminum bars was studied using both an aluminum bar and a finite element computer model. The spectral signature of the aluminum bar was examined with a spectrum analyzer, and the corresponding eigenmodes were imaged with an electronic speckle pattern interferometer. These methods were used to analyze the changes in natural frequencies of the bar as matter was removed from various locations. Additionally, the aural character of each cut was captured with an audio recording, and the fundamental tone was normalized over all recordings to make possible a subjective comparison of the timbral differences of differently cut bars.

3aED9. Investigations of the just-noticeable-difference of clarity index (C80). Christopher M. Jasinski, Meghan J. Ahearn, Clothilde B. Giacomoni, Caitlin I. Ormsbee, Matthew J. Schaeffler, and Adam P. Wells (Acoust. Prog. and Lab., Mech. Eng. Dept., Univ. of Hartford, 200 Bloomfield Ave., West Hartford, CT 06117, jasinski@hartford.edu)

The clarity index for music (C80) is a valuable room acoustic parameter, as it is an objective measure of how listeners perceive clarity. Knowing the just noticeable difference (JND) of C80 is of great importance to concert hall designers. Limited previous research has been conducted to find the C80 JND, and the major studies have limitations, including a small sample size. The reported JND is approximately 1 dB. An ongoing investigation is being conducted at the University of Hartford to establish the validity of these results by experimentally determining the JND, including an investigation on test method. In the first study (Ahearn *et al*., 2009), the subject pool was increased to 51 subjects, using test methods from prior research. The next study (Giacomoni *et al*., 2010) compared two C80 JND test methods, including allowing subjects to switch between signals in real-time. The results revealed the importance of subject training. In the third study (Wells *et al*., 2010), the switch method was used. The resulting JNDs were 1.6, 3.8, and 4.0 dB for each of the studies, respectively, showing that the C80 JND may be significantly higher than previously thought. The testing methods, results, and comparison to the previous studies will be described.

3aED10. Source localization in a reverberant environment using first reflections. Eric S. Haapaniemi, Andrew J. Femminineo, Laura M. Williamson, and David R. Dowling (Dept. of Mech. Eng., Univ. of Michigan, 2350 Hayward, Ann Arbor, MI 48109, ehaap@umich.edu)

Matched field processing (MFP) has been shown to be effective for remote sound source localization when the receiving array clearly records direct-path sound. Unfortunately, in imperfectly characterized confined environments, echoes and reverberation commonly degrade localization performance. However, inclusion of first reflections in the requisite MFP field model should improve localization performance and such inclusion may be tractable as well. This presentation describes an acoustic technology development effort focused on source localization in a simple confined environment using both direct-path and once-reflected sounds. Experiments were conducted in a 1.0-m-deep and 1.07-m-diameter cylindrical water tank having a reverberation time of ~10 ms using a single sound projector and a linear receiving array of 16 hydrophones. Measured localization performance is reported for impulsive (100 μs) and longer duration sounds having frequencies from 20 to 150 kHz. Appropriate cropping of the signal coda and inclusion of a simplified characterization of the reflection properties of the tank's walls allows successful integration of first reflections that improve MFP localization results. The intended application of this research is localizing sub-visual cavitation bubbles and other hydroacoustic sound sources in hydrodynamic test facilities. [Work sponsored by NAVSEA through the Naval Engineering Education Center.]

3aED11. Experimental investigation of shock structure in turbulent Coanda jets. Richard J. Shafer and Caroline P. Lubert (Dept. of Mathematics and Statistics, 102 Roop Hall, MSC 1911, Harrisonburg, VA 22807)

Although Coanda surfaces are extremely useful for applications in industrial exhaust, the generation of noise in supersonic flows around such

3a WED. AM

surfaces is not well understood. In order to effectively engineer a solution to this noise problem, the noise generation must be able to be accurately modeled. The two dominating generators of sound are turbulent mixing noise and shock associated noise (SAN). The first step necessary to model SAN is a precise predictor of the shock structure within the jet. A MATLAB algorithm is developed to translate Schlieren photos of the flow into qualitative shock location data. This data will be used in conjunction with the method of characteristics to create a model which predicts shock location around a Coanda surface for several slot widths and exit pressures.

3aED12. Effects of auditory feedback on instrumentalists timber production. Madeline R. Huberth and Timothy A. McKay (Dept. of Phys., Univ. of Michigan, 450 Church St., Ann Arbor, MI 48104, mhuberth@umich.edu)

Previous work has shown that the presence or absence of auditory feedback in musicians has little effect on the performance factors such as note accuracy, timing, and dynamics. This study explored the extent to which an instrumentalist's timber is affected by various feedback conditions. Nine cellists were recorded playing excerpts from the orchestral literature with their hearing completely unimpeded, masked by pink noise, and masked by orchestral recordings of the excerpts requested for the experiment, simulating orchestral playing. Changes in relative harmonic strengths across the three conditions were focused on in analysis. Overall, the results show that the auditory feedback condition has minimal effect on timber production.

3aED13. Dynamic behavior of microwave-cured guitar wood. David-M. Lozupone, Aldean J. Yasay, David A. Dick, Allen R. Beaudry, Andrew T. Sorenson, Stalin L. Vera, and Robert D. Celmer (Acoust. Prog. and Lab., Mech. Eng. Dept., Univ. of Hartford, 200 Bloomfield Ave., West Hartford, CT 06117)

An ongoing investigation has been conducted to measure and analyze vibration properties of guitar-body wood samples with different wood finish curing methods for Taylor Guitars. The current method uses ultraviolet radiation to cure a lacquer finish. Virginia Tech's (VT) Material Science & Engineering Department has proposed an alternative method using microwave radiation to cure the finishes, now in its third stage of development. Microwaves (MW) have longer wavelengths than ultraviolet (UV) radiation, allowing greater penetration depth and increased diffusion between wood and lacquer molecules. To investigate the effect of the finish on dynamic behavior, modal analyses have been conducted at the University of Hartford Acoustics Engineering Laboratory using unfinished, UV-cured, and MW-cured guitar wood samples (spruce, maple, ash, and rosewood). Samples were characterized at bending and torsional modes up to 3.2 kHz. Overall results to date demonstrate that both UV-cured and MW-cured samples have higher damping than unfinished samples. Further, the damping provided by the MW method appears to correlate with the number and thickness of filler, finish and top coats. The current results suggest it is possible to produce similar dynamic behavior to UV curing using the MW technique. [Work done in collaboration with Virginia Tech and Taylor Guitars.]

3aED14. Using acoustics to study the effectiveness of fish habitat restoration: A comparison of methods. Michael Bollinger (Biology Dept., The Penn State Univ., State College, PA 16804, mbollinger89@gmail.com), Jennifer L. Miksis-Olds (ARL, Penn State, State College, PA 16804), and Laura E. Madden (Wildlife and Fisheries Sci., Penn State, State College, PA 16804)

Many Pennsylvania reservoirs are created by clearing valleys prior to damming streams. These impoundments are typically devoid of appropriate benthic habitat for fish. The Pennsylvania Fish and Boat Commission has been placing artificial habitat structures in reservoirs across Pennsylvania for 30 years in an effort to increase game-fish production. However, the effectiveness of these efforts is unknown. Active acoustic technology is being considered as an alternative to conventional survey methods for assessing the effectiveness of artificial structure in F.J. Sayers Reservoir in Howard, PA. An autonomous echosounder [(ASL Environmental Sciences, 460 kHz Acoustic Water Column Profiler (AWCP)] was deployed horizontally near artificial rock structures to monitor associated fish activity. The AWCP recorded acoustic backscatter at 1-s intervals continuously for two days. Data was analyzed by (1) manually counting fish tracks on an echogram for each hour and range and (2) calculating integrated volume backscatter values per hour for each range. Both analysis methods showed similar trends in fish activity during the study period. Results agreed with expected behavioral patterns for the species in this body of water; fish activity increased during the night and decreased during the day, with the exception of a mid-day increase on one day.

3aED15. The influence of talker and accent variability on spoken word identification and discrimination. Kierra Villines, Tessa Bent, and Rachael F. Holt (Indiana Univ., 200 S. Jordan Ave., Bloominton, IN 47405, kvillines_08@yahoo.com)

In spoken word identification and memory tasks, stimulus variability from numerous sources impairs performance. The phonetic-relevance hypothesis (Sommers *et al*., 2006) proposes that only acoustic properties that influence phonetic perception (e.g., speaking rate) cause decrements in perception. In the current study, the influence of foreign-accent variability on identification and discrimination of spoken words was evaluated. In experiment 1, word identification in noise was tested in single-talker and two multiple-talker conditions: multiple talkers with the same accent or multiple talkers with different accents. In experiment 2, participants discriminated minimal pairs from a single talker, different talkers with the same accent, and talkers with different accents. Identification and discrimination performance was the highest in the single-talker conditions, but there was no difference between the single-accent and multiple-accent conditions. However, reaction time in the discrimination task was the highest in the multiple accent condition with no difference between the single-talker and single-accent conditions. Thus, the presence of multiple accents does not decrease accuracy beyond the multiple-talker effect, but processing time increases with the addition of multiple accents. These results provide partial support for the phonetic-relevance hypothesis. [Work supported by NIDCD R21DC010027 and Indiana University.]

3aED16. Application of time reversal to sources in non-ideal environments. Blaine Harker and Brian Anderson (Dept. of Phys. and Astron., Brigham Young Univ., N283 ESC, Provo, UT 84602)

Time reversal (TR) is a technique of localizing acoustical sources using a time reversal mirror (TRM) and is especially useful in reverberant environments. TR is commonly used to find acoustically small sources and in many cases pulsed waveforms are used. Here TR is applied to distributed sources using long duration waveforms. This is done using a straightforward, theoretical, point source propagation computer modeling and data from experimental measurements of jet noise. The quality of the TR focusing versus using different numbers of microphones to constitute the TRM is determined. These results are compared to other imaging techniques and theories of sound radiation from jets.

3aED17. On the directivity of musical instruments: Effects on room acoustics calculations. Kristina M. Sorensen, Nicholas J. Eyring II, Timothy W. Leishman, and Nathan G. W. Eyring (Acoust. Res. Group, Dept. of Phys. and Astronomy, Brigham Young Univ., N283 ESC, Provo, UT 84602, twleishman@byu.edu)

Measurements of musical instruments in an anechoic chamber at Brigham Young University are yielding high-resolution directivity data and balloon plots that may be analyzed and visualized as functions of time or frequency. Historically, room acoustics calculations involving directivity of sound sources have relied on well-defined radiation characteristics including principal radiation axes, directivity factors, beamwidths, etc. In recent times, room simulation software packages have incorporated more comprehensive steady state directivity data for loudspeakers that vary as functions of proportional frequency bands. However, in contrast to loudspeakers, many musical instruments have directivity patterns that are not well defined and that may vary more erratically as functions of pitch or other musical characteristics. This presentation explores the measured directivity properties of a few musical instruments and discusses how they might be encapsulated and incorporated into predictive room acoustics calculations.

Session 3aMU

Musical Acoustics: Expressivity in Digital Music Synthesis

James W. Beauchamp, Chair
School of Music, Univ. of Illinois at Urbana-Champaign, 1114 W. Nevada, Urbana, IL 61801

Chair's Introduction—8:00

Invited Papers

8:05

3aMU1. Extracting heterogeneous structured features from music performance. Gang Ren, Dave Headlam, Stephen Roessner, Justin Lundberg, and Mark F. Bocko (Dept. of Elec. and Comput. Eng., Edmund A. Hajim School of Eng. and Appl. Sci., Univ. of Rochester, Rochester, NY 14627)

In music performance, the musician adds artistic expressive elements beyond the information contained in conventional western music scores. As micro-fluctuations these expressive dimensions convey emotions and essential interpretative information and can be measured and compared quantitatively, over large and small scales, and evaluated for their effect on aspects of the performance. We present a heterogeneous expressive music feature description that include both inter-note features that extend over musical phrases composed of several notes, and intra-note features that represent the internal variations within each musical note. The intra-note features include pitch and pitch deviation, dynamic level, timber, articulation and vibrato. The inter-note features include timing and dynamics, as well as timber, pitch deviation, articulation, and vibrato extending across multiple notes and musical phrases. A complete multi-dimensional feature description for every note is unnecessary because there is a hierarchy of note functions in a musical phrase and certain features may be more or less important depending upon their function in the phrase. Thus, we introduce a heterogeneous representation that affords the flexibility to reflect the hierarchical significance of each note or phase segment. Specific visualization and animation schemes of the proposed representation, to facilitate user interaction with the music, are also presented.

8:30

3aMU2. Synthesis of bowing controls applied to violin sound generation. Esteban Maestre (CCRMA–Dept. of Music, Stanford Univ., 660 Lomita Dr., Stanford, CA 94305, esteban@ccrma.stanford.edu)

Within instrumental sound synthesis, one of the most challenging aspects of automatic performance resides on the ability to faithfully reproduce the expressive nuances naturally conveyed by a musician when controlling a musical instrument. To that regard, much research effort has been devoted in the seek for obtaining natural-sounding synthetic performances from an annotated input score. Despite continuous improvements on sound synthesis techniques, appropriately mapping score annotations to synthesize controls still remains an interesting research problem, especially for excitation-continuous musical instruments. Along these lines, it is presented here our recent work on modeling bowing control in violin performance and its application to sound synthesis. Nearly, non-intrusive sensing techniques allow for accurate acquisition of relevant timber-related bowing control parameter signals. The temporal contours of bow velocity, bow pressing force, and bow-bridge distance are modeled as sequences of short Bzier cubic curve segments. A database of parametric representations of real performance data is used to construct a generative model able to synthesize bowing controls from an annotated score. Synthetic bowing controls are used to generate realistic performances using a digital waveguide physical model, and a spectral-domain sample-based synthesizer. Obtained results demonstrate the potential of explicitly modeling instrumental control for expressive sound synthesis.

8:55

3aMU3. Where is the expressivity in singing? Johan Sundberg (Dept. of Speech Music Hearing, School of Comput. Sci. and Commun., KTH, SE-10044, Stockholm)

When computers convert music scores to sounding music, performances void of musical expression typically emerge, mostly perceived as musical disasters. This is a striking illustration of the importance of the contributions of musicians. Excellent tools for exploring these contributions are synthesized performances and/or processing of real performances. In the 1970s, the KTH music group started to formulate rules for music performance, now integrated into the KTH Director Musices performance grammar. Some principles underlying its performance rules can be identified: (1) marking of phrase structure, (2) increasing contrasts along acoustic parameters, and (3) emphasizing important and/or less expected notes. These principles can be observed also in phonetics and speech prosody. Also the acoustic codes used for these principles in music performance are similar to those used in speech. Just as pitch is raised in agitated speech, singers have recently been found to increase contrast along the pitch dimension by sharpening the peak note in agitated phrases. In a recent experiment, audio processing software was used to quasi-correct the intonation of the peak tone in a handful of examples sung by a professional baritone singer. A listening test indicated that the sharpening added to the expressivity of the performance.

9:20

3aMU4. Synthesizing melodic expression. Christopher S. Raphael (School of Informatics and Computing, Indiana Univ., 901 10th St., Bloomington, IN 47408)

A method for expressive melody synthesis is presented seeking to capture the prosodic (stress and directional) element of musical interpretation. An expressive performance is represented as a note-level annotation, classifying each note according to a small alphabet of symbols describing the role of the note within a larger context. An audio performance of the melody is represented in terms of two time-varying functions describing the evolving frequency and intensity. A method is presented that transforms the expressive annotation into the frequency and intensity functions, thus giving the audio performance. The problem of expressive rendering is then cast as estimation of the most likely sequence of hidden variables corresponding to the prosodic annotation. Examples are presented on a data set of around 50 folk-like melodies, realized both from hand-marked and estimated annotations.

9:45

3aMU5. Virtual instrument player using probabilistic gesture grammar. Eric Lindemann (Synful, 2975 18th St. Boulder, CO 80304)

Expressive instruments such as violin require subtle control gestures: variations in bow velocity and pressure over the course of a note, changes in vibrato depth and speed, portamento gestures, etc. Music synthesizers that attempt to emulate these instruments are often driven from keyboard controllers or directly from score editor or sequencer software. These synthetic control sources do not generally provide the level of detailed control required by expressive instruments. Even if a synthesizer offers a rich set of realistic continuous control inputs, the effort and skill required by the user to manage these controls, e.g., by drawing expression and vibrato controller curves in a MIDI sequencer, is considerable. Often the user would prefer to treat the synthesizer as a combination instrument and virtual player. The virtual player receives limited score-like control input and infers the more detailed gestural control. This presentation describes a virtual player mechanism that employs a probabilistic note-gesture grammar to aid with control inference. The virtual player uses the grammar to select various gestural patterns that are idiomatically appropriate given the limited control input. The virtual player is used to drive a concatenative music synthesizer employing a rich database of recorded note gestures.

10:10–10:25 Break

10:25

3aMU6. Playing a virtual drum from a real one. Miller Puckette (Dept. of Music (0099), Univ. of California, San Diego, La Jolla, CA 92093-0099)

An electronic musical percussion instrument is described which can be played by tapping or otherwise playing on a small physical object, whose vibrations are picked up and used to activate the virtual one. To do this, a ceramic tile or other small rigid object is attached to a contact microphone. Any resonances of the physical system are filtered out using linear predictive analysis, so that the resulting residual signal approximates the excitation the player introduces into the system. The resulting audio signal could then be forward filtered to recover the tile's own sound, but instead it is fed into a nonlinear reverberator to simulate a variety of real or fanciful percussion instruments. The result is a highly expressive and playable electronic percussion instrument, that is, both easy and inexpensive to build. The effects of various design parameters of the nonlinear reverberator on the resulting sound are discussed.

10:50

3aMU7. Music from "an evil subterranean beast." Chris Chafe (CCRMA, Stanford Univ., Stanford, CA 94305)

Two recent compositions have capitalized on the great expressive range of a custom-designed music synthesis algorithm. The "Animal" plays a part in Tomato Quintet (a music installation) and Phasor (a contrabass and computer piece). The algorithm is played with live controls, respectively, updates from CO_2 sensors and signals from a sensor bow. Nuanced deflections of parameter values elicit characteristic and sometimes quirky behavior. The bounded set of sonic behaviors goes into making up an identifiable personality whose moods or temperaments can be traversed in the music which exploits it. "The computerized sounds were spacey and sometimes menacing, sounding at times like Chafe was trying to tame an evil subterranean beast." (H. Ying, Global Times, 2011). Animal's algorithm is comprised of the logistic map with two parallel waveguides in its feedback path. Animal can be categorized as a "meta-physical" model or modeling abstraction as seen in earlier integrations of families of physical models (P. Cook, ICMC, 1992) and recombinations of physical model components in physically impossible ways (C. Burns, Composition, 2003). Abstraction opens the door to inclusion of mathematical "parts" from other domains. In Animal's case, the logistic map has been borrowed from population biology.

Contributed Papers

11:15

3aMU8. Nonlinear coupling and tension effects in a real-time physical model of a banjo. Rolf Bader and Florian Pfeifle (Musicological Inst., Univ. of Hamburg, Neue Rabenstrasse 13, 20354 Hamburg, Germany, Florian.Pfeifle@haw-hamburg.de)

A real-time physical model of the Banjo as proposed by Pfeifle and Bader [J. Acoust. Soci. Am. (2009)] is extended with nonlinear effects on the membrane resulting from the added force of the bridge and the thereby arising anisotropic tension distribution on the membrane. It is shown that this effect directly influences the vibrational behavior of the physical model and yields a more realistic sound. In a second step a non-linear excitation mechanism of the string is presented, modeling the interaction between a standard metal banjo fingerpick and the string of the banjo. The pressure and velocity of the fingerpick are used as control parameters for the model. The whole model is implemented on a FPGA and can be played and controlled in real-time.

11:30

3aMU9. An active learning-based interface for synthesizer programming. Mark B. Cartwright and Bryan Pardo (Dept. of EECS, Northwestern Univ., 2145 Sheridan Rd., Tech. L359, Evanston, IL 60208, mcartwright@u.northwestern.edu)

Commercial software synthesizer programming interfaces are usually complex and tedious. They discourage novice users from exploring timbers outside the confines of "factory presets," and they take expert users out of their flow state. We present a new interface for programming synthesizers that enables both novices and experts to quickly find their target sound in the large, generative timber spaces of synthesizers. The interface does not utilize knobs and sliders that directly control synthesis parameters. It instead utilizes a query-by-example based approach combined with relevance feedback and active learning to create an interactive, personalized search that allows for exploration. We analyze the effectiveness of this new interface through a user study with four populations of varying types of experience in which we compare this approach to a traditional synthesizer interface. [Work supported by the National Science Foundation Graduate Research Fellowship.]

WEDNESDAY MORNING, 2 NOVEMBER 2011 GARDEN SALON 2, 7:40 A.M. TO 12:00 NOON

Session 3aNS

Noise and Committee on Standards: Impact of New Environmental Protection Agency (EPA) Regulation on Hearing Protection

William J. Murphy, Chair
National Inst. for Occupational Safety and Health, 4676 Columbia Pkwy., Cincinnati, OH 45226

Chair's Introduction—7:40

Invited Papers

7:45

3aNS1. Hearing protector labeling—Yes or no? Ken Feith (13404 Query Mill Rd., North Potomac, MD 20878, feithk@comcast.net)

The federal government has been accused of requiring needless product labeling in order to protect citizens from themselves. While there is probably some truth in these accusations, the key point of labeling is lost in the noise of criticism—at least in the case of hearing protector devices. If we were to consider the one word that most influences our life actions we would find it to be the word "decisions." A moment's thought would reveal that from the minute we wake in the morning until we achieve a deep state of sleep, we are confronted with the need to make decisions. Every move, every action, is determined by either conscious or subconscious decision making. To that end, we need to consider those elements that influence our decision making process—understandable and accurate information. Thus, the role of federal labeling of select products is to provide an understandable and accurate means for making decisions that may have an impact on our ability to function effectively in our work, maintain our good health, avoid personal injury, and the list goes on. This presentation reveals the secrets behind regulatory labeling.

8:15

3aNS2. Evaluation of hearing protection devices with high-amplitude impulse noise. Karl Buck, Sebastien DeMezzo, and Pascal Hamery (ISL, APC, 5 r. du Gn. Cassagnou, BP 70034, 68301 St. Louis, France, karl.buck@isl.eu)

Presently, the evaluation of HPDs (hearing protection devices) is mainly based on audiometric threshold methods. However, in the military environment soldiers may be exposed to impulse noise levels of 190 dB and higher if large caliber weapons or IEDs (improvised explosive devices) are considered. In this case it seems reasonable to expect that an evaluation done at hearing threshold might not represent the protection which will be encountered at these highest levels. Therefore, it is necessary to evaluate the HPDs with signals close to the effective exposure. ISL has developed procedures using explosive charges creating very high levels of impulse noise (up to 195 dB peak) for the evaluation of HPDs. ISL also has developed an artificial head that can withstand this type of exposure and has sufficient self-insertion loss for the measurement of double hearing protection. The presentation will give a description of the techniques used at the ISL. It will present some results showing the possibilities to determine the performance of different types of HPDs when subjected to extreme impulse noise levels. Moreover, the new version of the ISL artificial head, compatible ANSI-ASA-S12.42-2010, and possible problems when using it will be presented.

8:45

3aNS3. Comparison of three acoustics test fixtures for impulse peak insertion loss. William J. Murphy (Hearing Loss Prevention Team, NIOSH, 4676 Columbia Parkway MS C-27, Cincinnati, OH 45226, wjm4@cdc.gov), Gregory A. Flamme (Western Michigan Univ., Kalamazoo, MI 49008-5243), Deanna K. Meinke, Donald S. Finan (Univ. of Northern Colorado, Greeley, CO 80639), James Lankford (Northern Illinois Univ., DeKalb, IL 60115), Amir Khan (NIOSH, Cincinnati, OH 45226-1998), Jacob Sondergaard (G.R.A.S. Sound and Vib., North Olmsted, OH 44070), and Michael Stewart (Central Michigan Univ., Mount Pleasant, MI 48859)

Acoustic test fixtures (ATF) for testing the impulse peak insertion loss (IPIL) of a hearing protector are described by American National Standard ANSI S12.42-2010. The self-insertion loss, ear simulator design (canals, microphone, and temperature), hardness of the area surrounding the pinna, and the anthropometric shape of the head has been specified in the standard. The IPILs of four protector conditions were evaluated with three ATFs during an outdoor field study using firearm noise. The Etymotic Research ER20 musicians' earplug and electronic (EB1 earplugs), the Peltor Tactical Pro earmuffs, and a combination of the TacticalPro and ER20 protectors were

tested at 130, 150, and 170 dB peak sound pressure level with the Institute de Saint Louis heated and unheated fixture and the GRAS 45CB heated ATF. IPILs exhibited good agreement across all three fixtures for earplugs. Significant differences were observed between the fixtures for the earmuff-only condition. These differences were more evident for the double-protection condition. [Portions of this work were supported by the U.S. EPA Interagency Agreement DW75921973-01-0.]

9:15

3aNS4. Insertion-loss and transfer-function performance of two new acoustical test fixtures complying with ANSI S12.42-2010, relative to performance of prior test fixtures and to real-ear data. Elliott H. Berger, Ronald W. Kieper, and Michael E. Stergar (3M Occupational Health & Environ. Safety Div., 7911 Zionsville Rd., Indianapolis, IN 46268-1657)

The most recent American National Standards Institute (ANSI) standard for the measurement of the insertion loss of hearing protection devices (HPDs), ANSI/American Standards Association (ASA) S12.42-2010, specifies a new-concept acoustical test fixture (ATF). It is similar to some existing ATFs but differs in terms of the required earcanal length, inclusion of a simulated flesh lining the earcanal, and a heater to bring the test fixture to approximate body temperature. These features were deemed necessary to develop a device that provides insertion loss data with reasonable correspondence to performance on human heads, as the ATF is the preferred method in the standard for tests on certain electronic earplugs and for all impulse testing. Within a year of the issuance of the standard, at least two ATFs [one produced by G.R.A.S. Sound and Vibration and the other produced by the Institute of St. Louis] became available. The studies reported herein will provide an initial evaluation of these two heads compared to prior art, based on ATF insertion-loss measurements for a sample of passive earplugs and earmuffs versus real-ear attenuation at threshold per ANSI S3.19-1974. Additionally, the ATFs' transfer function of the open ear in a diffuse field will also be reported.

9:45

3aNS5. Attenuation performance of active noise reduction headsets using American National Standard s12.42 and the proposed U.S. Environmental Protection Agency Hearing Protector Labeling Regulation. Richard L. McKinley, Hillary L. Gallagher (Air Force Res. Lab., 2610 Seventh St. WPAFB, OH 45433, richard.mckinley@wpafb.usaf.mil), Melissa A. Theis (Oak Ridge Inst. for Sci. and Education, TN 37831), and Paul C. Schley (Ball Aerosp. and Technologies Corp., Fairborn, OH 45324)

The attenuation performance and noise reduction rating (NRR) of six commercially available active noise reduction (ANR) headsets was assessed using the proposed environmental protection agency (EPA) regulation. The passive attenuation results were collected using American National Standard Institute (ANSI) S12.6 method for measuring real-ear attenuation at threshold (REAT) of hearing protectors while the active attenuations results were collected using ANSI S12.42 methods for the measurement of insertion loss of hearing protection devices in continuous or impulsive noise using microphone-in-real-ear (MIRE) or acoustic test fixture procedures. ANSI/ASA S12.68 methods of estimating effective A-weighted sound pressure levels when hearing protectors are worn was used to compute noise reduction metrics including the noise reduction statistic A-weighted (NRSA) and the graphical noise reduction statistic (NRSG). The proposed NRR labels for the ANR headsets were computer per the guidance in the draft U.S. EPA regulation. The presentation will include the baseline passive, active, and total attenuation, the NRSA and the Graphical NRSG, and the proposed EPA labels for passive attenuation and total attenuation while in an active mode.

10:15–10:30 Break

Contributed Papers

10:30

3aNS6. 1.Comparison of the HPDLAB and REATMASTER software/hardware systems for ANSI S12.6 testing. David C. Byrne (NIOSH–Taft Labs., 4676 Columbia Parkway, MS C-27, Cincinnati, OH 45226, DByrne@cdc.gov), Caryn C. Perry (Univ. of Cincinnati, Cincinnati, OH 45267), and William J. Murphy (NIOSH–Taft Labs., Cincinnati, OH 45226)

The American National Standard Methods for Measuring the Real-Ear Attenuation of Hearing Protectors (ANSI S12.6-2008) requires a Békésy procedure for testing occluded and unoccluded thresholds. Since 2002, the National Institute for Occupational Safety and Health (NIOSH) has used the custom-designed HPDLAB software operating Tucker-Davis Technologies System 3 hardware. ViAcoustics, Nelson Acoustics, NASA, and NIOSH researchers recently developed REATMASTER which runs on National Instruments hardware in the LABVIEW environment. Ten subjects were trained by the experimenter on how to fit a passive earmuff and were qualified according to the requirements of ANSI S12.6-2008. The laboratory was configured such that diffuse sound field thresholds were tested with either the HPDLAB or REATMASTER hardware by flipping a toggle switch. The earmuff was not touched or re-positioned between test trials with the two different hardware/software systems. The test sequence for the order of open and occluded measurements was counterbalanced across occluded conditions and hardware system. Results from this testing were used to validate the REATMASTER system for its ability to produce accurate threshold data. Preliminary results indicate no significant differences between the two systems.

10:45

3aNS7. Calibration details for the impulse peak insertion loss measurement. William J. Murphy and Julia A. Vernon (Hearing Loss Prevention Team, NIOSH, 4676 Columbia Parkway MS C-27, Cincinnati, OH 45226-1998, wjm4@cdc.gov)

The American National Standard ANSI S12.42-2010 specifies the measurement of hearing protector performance in the presence of impulse noise. A series of calibration impulses are recorded from an acoustic test fixture (ATF) and a field microphone for peak sound pressure levels of 130, 150, and 170 dB. The averaged acoustic transfer function between the ATF and field microphone is calculated as follows:

$$\overline{H}_{\overline{ATF-FF}}(f) = \overline{\langle FFT(P_{\mathrm{ATF},i}(t))/FFT(P_{FF,i}(t))\rangle}.$$

The transfer function is computed for each of the ranges of impulse levels and is applied to the field microphone measurements to estimate the unoccluded fixture levels of the ATF when hearing protection is being tested. This method allows a comparison between occluded and unoccluded waveforms. The calibration transfer function is affected by the time-alignment of the field impulse peaks, time-windowing of the impulses, and compensation for any dc bias. Time-alignment significantly affected the accuracy of predicting individual calibration levels with $\overline{H}_{\overline{\mathrm{ATF}-\mathrm{FF}}}$. The prediction error variance was less at 170 dB than at 130 dB impulses. The time-window was varied from 2.5 to 100 ms preceding the peak of the field impulse. [Portions of this work were supported by the U.S. EPA Interagency Agreement DW75921973-01-0.]

11:00

3aNS8. Measuring, rating, and comparing the real ear attenuation at threshold of four earplugs. William J. Murphy, Mark R. Stephenson, and David C. Byrne (Hearing Loss Prevention Team, NIOSH, 4676 Columbia Parkway MS C-27, Cincinnati, OH 45226-1998, wjm4@cdc.gov)

The effect of training instruction, whether presented as the manufacturer's printed instructions, a short video training session, specific to the product, or as a one-on-one training session, was evaluated using four hearing protection devices with eight groups of subjects. The Howard Leight Fusion and Airsoft premolded earplugs and the Moldex PuraFit and EAR Classic foam earplugs were tested. Naïve subjects were recruited and tested using three different forms of training: written, video, and individual training. The differences between group averages for A-weighted attenuation were not statistically significant when compared between the video or the written instruction conditions, regardless of presentation order. The experimenter-trained A-weighted attenuations were significantly greater than the written and video instruction for most of the protectors and groups. For each earplug, the noise reduction statistic for A-weighting (NRSA) and the associated confidence intervals were calculated for the 90th and 10th percentiles of protection. Across subject groups for each protector, the differences between NRSA ratings were found to be not statistically significant. Several comparisons evaluating the order of testing, the type of testing, and statistical tests of the performance across the groups are presented. [Portions of this work were supported by the U.S. EPA Interagency Agreement DW75921973-01-0.]

11:15

3aNS9. ANSI S12.42-2010 measurements of impulse peak insertion loss for passive hearing protectors. Kevin Michael (Michael & Assoc., Inc., 2766 W. College Ave., St. 1, State College, PA 16801, kevin@michaelassociates.com) and Jeff G. Schmitt (ViAcoust., Austin, TX 78745)

Until recently, a standardized measurement procedure to determine peak insertion loss for hearing protectors was not available. This has led to confusion and uncertainty for hearing protector users who commonly use the devices in impulse noise, such as gunfire. Released in 2010, ANSI S12.42-2010 defines a test method and analysis procedure for measuring hearing protector impulse peak insertion loss. The required test fixture has recently become commercially available and laboratories are gaining experience making these measurements. Impulse peak insertion loss data will be presented for a variety of hearing protector types along with a description of the measurement procedure.

11:30

3aNS10. Real-ear attenuation of custom-fit earplugs with the communications earplug (CEP). Elmaree Gordon and Efrem R. Reeves (USAARL, Acoust. Branch, 6901 Farrel Rd., P.O. Box 620577, Fort Rucker, AL 36362-0577, elmaree.gordon@us.army.mil)

Hearing loss is one of the most common occupational injuries in the Department of Defense. One reason is limited access to adequate hearing protection during combat operations. Even if traditional pre-formed foam earplugs are used during combat, approximately 15% of the military population remains unprotected because the pre-formed earplugs do not adequately fit extremely small ear canals, extremely large ear canals, or ear canals with sharp turns. Custom-fit earplugs provide a potential solution for this hard-to-fit population. However, producing traditional wax-dipped custom-fit earplugs in a combat environment is not always practical. Fortunately, methodology exists to scan ear impressions to create a digital data set, transmit this data to a remote manufacturer via the Internet, and fabricate a set of custom-fit earplugs. The purpose of this study was to investigate how the attenuation provided by custom-fit earplugs created using the digital scanning technique compares to the attenuation provided by custom-fit earplugs created using the traditional wax-dipped technique when used with the CEP. Results show the digitally scanned custom-fit earplugs provide significantly poorer attenuation than the traditional wax-dipped custom-fit earplugs. Comply Canal Tip foam earplugs were also evaluated and shown to provide significantly greater attenuation than custom-fit earplugs manufactured by either method.

11:45

3aNS11. Noise surveys and hearing protection at the Royal Canadian Mint plant. (Jimmy) Jianping Yan (Corporate Eng., Royal Canadian Mint, 320 Sussex Dr., Ottawa, ON, K1A 0G8, Canada)

Royal Canadian Mint is a Crown Corporation to produce circulation and non-circulation coins for Canada and other countries. Noise measurements have been conducted regularly for production environment and before any new equipment operation in Royal Canadian Mint, and to protect our employee from hearing loss is one of our health and safety issues applied in whole plant. The formula used in the noise measurement is introduced in this paper, and the noise calculations are also discussed for the continuous noise sources, intermittent noise sources, and the combinations of individual noise sources. Some noise survey results are listed and analyzed, and the related hearing protective methods in our plant are also described.

WEDNESDAY MORNING, 2 NOVEMBER 2011 ROYAL PALM 3/4, 7:45 A.M. TO 12:00 NOON

Session 3aPA

Physical Acoustics: Theoretical and Computational Advances

Bonnie Schnitta, Chair
SoundSense, LLC, 46 Newtown Ln., Ste. 1, East Hampton, NY 11937

Contributed Papers

7:45

3aPA1. Formulation and applications of an integral-equation approach for solving scattering problems involving an object consisting of a set of piecewise homogeneous material regions. Elizabeth Bleszynski, Marek Bleszynski, and Thomas Jaroszewicz (Monopole Res., 739 Calle Sequoia, Thousand Oaks, CA 91360)

The paper presents the formulation and selected applications of the surface integral equation approach for finding pressure, displacement, and traction fields in a complex object consisting of a set of piece-wise homogeneous regions characterized by different Lame' material parameters. Representative applications of the approach are presented, which involve finding pressure field distribution inside human inner ear treated as an inclusion embedded in the surrounding inhomogeneous material. (the embedding region is treated with volumetric integral equations with suitable coupling to the inclusion). The method uses a set of coupled elastodynamics integral equations for two unknown fields: displacement and traction at each interface, which allow us to find displacement and traction field distribution inside a general complex object composed of sub-regions with different material parameters. The matrix elements of strongly singular kernels appearing in the integral equations were converted, through suitable integration by parts, to equivalent forms involving weakly singular integrands having at most 1/r singularity. Applications of the method are presented in simulation of acoustic and elastic fields in a human head and in the cochlea region. [This work is supported by AFOSR.]

8:00

3aPA2. Analysis and extension of scattering from rigid infinite wedge. Ambika Bhatta, Timothy Pflanz, Charles Thompson, and Kavitha Chandra (Dept. of Elec. and Comput. Eng., Univ. of Massachusetts Lowell, 1 Univ. Ave., Lowell, MA 01854)

A new representation of the scattered pressure of a point source due to infinite rigid wedge is derived by simplifying the Bessel functions in the modal solution and by implementing the image based representation of the source. The solution comprises of six impulses contributing to the reflected and transmitted pressure and an integral for the diffracted pressure. The inverse Fourier transform of the solution yields the time domain pressure and the asymptotic solution for higher frequency is evaluated and compared with solution obtained by steepest descent method. The analytical approach based on wedge diffraction is extended to evaluate the scattered pressure from finite rigid polygon with three vertices and edges. The computational results for the scattered pressure will be explained in detail.

8:15

3aPA3. Acoustic scattering strength of turbulence generated waves in a shallow water flow. V. Kirill Horoshenkov, Andrew Nichols, J. Simon Tait (School of Eng., Univ. of Bradford, Bradford, West Yorkshire, BD7 1DP, United Kingdom), and A. German Maximov (N. N. Andreyev Acoust. Inst., Moscow, Moscow 117 036, Russian Federation)

This work examines an airborne method of ultrasonic characterization of the dynamically rough water surface in an shallow water flow. A new experimental setup has been developed and tested in a 12 m long, 450 mm wide laboratory flume. This equipment includes a source of a 45 kHz airborne wave, microphone array, and an array of wave probes. It has been shown experimentally and theoretically that the directivity of the ultrasonic wave scattered by the dynamically rough water surface can be related directly to the roughness correlation function which can be measured non-acoustically. It is shown that the form of the roughness correlation function is determined unambiguously by the scale and intensity of the turbulence processes in the flow. The proposed acoustic method is accurate, remote, and rapid. It can be used to relate the spectral and statistical characteristics of the rough flow surface to the scale and intensity of the turbulent flow structures which cause the water surface to appear rough.

8:30

3aPA4. Distributed point source method and its applications in solving acoustic wave scattering problems. Tribikram Kundu, Ehsan Kabiri Rahani, and Talieh Hajzargerbashi (Dept. of Civil Eng. Eng. Mech., Univ. of Arizona, 1209 E. 2nd St., Tucson, AZ 85721)

A recently developed semi-analytical technique called distributed point source method is used for solving various wave scattering problems. Scattering of focused ultrasonic fields by air bubbles or cavities in fluid and solid media is investigated in this presentation. Results for both single and multiple cavity geometries are presented. It is investigated when two cavities in close proximity can be distinguished and when it is not possible. The interaction effect between two cavities prohibits simple linear superposition of single cavity solutions to obtain the solution for the two cavities placed in close proximity. Therefore, although some analytical and semi-analytical solutions are available for the single cavity in a focused ultrasonic field, those solutions cannot be simply superimposed for solving the two-cavity problem even in a linear elastic material. The comparison between the ultrasonic energies reflected from two small cavities versus a single big cavity is also investigated. Detection and characterization of cavities is important for both materials science and medical applications because air bubbles in molten metal as well as in blood must be first detected and characterized before taking necessary actions.

8:45

3aPA5. Effective medium theory applied to bubble scattering well below resonant frequency. Dale I. McElhone, Robert W. Smith, and R. Lee Culver (Appl. Res. Lab., Penn State Univ., P.O. Box 30, State College, PA 16804)

When ensonifying bubble assemblages using acoustic frequencies well below the resonant frequencies of the individual bubbles, resonant scattering theory for individual bubbles would suggest very low backscatter, whereas an "effective medium theory" predicts a result for backscattering that is orders of magnitude higher. For this case, a number of authors have suggested resonant scattering associated with the size of the bubble clusters, and backscatter measurements from bubble clouds have supported this hypothesis. Scattering can also be considered from a much larger bubble structure: a ship wake. At the stern, ship wakes contain a wide range of bubble sizes. Buoyancy quickly drives the largest bubbles to the surface, but small bubbles can persist for large distances. Ensonification at mid frequencies (1 kHz–10 kHz) therefore cannot rely on resonant bubble scattering. Using a generic wake model developed from the literature, acoustic backscatter can be modeled at frequencies below those of the resonant bubbles. The approach discretizes the spatial bubble distribution and sums the backscatter coherently, following the work of Commander and Prosperetti [JASA (1989)]. Ongoing work includes a lab-scale comparison of the backscattering approach applied to the ship wake model. Work sponsored by the Office of Naval Research, Code 321US.

9:00

3aPA6. Image reconstruction using singular-value decomposition of scattering operators. W. Jiang, J. Astheimer (Dept. of ECE, Univ. of Rochester, NY 14627, wejiang@ece.rochester.edu), and R. Waag (Dept. of ECE, Dept. of Imaging Sci., Univ. of Rochester, NY 14627)

Efficient inverse scattering algorithms for nonradial lossy objects are presented using singular-value decomposition to form reduced rank representations of the scattering operator. These algorithms extend eigenfunction methods that are not applicable to nonradial lossy scattering objects because the scattering operators for these objects are not normal. A method of local reconstruction by segregation of scattering contributions from different local regions is also presented. Scattering from each region is isolated by forming a reduced rank representation of the scattering operator that has domain and range spaces comprised of far-field patterns with retransmitted fields that focus on the local region. Accurate methods for the estimation of the boundary, average sound speed, and average attenuation slope of the scattering object are also given. These methods yield initial approximations of scattering objects that reduce the number of distorted Born iterations needed for high quality reconstruction. Calculated scattering from two lossy elliptical objects is used to evaluate the performance of the proposed methods. In both cases, the reconstruction procedures result in high-quality quantitative images of tissue parameters with sub-wavelength resolution.

9:15

3aPA7. Observations of Biot compressional waves propagating in snow. Donald G. Albert, Stephen N. Decato, and Frank E. Perron, Jr. (ERDC-CRREL, Cold Regions Res. and Eng. Lab., 72 Lyme Rd., Hanover, NH 03755)

Many laboratory measurements of compressional waves propagating in a porous material have been conducted, but similar measurements are less frequent outdoors. Here, we present velocity measurements of Biot compressional waves of the first and second kind propagating through a thick natural snow cover. Vertical sledge hammer blows and blank pistol shots were used to generate horizontally propagating pulses that were recorded by acoustic and seismic sensors. For both sources, seismic P1 compressional, shear, and Rayleigh waves are visible on the measured time series, followed by the airborne acoustic arrival. The acoustic arrival generates additional P1 ice frame waves and Biot P2 pore waves that propagate downward from the snow surface at measured velocities of 460 and 95 m/s, respectively. Theoretical calculations for a rigid porous medium show that the Biot P2 wave can account for the distortion and attenuation observed in the atmospheric acoustic wave. Separate small-scale laboratory measurements demonstrate that the P2 wave in snow is dispersed (the velocity increases with increasing

frequency) in agreement with the trend predicted by Biot's theory, but with some parameter adjustments needed to obtain the best agreement. [Work funded by U.S. Army basic research program.]

9:30

3aPA8. Refraction of the cyclonic microbarom signal by the cyclonic winds. P. Blom, R. Waxler, W. G. Frazier, C. Talmadge, and C. Hetzer (NCPA, Univ. of Mississippi, University, MS 38677, psblom@olemiss.edu)

Non-linear interaction of the ocean surface and atmosphere is known to produce narrow-band, low frequency, continuous acoustic and seismic radiation termed microbaroms and microseisms, respectively. The microbarom signal typically has an amplitude of a few microbars and a peak at 0.2 Hz. The microbaroms are generated by counter-propagating surface waves of equal period. The microbarom source location associated with a hurricane is believed to be due to the interaction of the waves produced by the cyclonic winds with the background ocean wave field and is generally located many kilometers from the eye of the storm, along a perpendicular to the direction of the ambient winds. Following up on a suggestion of Bedard and co-workers, propagation of the microbarom signal through the storm wind structure has been investigated using geometric acoustics in an inhomogeneous moving medium. Strong refraction of the signal is predicted. To observe this refraction we have deployed infrasound arrays along the US eastern seaboard. Predicted and measured back azimuths for propagation through the wind structure have been compared to the data recorded during the 2010 and 2011 Atlantic hurricane seasons.

9:45

3aPA9. Numerical examination of the impact of random terrain elevation and impedance variations on sound-field coherence. D. Ketih Wilson, Santosh Parakkal, Sergey N. Vecherin (U.S. Army Engineer Res. and Dev. Lab, 72 Lyme Rd., Hanover, NH 03755-1290), and Vladimir E. Ostashev (Univ. of Colorado, 325 Broadway, Boulder, CO 80305)

Like turbulent fluctuations in the atmosphere, variations in terrain elevations (surface roughness) and ground impedance may be regarded as diminishing the coherence of the sound field and introducing uncertainty into predictions. Here we examine, using a numerical approach, the characteristics and relative importance of terrain-elevation, ground-impedance, and turbulent variations on sound-field coherence. The calculations utilize parabolic equation solutions for a refractive atmosphere, with a Beilis–Tappert transform to incorporate terrain elevation variations and a phase-screen method for the turbulence. The relative importance of terrain variability and turbulence is studied for different refractive conditions. In particular, we examine whether surface roughness has a more significant impact on coherence in downward or upward refracting conditions. Interdependencies between the various mechanisms of coherence degradation are also investigated. The numerical calculations are compared to a recently developed theory for waveguide propagation conditions, which predicts existence of an effective spectrum with independent, additive contributions for turbulence and roughness. Limitations of the Markov approximation for modeling the elevation and impedance variations are also examined.

10:00–10:15 Break

10:15

3aPA10. Detection of turbulence aloft by infrasonic wind noise measurements on the ground. Jericho Cain and Richard Raspet (Jamie Whitten Natl. Ctr. for Physical Acoust., Univ. of Mississippi, 1 Coliseum Dr., University, MS 38677)

Wind turbines can be damaged by the inflow of high amplitude wind turbulence. The goal of this project is to determine if potentially damaging turbulent structures at hub height can be detected by infrasonic wind noise measurements at the ground. A large eddy simulation is used to model the atmospheric turbulence structures of interest and the resulting pressure fluctuations on the ground. Preliminary studies of the relationship between the time development of the pressure structures on the ground and the velocity fluctuations in the atmosphere will be described. If elevated turbulent structures can be detected on the ground, the predictions will be used to design an optimized pressure sensing array for experimental tests.

10:30

3aPA11. A linearized contrast source method for full-wave modeling of nonlinear acoustic wave fields in homogeneous media. Martin D. Verweij, Libertario Demi, and Koen W. A. van Dongen (Lab. of Acoust. Imaging and Sound Control, Fac. of Appl. Sci., Delft Univ. of Technol., Lorentzweg 1, 2628 CJ Delft, The Netherlands, m.d.verweij@tudelft.nl)

Many novel medical echography modalities, like super harmonic imaging (SHI), employ imaging of higher harmonics arising from nonlinear propagation. Design optimization of dedicated imaging probes for these modalities requires accurate computation of the higher harmonic beam profiles. The existing iterative nonlinear contrast source method can perform such simulations. With this method, the nonlinear term of the Westervelt equation is considered to represent a distributed contrast source in a linear background medium. The full transient nonlinear acoustic wave field follows from the Neumann iterative solution of the corresponding integral equation. Each iteration involves the spatiotemporal convolution of the background Green's function with an estimate of the contrast source, and appropriate filtering enables a discretization approaching two points per smallest wavelength or period. To further reduce the computational load and to anticipate extension of the method, in this presentation, the effect of linearizing the nonlinear contrast source is investigated. This enables the replacement of the Neumann scheme by more efficient schemes. Numerical simulations show that for design purposes the effect of linearization on the harmonic components involved with SHI remains sufficiently small. Moreover, it is shown that a significant decrease in computational costs may be achieved by using a Bi-CGSTAB scheme.

10:45

3aPA12. Angular spectrum representations of focused acoustic beams and a comparison with the complex source point description. Philip L. Marston (Phys. and Astronomy Dept., Washington State Univ., Pullman, WA 99164-2814)

One approach to describing the wave fields of focused acoustic beams is to use an angular spectrum representation. This has the advantage of facilitating a simple evaluation of the scattering by a sphere placed at the focal point of the beam in terms of the scattering by a superposition of Bessel beam components. This was recently demonstrated for a quasi-Gaussian beam [Marston, J. Acoust. Soc. Am. **129**, 1773-1782 (2011)]. This approach also has the advantage of separating the propagating spectral components from the evanescent spectral components [Marston, J. Acoust. Soc. Am. (submitted)]. While, for most practical acoustic sources, the evanescent spectral components may be neglected, the spectral representation is helpful for explaining why in principle evanescent spectral components can be important in the production of extremely tightly focused beams. A tightly focused beam with a Gaussian profile at the focal plane is not as easily describable using the more widely investigated complex source point description of the focused wave field. In addition to a Gaussian beam, the propagating-component angular spectrum representation is easily applied to the lowest radial-order Laguerre-Gauss helicoidal acoustic beam in the usual case where the beam is not tightly focused. [Work supported by ONR.]

11:00

3aPA13. Attenuation of elastic waves due to wave-induced vorticity diffusion in porous media. Tobias M. Mueller (CSIRO Earth Sci. and Resource Eng., 26 Dick Perry Ave., Kensington, 6151 WA, Australia, tobias.mueller@csiro.au) and Pratap N. Sahay (CICESE, 22860 Baja California, Mexico)

A theory for attenuation of elastic waves due to wave-induced vorticity diffusion in the presence of randomly correlated pore-scale heterogeneities in porous media is developed. It is shown that the vorticity field is associated with a viscous wave in the pore space, the so-called slow shear wave. The latter is linked to the porous medium acoustics through incorporation of the fluid strain rate tensor of a Newtonian fluid in the poroelastic constitutive relations. The method of statistical smoothing in random media is used to derive dynamic-equivalent elastic wave numbers accounting for the conversion scattering process into the slow shear wave. The result is a model for wave attenuation and dispersion associated with the transition from viscosity- to inertia-dominated flow regime in porous media. It is also shown

that the momentum flux transfer from the slow compressional into the slow shear wave is a proxy for the dynamic permeability in porous media. A dynamic permeability model is constructed that consists of an integral over the covariance function of the random pore-scale heterogeneities modulated by the slow shear wave. In a smooth pore-throat limit, the results reproduce the dynamic permeability model proposed by Johnson *et al*. [J. Fluid Mech. **176**, 379 (1987).]

11:15

3aPA14. Application of the Beilis and Tappert parabolic equation to long-range sound propagation over irregular terrain. Santosh Parakkal, Kenneth E. Gilbert, and Xiao Di (Natl. Ctr. for Physical Acoust., Univ. of Mississippi, University, MS 38677)

The Beilis and Tappert parabolic equation method is studied in the applications of sound propagation over irregular terrain. The exact ground impedance condition for porous ground is derived and applied to propagation over hills with slopes from 5° to 22°. It is found that for slopes less than approximately 20°; the flat-ground impedance condition is sufficiently accurate. For slopes greater than about 20°, the limiting factor on numerical accuracy becomes the narrow-angle approximation used in the Beilis and Tappert method. The generalized polar coordinate parabolic equation method (Polar-PE method) was developed partly to provide a very solid numerical benchmark to the Beilis and Tappert PE method.

11:30

3aPA15. A linearized contrast source method for full-wave modeling of nonlinear acoustic wave fields in media with strong and inhomogeneous attenuation. Libertario Demi, Martin D. Verweij, and Koen W. A. van Dongen (Lab. of Acoust. Imaging and Sound Control, Fac. of Appl. Sci., Delft Univ. of Technol., Lorentzweg 1, 2628 CJ Delft, The Netherlands, l.demi@tudelft.nl)

The iterative nonlinear contrast source (INCS) method is a numerical method originally developed for modeling transient nonlinear acoustic wave fields in homogeneous (i.e., spatially independent) lossless or lossy media. This method is based on the Westervelt equation and assumes that the nonlinear term describes a distributed contrast source in a linear background medium. Next, the equivalent nonlinear integral equation is iteratively solved by means of a Neumann scheme. In medical diagnostic ultrasound, the observed acoustic nonlinearity is weak, resulting in a weak nonlinear contrast source and a fast convergence of the scheme. To model spatially dependent attenuation, the Westervelt equation has recently been extended with an attenuative contrast source. This presentation deals with the situation of strong local attenuation. In that case, the attenuative contrast source is large and the Neumann scheme may fail to converge. By linearizing the nonlinear contrast source, the integral equation becomes linear and can be solved with more robust schemes, e.g., Bi-CGSTAB. Numerical simulations show that this approach yields convergent results for nonlinear acoustic fields in realistic media with strong local attenuation. Moreover, it is shown that in practical situations, the error due to the linearization remains sufficiently small for the first few harmonics.

11:45

3aPA16. A full wave method for modeling nonlinear acoustic wave fields in media with inhomogeneous wave speed, attenuation, and parameter of nonlinearity. Koen W. A. van Dongen, Libertario Demi, and Martin D. Verweij (Lab. of Acoust. Imaging and Sound Control, Fac. of Appl. Sci., Delft Univ. of Technol., Lorentzweg 1, 2628 CJ Delft, The Netherlands, k.w.a.vandongen@tudelft.nl)

The iterative nonlinear contrast source (INCS) method is a numerical method that has originally been developed for modeling transient nonlinear acoustic wave fields in homogeneous (i.e., spatially independent) lossless or lossy media. Starting from the Westervelt equation, the INCS method considers the nonlinear term of the latter as a nonlinear contrast source in an otherwise linear wave equation and iteratively solves the corresponding nonlinear integral equation using a Neumann scheme. By adding an attenuative contrast source, the method has recently been extended to deal with spatially dependent attenuation, and a spatially dependent parameter of nonlinearity has also been introduced. This presentation is about the introduction of an additional contrast source that account for a spatially dependent wave speed. In this case, convergence problems arise with the Neumann scheme. This problem is solved by replacing the Neumann scheme with a conjugate gradient scheme in which the error functional is based on the complete nonlinear integral equation. Results show that this approach yields very accurate results for the nonlinear wave field (tested up to the fifth harmonic) in media with a spatially dependent wave speed, attenuation, and parameter of nonlinearity, as encountered in realistic medical ultrasound applications.

Session 3aSCa

Speech Communication: The Voice Source: From Physiology to Acoustics

Abeer Alwan, Cochair
Dept. of Electrical Engineering, Univ. of California, Los Angeles, 405 Hilgard Ave., Los Angeles, CA 90095

Patricia A. Keating, Cochair
Dept. of Linguistics, Univ. of California, Los Angeles, Los Angeles, CA 90095-1543

Jody E. Kreiman, Cochair
Div. of Head and Neck Surgery, Univ. of California, Los Angeles, 1000 Veteran Ave., Los Angeles, CA 90095-1794

Chair's Introduction—8:00

Invited Papers

8:05

3aSCa1. Use of laryngeal high-speed videoendoscopy systems to study voice production mechanisms in human subjects. Daryush D. Mehta (Ctr. for Laryngeal Surgery and Voice Rehabilitation, Massachusetts General Hospital, One Bowdoin Sq., 11th Flr., Boston, MA 02114, daryush.mehta@alum.mit.edu), Matías Zañartu (Univ. Técnica Federico Santa María, Valparaíso, Chile), Thomas F. Quatieri (Massachusetts Inst. of Technol., Lexington, MA 02421), Dimitar D. Deliyski (Cincinnati Children's Hospital Med. Ctr., Cincinnati, OH 45229), and Robert E. Hillman (Voice Rehabilitation & Massachusetts General Hospital, Boston, MA 02114)

Advances in laryngeal high-speed videoendoscopy (HSV) are making it possible to investigate critical relationships between vocal fold physiology and acoustic voice production in human subjects. Our group has developed HSV systems for clinical research with synchronous acquisition of acoustics, electroglottography, neck skin acceleration, and, in the most comprehensive setup, airflow and intraoral pressure. Key results will be presented from examinations of source-filter coupling and studies of the acoustic impact of vocal fold vibratory asymmetry in subjects with and without voice disorders. Findings hold potential clinical significance by revealing acoustic-HSV relationships not observable using standard stroboscopic imaging, as well as contributing to the direct evaluation and eventual improvement of voice production models. The work of T. F. Quatieri was supported by the Department of Defense under Air Force contract FA8721-05-C-0002. The work of other authors was supported by grants from the NIH National Institute on Deafness and Other Communication Disorders (T32 DC00038 and R01 DC007640) and by the Institute of Laryngology and Voice Restoration.

8:25

3aSCa2. From high-speed imaging to perception: In search of a perceptually relevant voice source model. Abeer Alwan, Patricia Keating, and Jody Kreiman (Elec. Eng., UCLA, 405 Hilgard Ave., Los Angeles, CA 90095, alwan@ee.ucla.edu)

The goal of this project is to develop a new source model from high-speed recordings of vocal fold vibrations and simultaneous audio recordings. By analyzing area waveforms and acoustics and performing perception experiments, we will better parameterize the new source model and also uncover those aspects of the model that are perceptually salient, leading to improved TTS systems. We recorded acoustic and high-speed video signals from eight speakers producing the vowel /i/. Each speaker produced four phonation types (breathy, modal, pressed, and creaky) and three pitches (low, normal, and high). Glottal area waveforms were extracted from the high-speed recordings, semi-automatically, on a frame-by-frame basis. The acoustic results indicated that phonation types are differentiated by a number of spectral and noise measures, including H1*-H2* and the harmonics-to-noise ratio. In addition, the different pitch levels were responsible for changes in quality within each phonation type, but such changes were subject to inter-speaker variability. Perceptual results highlight the importance of lower-frequency harmonics in voice quality perception. These studies are critical to our computational modeling efforts, because they help us understand which aspects of the source model are perceptually important. [Work supported in part by NSF and NIH.]

8:45

3aSCa3. The affect of epilarynx tube dimensions on glottal airflow. Ingo R. Titze (Natl. Ctr. for Voice and Speech, 136 South Main St., Ste. 320, Salt Lake City, UT 84101-3306)

Since the early 1980s, it has been known that an inertive supraglottal acoustic load skews the peak of the glottal flow pulse. It also lowers the phonation threshold pressure. The skewing in turn affects the vocal intensity by increasing the rate of flow declination. High frequencies in the source spectrum are emphasized. More recent is the discovery that the airway between the vocal folds and the aryepiglottic folds, often called the epilarynx tube, provides much of the inertance in a vowel production. This presentation shows how epilarynx tube geometry skews the glottal flow pulse and thereby may affect voice quality.

9:05

3aSCa4. An *ex vivo* perfused human larynx for studies of human phonation. Gerald S. Berke, Scott Howard, and Abie Mendelsohn (Head and Neck Surgery, UCLA School of Medicine, 62-132 CHS, Los Angeles, CA 90095-6214, gberke@mednet.ucla.edu)

For many years, non-human animal models and excised human larynges have been used to study the physiology and biomechanics of human laryngeal function. While these studies have provided a wealth of information, experimental data concerning actual human laryngeal function have been limited by the inability to parametrically study the complete set of variables that controls vocal fold function in a living human. This presentation describes the development of a perfused *ex vivo* human larynx. In this model, a human larynx is harvested at the time of organ donation and perfused with the donor's blood to maintain viability of its nerves and muscles, which can then be stimulated as in an *in vivo* preparation. This model allows parametric, experimental control of the larynx (including stimulation of the thyroarytenoid muscle) while it is still virtually living. Applicability of this new laryngeal model to studies of physiology, biomechanics, and to treatment of human disease will be discussed.

9:25

3aSCa5. Voice quality as a correlate of stop coda voicing in developing speech. Helen M. Hanson (Elec. and Comput. Eng. Dept., Union College, 807 Union St., Schenectady, NY 12308, hansonh@union.edu) and Stefanie Shattuck-Hufnagel (MIT, Cambridge, MA 02139)

Previous work reported that two children from the Imbrie Corpus (2005) tend to produce noise at the end of the nuclear vowel in words with [-voice] stop codas (duck, cup), but not in words with [+voice] codas (tub, bug) [JASA **123**, 3320]. This noise resembles the preaspiration reported for adult speech in some languages. In this paper the earlier work is extended to the remaining eight children in this corpus of speech from 2- to 3-year old children learning American English. Furthermore, for nine subjects we compare their behavior to that of their primary female caregiver. Many of the children produce unvoiced coda stops that are coded as being strongly preaspirated. Their caregivers also produce vowel-final noise preceding [-voice] stops, but these tend to be coded as being weakly preaspirated. For both child and adult subjects preaspiration is more likely to occur for /k/ than for /p/. A few subjects produce irregular voicing rather than noise before [-voice] coda stops. More detailed acoustic measures such as durations and relative intensities will be presented. In addition, the acoustics are used to infer the physiology behind these voice quality differences between children and adults. [Work support from NIH R01DC00075, R01HD057606.]

9:45

3aSCa6. Hemilarynx experiments: Statistical analysis of vibrational parameters over changing glottal conditions. M. Doellinger (Dept. for Phoniatrics and Pedaudiology, Univ. Hospital Erlangen, Bohlenplatz 21, 91054 Erlangen, Germany, michael.doellinger@uk-erlangen.de) and D. A. Berry (The Laryngeal Dynam. Lab., Div. of Head & Neck Surgery, UCLA School of Medicine, 31-24 Rehab Ctr., 1000 Veteran Ave., Los Angeles, CA 90095-1794)

In vitro (humans, animals) or *in vivo* (animals) hemilarynx experiments are still the only way to visualize and analyze 3-D vibrations of the entire medial surface of the vocal fold. However, to date, only a few studies have been performed. For three different larynges during sustained phonation, vibrational output was statistically analyzed as a function of glottal airflow (800 to 1800 ml/s), adductory forces (10, 50, 100 g), and two different anterior–posterior pre-stress conditions (10, 50 g). Using a homogeneous grid of 30 microsutures sewn along the medial surface of the vocal fold, global parameters (empirical eigenfunctions/EEF, f0, subglottal pressure, and sound intensity level) were computed. Similarly, local parameters (displacements, velocities, and accelerations) were analyzed at the 30 different suture locations. The recordings were obtained using a digital high-speed camera (2000 and 4000 fps). Increased airflow resulted in significant statistical changes in all parameter values except the empirical eigenfunctions. For increased adduction and a lower pre-stress (10 g), the local parameters increased more than for the higher pre-stress condition (50 g). For the two pre-stress conditions and an increased adductional force, most global parameters exhibited significant changes (f0, sound intensity, EEF2), while other global parameters (subglottal pressure) did not.

10:05–10:20 Break

10:20

3aSCa7. Electroglottographic and acoustic measures of phonation across languages. Patricia Keating and Jian-Jing Kuang (Dept. Linguist., UCLA, Los Angeles, CA 90095-1543, keating@humnet.ucla.edu)

Many less-commonly studied languages use phonation contrastively: vowels, and therefore words, can differ in phonation type. Most such languages contrast two or three phonation categories. In other languages, a lexical tone (contrastive pitch) may typically have a redundant non-modal phonation. We have collected electroglottographic and audio recordings from speakers of six languages with one or both of these uses of phonation (Bo, Gujarati, Hani, White Hmong, Mandarin, Yi) and analyzed them with EGGWORKS. EGGWORKS is a free program which automates several EGG measures and works together with VoiceSauce, a free program which automates many acoustic voice quality measures. Comparisons of the EGG and acoustic measures within and across languages, and their correlations, will be presented. We have also applied functional data analysis to EGG pulse shapes and will describe the relations of the factors underlying the shapes to the EGG and acoustic measures. [Work supported by NSF.]

10:40

3aSCa8. The two-source problem: Producing voiced obstruents. Christine H. Shadle (Haskins Labs., 300 George St., New Haven, CT 06511, shadle@haskins.yale.edu)

The relationship between the physiological parameters and the acoustic output of the voice source is complex, and even more so when voiced obstruents are considered. Standard models assume that the subglottal pressure is divided across the glottis and the supraglottal constriction, which explains both the weaker noise source of voiced compared to voiceless fricatives and the tendency of voiced stops and fricatives to devoice. However, such models do not take into account small adjustments in vocal fold posture that may, for example, decrease the phonation threshold pressure, or in the constriction that may act to optimize noise production. Larger adjustments are made for voiced fricatives, such as lowering the larynx and advancing the tongue root, that may also enhance a speaker's ability to produce both voice and noise sources at a given subglottal pressure. In this talk results from a wide range of studies will be considered to arrive at a more nuanced understanding of the interaction of two source mechanisms in voiced obstruents.

Contributed Papers

11:00

3aSCa9. The formation of flow separation vortices inside the glottis during vocal fold closing. Liran Oren, Sid Khosla, and Ephraim Gutmark (Dept. Otolaryngol., Univ. of Cincinnati, P.O. Box 670528, Cincinnati, OH 45267, orenl@mail.uc.edu)

Our previous work highly suggests that glottal airflow contains certain vortical structures, especially flow separation vortices (FSV) that significantly contribute to acoustics. We have previously identified these FSV directly above the folds but not between the folds. We have now developed a technique that allows us to see both intraglottal velocity fields and the medial wall of the folds during vocal fold closing. In this study, we used the new technique in an excised canine larynx to simultaneously measure intraglottal velocity fields (using high speed particle image velocimetry) and the medial surface of the folds. We observed the formation of FSV inside the glottis. Intraglottal pressures were calculated from measured intraglottal velocity fields using the Navier–Stokes equations. The maximum negative intraglottal pressure on the medial surface of one fold was −6 cm H2O. The circulation of the transglottal flow was highly correlated to the maximum closing speed (MCS) (r=0.98, $p<0.001$), and had good correlation to the spectral slope (r=-0.8, $p<0.01$) and HNR (Harmonic to Noise Ratio) (r=0.85, $p<0.005$). These results substantiate our hypothesis and our new technique is significant because it will allow us to experimentally characterize the intraglottal flow-structure relationship during vocal fold closing.

11:15

3aSCa10. Simulation of the coupling between vocal-fold vibration and time-varying vocal tract. Yosuke Tanabe (Automotive Product Res. Lab., Hitachi America, Ltd., 34500 Grand River Ave., Farmington Hills, MI 48335, yosuke.tanabe@hitachi-automotive.us), Parham Mokhtari, Hironori Takemoto (Natl. Inst. of Information and Commun. Technol., Kyoto 619-0288, Japan), and Tatsuya Kitamura (Konan Univ., 8-9-1, Hyogo 658-8501, Japan)

Acoustic coupling between the voiced sound source and the time-varying acoustic load during phonation was simulated by combining the vocal-fold model of [S. Adachi *et al*., J. Acoust. Soc. Am **117**(5) (2005)] with the vocal-tract model of [P. Mokhtari *et al*., Speech Commun. **50**, 179–190 (2008)]. The combined simulation model enables to analyze the dynamic behavior of the vocal folds due to the time-varying shape of the vocal tract. An example of vocal-fold behavior is shown for sustained phonation of the Japanese vowel sequence "aiueo." [This research was partly supported by Kakenhi (Grant Nos. 21500184, 21300071).]

11:30

3aSCa11. Restraining mechanisms in regulating glottal closure during phonation. Zhaoyan Zhang (UCLA School of Medicine, 31-24 Rehab Ctr., 1000 Veteran Ave., Los Angeles, CA, 90095-1794)

Recent experimental studies showed that isotropic vocal fold models were often blown wide apart and thus not able to maintain adduction, resulting in voice production with noticeable breathy quality. This study showed that the capability of the vocal fold to resist deformation against airflow and maintain position can be improved by stiffening the anterior-posterior tension in either the body-layer or the cover-layer of the vocal folds, which presumably can be achieved through the contraction of the thyroarytenoid and cricothyroid muscles, respectively. Experiments in both physical models and excised larynges showed that, when these restraining mechanisms were activated, the vocal folds were better able to maintain effective adduction, resulting in voice production with much clearer quality and reduced breathiness. [Work supported by NIH.]

11:45

3aSCa12. The effects of age on speech production in a non-pathological system: A case study of 48 years. Eric J. Hunter, Wesley R. Brown, Patrick Pead, and Megan Engar (Natl. Cntr for Voice and Speech, Univ. of Utah, 136 S. Main St., #320, Salt Lake City, UT, 84101, eric.hunter@utah.edu)

The current study examines 36 recordings (mean: 30 min) of a male speaker, spanning the years 1958 to 2007. These recordings provide a rare opportunity to track a single individual's age-related voice and speech mechanism changes (48–98 y/o). Aging effects became noticeable between the ages of 68–74, indicating a fundamental change in the body's maintenance of the speech mechanism, implying the likely onset of other aging symptoms such as swallowing problems. Voice F_0, which usually begins to drop during puberty, continued to decrease for this individual until 70 y/o; from 70 to 98 y/o, average vocal F_0 then increased from 140 to 160 Hz. Rate of speech (syll/min) began to decrease precipitately at 78 y/o (175 to 165 per/min.), while length of speech breathing reduced from 70 y/o (35 to 25 phonemes per breath group). The results of this case study can be used as a baseline for future studies. The aging of the voice and speech mechanism affects breathing, swallowing, and communication. The world's 50+ population is the fastest growing segment, affecting society by its sheer number as well as by historically high life spans. Healthcare practitioners must understand and accommodate the needs of this population.

Session 3aSCb

Speech Communication: Production, Processing, Acquisition, and Hearing Among Typical and Atypical Populations (Poster Session)

Sam Tilsen, Chair
Univ. of Southern California, Grace Ford Salvatori 301, Los Angeles, CA 90089

Contributed Papers

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

3aSCb1. Speaker adaptation in infancy: The role of lexical knowledge. Marieke van Heugten and Elizabeth K. Johnson (Dept. of Psych., Univ. of Toronto, 3359 Mississauga Rd. N., Mississauga, ON L5L1C6, Canada, marieke.vanheugten@utoronto.ca)

The acoustic realization of words varies greatly between speakers. While adults easily adapt to speaker idiosyncrasies, infants do not possess mature signal-to-word mapping abilities. As a result, the variability in the speech signal has been claimed to impede their word recognition. This study examines whether exposure to a speaker may allow infants to better accommodate that speaker's accent. Using the Headturn Preference Procedure, 15-month-olds were presented with lists containing either familiar (e.g., ball) or unfamiliar words (e.g., bog). In experiment 1, these words were produced in infants' own accent (Canadian English); in experiment 2, they were produced in a foreign accent (Australian English). Comparable to previous work (Best *et al*., 2009), only infants presented with their own accent preferred to listen to familiar over unfamiliar words. Thus, without access to speaker characteristics, word recognition is limited to familiar accents. In experiment 3, the same Australian-accented stimuli were preceded by exposure to the Australian speaker. Speaker adaptation tended to correlate with the infants' vocabulary size, with greater vocabularies being indicative to more robust adaptation. We are currently testing whether vocabulary size is a mediating factor caused by general processing abilities or whether speaker adaptation is lexically driven.

3aSCb2. Social accountability influences phonetic alignment. Holger Mitterer (Max Planck Inst. for Psycholinguistics, P.O. Box 310, 6500 AH Nijmegen, The Netherlands, holger.mitterer@mpi.nl)

Speakers tend to take over the articulatory habits of their interlocutors [e.g., Pardo, JASA (2006)]. This phonetic alignment could be the consequence of either a social mechanism or a direct and automatic link between speech perception and production. The latter assumes that social variables should have little influence on phonetic alignment. To test that participants were engaged in a "cloze task" (i.e., Stimulus: "In fantasy movies, silver bullets are used to kill ..." Response: "werewolves") with either one or four interlocutors. Given findings with the Asch-conformity paradigm in social psychology, multiple consistent speakers should exert a stronger force on the participant to align. To control the speech style of the interlocutors, their questions and answers were pre-recorded in either a formal or a casual speech style. The stimuli's speech style was then manipulated between participants and was consistent throughout the experiment for a given participant. Surprisingly, participants aligned less with the speech style if there were multiple interlocutors. This may reflect a "diffusion of responsibility:" Participants may find it more important to align when they interact with only one person than with a larger group.

3aSCb3. The influence of socioindexical expectations on speech perception in noise. Kevin B. McGowan (Dept. of Linguist., Rice Univ., 6100 Main St., Houston, TX 77005, clunis@umich.edu)

Previous research has shown that listener percepts can be altered by the manipulation of listener expectations of speaker identity [e.g., Niedzielski (1999), Hay *et al*. (2006)]. This study examines the extent to which socioindexical information can be *useful* to listeners. English speakers with little or no Chinese experience and Heritage Mandarin English speakers transcribed Chinese-accented speech in noise. In a between-subjects design, listener expectations about speaker identity were manipulated by presenting an Asian face, a socioindexically uninformative silhouette, or a Caucasian face as the purported speaker of high and low predictability sentences. Consistent with previous findings, listeners presented with an Asian face were significantly more accurate transcribers than those presented with a Caucasian face. Intriguingly, the silhouette condition patterned with the Asian face for experienced listeners but with the Caucasian face for inexperienced listeners. Unexpectedly, inexperienced listeners, while overall less accurate than experienced, Heritage listeners, saw the same magnitude improvement with the Asian face. Furthermore, transcription errors are inconsistent with suggestions (e.g., Staum Casasanto, 2009; Johnson, 2006) that listeners will alter base activations of prelexical or lexical forms to accommodate an expected accent. Implications for theories of speech perception and word recognition will be explored.

3aSCb4. Attention modulates the time-course of talker-specificity effects in lexical retrieval. Rachel M. Theodore and Sheila E. Blumstein (Dept. of Cognit., Linguistic & Psychol. Sci., Brown Univ., P.O. Box 1821, Providence, RI 02912, rachel_theodore@brown.edu)

Listeners retain in memory phonetic characteristics associated with individual talkers. It has been proposed that talker-specificity effects, while robust, emerge late in processing, reflecting increased time to activate lower-frequency instantiations for individual talkers. Here we test the hypothesis that attention to talker characteristics modulates the time-course of talker-specificity effects. Two groups of listeners participated in encoding and recognition phases. During encoding, all listeners heard words produced by two talkers, but only half of them were required to identify talker gender. During recognition, all listeners heard a series of words and indicated whether each word had been presented during encoding. Critically, some of the words were previously presented in the same voice and some in a different voice. Results showed no difference in mean hit rate or reaction time between the two groups, indicating that the encoding manipulation did not globally affect recognition memory or response latency. However, the latency data showed an interaction such that a talker-specificity effect emerged only for listeners who identified talker during encoding. These results indicate that the talker-specificity effect reflects an attention shift, not increased processing time, consistent with a dynamic perceptual system where attention to specific aspects of the signal can influence representational frequency.

3aSCb5. Lexical-neighborhood and competing-task effects on word recognition and word recall by native and non-native listeners. Catherine L. Rogers, April M. Frenton, and Heather A. Hoefer (Dept. of Comm. Sci. and Dis., Univ. of South Florida, 4202 E. Fowler Ave., PCD1017, Tampa, FL 33620)

In spoken-word recognition, high-frequency words with few and less frequently occurring minimal-pair neighbors (lexically easy words) are recognized more accurately than low-frequency words with many and more frequently occurring neighbors (lexically hard words). This easy-hard word effect has been found to be larger for non-native listeners with a relatively late age of immersion in an English-speaking environment. Previous research found no effect of a competing digit-recall task on spoken-word recognition and no effect of listener group or listening environment on digit-recall accuracy. The present study compares word recognition by native English-speaking listeners and non-native listeners with either earlier (age 10 or earlier) or later (age 14 or later) ages of immersion in an English-speaking environment. Spoken word lists composed of equal numbers of lexically easy and hard target words were presented in an open-set word-identification task. Spoken words were presented in quiet and in moderate background noise and preceded by a list of zero, three, or six visually presented words, which listeners were asked to recall following the spoken word-recognition task (competing task). The size of the easy-hard word effect, spoken-word recognition accuracy competing-task word-recall accuracy, and word-entry response time will be compared across listener groups and listening conditions.

3aSCb6. Reading between the turns: Social perceptions of turn-taking cues in conversation. Marisa Tice and Tania Henetz (Dept. of Linguist., Margaret Jacks Hall, Bldg. 460, Stanford Univ., Stanford, CA 94305-2150, middyp@stanford.edu)

The perception of turn-taking cues provides an alluring new avenue for speech perception research since it requires complex cognitive and social processes that speakers must perform online during conversation. This study investigates whether inter-speaker gap duration, overlap, and backchanneling play a role in the social perception of speech during conversation. In a matched-guise task, participants heard clips of spontaneous, dyadic conversation. Clips were phonetically manipulated to control for the three turn-taking cues of interest by stretching and shrinking gap durations and replacing the naturally occurring backchanneling with silence. Participants then rated the speakers and conversations on a range of social measures, including dominance, formality, and similarity. Initial results indicate that listeners' social judgments are sensitive to these cues. For example, the removal of backchanneling in a male–male conversation resulted in ratings of higher speaker interest and lower speaker similarity, running contrary to some previous descriptions of backchannels as supportive signals (Yngve, 1970). This effect may interact with gender, such that backchanneling is interpreted differently in female-female or mixed-gender dyads, supporting Pearson *et al.*'s (2008) finding that turn-timing effects interact with (mis)matches in speaker race. It is clear these judgments follow from a complex interaction of subtle acoustic cues, social information, and context. [This work was supported by a National Science Foundation Graduate Research Fellowship to M. Tice.]

3aSCb7. Use of waveform mixing to synthesize a continuum of vowel nasality in natural speech. William F. Styler IV, Rebecca Scarborough, and Georgia Zellou (Dept. of Linguist., Univ. of Colorado, 295 UCB, Boulder, CO 80309-0295)

Studies of the perception of vowel nasality often use synthesized stimuli to produce controlled gradience in nasality. To investigate the perception of nasality in natural speech, a method was developed wherein vowels differing naturally in nasality (e.g., from CVC and NVN words) are mixed to yield tokens with various degrees of nasality. First, monosyllables (e.g., CVC, NVC, CVN, NVN) matched for vowel quality and consonant place of articulation were recorded. The vowels from two tokens were excised, matched for amplitude, duration and pitch contour, and then overlaid sample-by-sample according to a specified ratio. The resulting vowel was spliced back into the desired consonantal context. Iterating this process over a series of ratios produced natural-sounding tokens along a continuum of vowel nasality. Acoustic measurements of the nasality of output tokens [using A1-P0 (Chen, 1997)] confirmed a relation between the ratio used and the nasality of the output. Stimuli created in this manner were used in a perception experiment (Scarborough *et al.*, 2011) where degree of nasality affected perception without any slowdown in response time for nasality-altered vs. unaltered tokens. This demonstrates the quality of the nasal-altered stimuli and suggests the potential usefulness of this process for other speech perception studies.

3aSCb8. Formant frequencies, vowel identity, and the perceived relative tallness of synthetic speakers. Santiago Barreda and Terrance M. Nearey (Univ. of Alberta, 4-32 Assiniboia Hall, Edmonton, AB, Canada T6G 2E7, sbarreda@ualberta.ca)

Listeners can make consistent judgments regarding the tallness of speakers [Rendall *et al.*, J. Exp. Psych: Human Percep. Perform. **33**, 1208 (2007)]. These judgments are informed by the f0 and formant frequencies (FFs) of a speaker's voice. However, FFs are also cues to vowel identity, such that a small speaker producing an /u/ might have lower average FFs than a larger speaker producing an /æ/. Do listeners use absolute FFs to judge speaker tallness, or do they "correct" for phonetic identity and consider the FFs of a vowel relative to those expected for that category? To test this, a series of synthetic vowels (/i æ u/) with different FF scalings and different numbers of formants (2–5) were created. These scalings were intended to replicate speakers of different vocal tract lengths (i.e., sizes). Participants were presented with pairs of vowels and asked to indicate which vowel sounded like it had been produced by a taller speaker. Results indicate that listeners consider both absolute and phonetically "corrected" FF information and that formants higher than F3 greatly reduce listener's reliance on absolute F1 and F2 information in making speaker size judgments.

3aSCb9. Acoustic variability affects asymmetry in infant speech discrimination. Stephanie Archer (Dept. of Linguist., Univ. of Calgary, 2500 University Dr. NW, Calgary, AB T2E 0L7 Canada) and Suzanne L. Curtin (Univ. of Calgary, NW, Calgary, AB))

From birth, infants are capable of discriminating speech sounds that occur cross-linguistically, but by their first year, infants have difficulty discriminating most non-native contrasts (Werker and Tees, 1984). Yet, language experience is not the only factor affecting discrimination. Further research into language-specific perception reveals asymmetries occur in the discrimination of vowels according to perceptual space (Polka and Bohn, 2003) and consonants based on frequency in the input (Anderson *et al.*, 2003). It might be that acoustic variability also causes asymmetries in infant speech perception. Seventy-six 6- and 9-month olds participated in a discrimination task comparing bilabial and velar stop - /l/ onsets to unattested coronal stop - /l/ onsets (e.g. /kla/ - /tla/; /pla/ - /tla/) in both voiced and voiceless conditions. Infants successfully discriminated coronal from bilabial onsets ($p<0.05$), but not velar ($p>0.05$), with no effects of age or voicing. Adult productions of the attested clusters were subjected to an acoustic analysis. Of the four tested characteristics, higher standard deviations were found in velar onsets (/kla/ > /pla/: F2 onset and liquid duration; /gla/ > /bla/: F2 slope, VOT, and liquid duration). This suggests that acoustic variability in the input affects infants' speech discrimination.

3aSCb10. Adapting to foreign-accented speech: The role of delay in testing. Marijt J. Witteman, Neil P. Bardhan, Andrea Weber (MPI for Psycholinguistics, Wundtlaan 1, 6500AH Nijmegen, The Netherlands, marijt.witteman@mpi.nl), and James M. McQueen (Behavioural Sci. Inst., Radboud Univ. Nijmegen, 6500HE, The Netherlands)

Understanding speech usually seems easy, but it can become noticeably harder when the speaker has a foreign accent. This is because foreign accents add considerable variation to speech. Research on foreign-accented speech shows that participants are able to adapt quickly to this type of variation. Less is known, however, about longer-term maintenance of adaptation. The current study focused on long-term adaptation by exposing native listeners to foreign-accented speech on Day 1, and testing them on comprehension of the accent one day later. Comprehension was thus not tested immediately, but only after a 24 h period. On Day 1, native Dutch listeners listened to the speech of a Hebrew learner of Dutch while performing a phoneme monitoring task that did not depend on the talker's accent. In particular, shortening of the long vowel /i/ into /ɪ/ (e.g., lief [li:f], 'sweet', pronounced as [lɪf]) was examined. These mispronunciations did not create lexical ambiguities in Dutch. On Day 2, listeners participated in a cross-modal priming task to test their comprehension of the accent. The results will be contrasted with results from an experiment without delayed testing and related to accounts of how listeners maintain adaptation to foreign-accented speech.

3aSCb11. The interrelation between perceived and produced vowel spaces in Brazilian and European Portuguese. Katerina Chladkova (Amsterdam Ctr. for Lang. and Commun., Univ. of Amsterdam, Spuistraat 210, 1012VT Amsterdam, The Netherlands, k.chladkova@uva.nl) and Paola Escudero (MARCS Auditory Labs., Univ. of Western Sydney, Milperra, NSW 2214, Australia)

Previous research has shown that the interrelation between the perception and production of vowels is not straightforward. For instance, it has been shown that the vowel spaces of Spanish and Czech may have different structures in production than in perception (Boersma and Chladkova, 2011). This study compares vowel perception and production of 40 speakers in Brazilian- (BP) and European Portuguese (EP). Productions were elicited in a sentence-reading task and were reported in Escudero *et al.* (2009). Perception was tested by means of a phoneme identification task with tokens sampled from the whole vowel space. Perceived and produced vowel spaces were compared through the location of vowel categories (the vowels' F1 and F2 averaged across speakers/listeners) and their boundaries with neighboring categories. The results show that the perceived vowel space of BP listeners differs from that of EP listeners. Specifically, low-mid vowels have a higher F1 in BP than in EP, a finding that is in line with BP and EP vowel production differences. However, BP low-mid vowels are less peripheral than EP vowels along the F2 dimension only in perception. Other dialectal differences as well as the degree of correspondence between the production and perception of Portuguese vowels are discussed.

3aSCb12. Phonetic imitation by school-age children. Kuniko Nielsen (Dept. of Linguist., Oakland Univ., Rochester, MI 48309, nielsen@oakland.edu)

Phonological representations such as phoneme and feature are often assumed in prevailing linguistic theories. However, little is known about how these representations are formed during the course of language acquisition. The current study aims to investigate developmental changes of phonological representations by examining how children imitate fine phonetic details of recently heard speech. Previous studies have shown that adult speakers implicitly imitate the phonetic properties of recently heard speech (e.g., Goldinger, 1998; Pardo, 2009). Recently, Nielsen (2011) showed that phonetic imitation was generalized at phoneme and feature levels, providing support for sub-lexical representations. The current study extends these findings and examines whether school-age children manifest similar patterns of phonetic imitation. The experiment employed the imitation paradigm in which participants' speech is compared before and after they are exposed to model speech with extended VOT on the target phoneme /p/. A preliminary analysis of data revealed that participants produced longer VOTs after being exposed to model speech with extended VOTs. The change was generalized to new instances of the target phoneme /p/ and the new phoneme /k/, indicating that sub-lexical units were also involved in phonetic imitation by children.

3aSCb13. Classification of foreign accents. Eriko Atagi and Tessa Bent (Dept. of Speech and Hearing Sci., Indiana Univ., 200 S. Jordan Ave., Bloominton, IN 47405, eatagi@indiana.edu)

Regional dialects and foreign accents account for a significant source of between-talker variability. The auditory free classification task—in which listeners freely group talkers based on audio samples—has been a useful tool for examining listeners' cognitive representations of regional dialects (e.g., Clopper and Pisoni, 2007) and was employed here to examine perceptual representation of foreign accents. This task assesses listeners' perception of variation without imposing category structure. In the present study, native listeners completed four free classification tasks in which they grouped 24 talkers from six native language backgrounds by perceived native language. The four tasks varied in overall sentence intelligibility and whether four native talkers were also included, to explore how accent level and presence of native language exemplars influence classification. Categorization performance was less accurate with lower intelligibility sentences, even when sentence content was known, but more accurate when native talkers were included. Thus, listeners attend to phonological features that characterize differences across foreign accents, and are more attentive to these features when accented and native speech can be directly compared. Moreover, perception of lower intelligibility sentences may impose a higher cognitive load, thus restricting the cognitive resources available for classification. [Work supported by NIDCD R21DC010027.]

3aSCb14. Does second language phonological acquisition affect native language word recognition? Melinda D. Woodley (Dept. of Linguist., Univ. of California, Berkeley, 1203 Dwinelle Hall #2650, Berkeley, CA 94720-2650, melinda.woodley@berkeley.edu), Marisa Tice, and Meghan Sumner (Stanford Univ., Stanford, CA 94305-2150)

Research with monolingual adults has shown that words whose initial stop consonants have canonical VOTs show larger semantic priming effects than words beginning with phonetically non-canonical stops; the prime [k^h] (long-lag) facilitates recognition of a semantically related target (*queen*) more than [k] (short-lag), though both facilitate target recognition (Andruski *et al.*, 1994; van Alphen and McQueen, 2006). The present study asks how the acquisition of a second (L2) phonological system affects word recognition in the native language (L1). A longitudinal semantic priming study ([k]... *queen*) is being administered to American students in Paris at the beginning, middle, and end of their 4–6 weeks of immersion class enrollment. Two potential patterns may emerge: (1) exposure to instances of French short-lag /k/ may lead to increased facilitation of short-lag [k] primes, indicating an early established link between shared L1 and L2 phonological categories or (2) since stops in the short-lag VOT range are categorized differently in French versus English ([k] = /k/ vs. /g/), increased ambiguity introduced by the L2 system may incur a greater cost for non-canonical variants, causing [k] to become a less effective prime. Both findings have implications for theories of speech perception and representation.

3aSCb15. When do which sounds tell you who says what? A phonetic investigation of the familiar talker advantage in word recognition. Stephen J. Winters (Dept. of Linguist., Univ. of Calgary, SS 820, 2500 University Dr. NW, Calgary, AB T2N 1N4, Canada, swinters@ucalgary.ca)

This study investigated whether voice quality, which is a highly salient cue to talker identity, facilitates the process of word recognition for familiar talkers. Two groups of listeners were trained to identify the same set of talkers. One group of listeners heard these talkers speaking in a variety of voice qualities (modal, creaky, and breathy), while another group of listeners heard each talker speaking in only one voice quality. Both groups of listeners were then tested on their ability to identify words spoken by these familiar talkers—in a variety of voice qualities—and also words spoken by a group of unfamiliar talkers. Results showed that voice quality had a significant effect on word recognition scores. Both groups also exhibited better word recognition scores for familiar talkers. However, the familiar talker advantage in word recognition did not depend on voice quality; the familiar voice qualities of familiar talkers did not produce better word recognition scores than the unfamiliar voice qualities of familiar talkers. These combined results suggest that any integration of linguistic and indexical information in word recognition emerges from a subset of the phonetic features of the speech signal and is not, therefore, a strictly general property of speech perception.

3aSCb16. All representations are not equal: Differential activation of words and phonological variants across accents. Meghan Sumner and Reiko Kataoka (Dept. of Linguist., Stanford Univ., Stanford, CA 94305-2150, sumner@stanford.edu)

Given the massive variation in natural speech, how listeners recognize words is a central issue for linguistic theory. Listener sensitivity to acoustic fluctuations in speech has provided us with one piece of the puzzle: detailed, specific representations. We provide another piece: differential activation of these representations. We conducted a semantic priming with General American listeners of positively viewed non-rhotic British English accents and negatively viewed non-rhotic New York City accents (slender'THIN; slend-UH(BE)'THIN; slend-UH(NYC)'THIN). Controlling participant experience and self-reported familiarity with both accents, only the words produced in a British accent primed semantically related targets. There is no *a priori* reason to expect this pattern, as the phonological variant and lexical items are controlled. We suggest that two variants equally experienced in number (raw frequency) are perceived differentially because positive and negative attitudes influence the activation lexical representations. Additionally, by examining particular participant groups from non-rhotic regions, we show an oscillation between effects of raw frequency and listener attitudes. This work (1) provides a broader understanding of factors influencing activation, (2) increases our understanding of how frequency effects are modulated by other factors (subjective listener attitudes), and (3) illuminates the interactive nature of linguistic and social factors in speech perception.

3aSCb17. Recognition memory for speech of varying intelligibility. Kristin J. Van Engen, Bharat Chandrasekaran, Lauren Ayres, Natalie Czimskey (Dept. of Linguist., Univ. of Texas, 1 Univ. Station B5100, Austin, TX 78712, k-van@mail.utexas.edu), and Rajka Smiljanic (Dept. of Linguist., Univ. of Texas, 1 Univ. Station B5100, Austin, TX 78712)

Previous research has demonstrated that the internal representation of spoken words includes both a phonetic description and information about the source characteristics of specific talkers. Spoken utterances also vary greatly in intelligibility. However, relatively little is known about how such variability affects the encoding of speech signals in memory. In the current study, we hypothesize that increased intelligibility leads to better recognition memory for speech. We test this hypothesis by comparing recognition memory for clear and conversational speech produced by native and non-native speakers of English. In experiment 1, listeners were exposed to semantically anomalous sentences produced by a native speaker of English in clear and conversational speech styles. In experiment 2, listeners heard sentences in both styles produced by a Croatian-accented speaker of English. In each experiment, after initial exposure, recognition memory was tested with an old-new identification task. Preliminary results suggest that native-accented speech and clear speech may facilitate recognition memory compared to foreign-accented and conversational speech. The neural mechanisms involved in processing speech of varying intelligibility are currently being investigated using fMRI. The results of this project will expand our understanding of how different sources of variability in the speech signal affect speech processing and memory representations.

3aSCb18. Perceptual accommodation to rate, talker, and context in Mandarin. Joan A. Sereno, Hyunjung Lee, and Allard Jongman (Dept. of Linguist., Univ. of Kansas, Lawrence, KS 66045, sereno@ku.edu)

The present research examines how perceptual constancy is achieved by exploring three fundamental sources of acoustic variability: changes in the rate at which a sentence is spoken, differences due to which speaker produced the sentence, and variations resulting from the sentence context. These sources of variability are investigated by examining the processing of Mandarin tone (specifically Tone 2 and Tone 3). Acoustic analyses (fundamental frequency and duration) of tone productions of 12 native speakers were conducted. These data were then used to construct stimuli to evaluate perceptual adjustments to speaking rate, talker, and context. 14 target stimuli were created varying in temporal, spectral, and both temporal and spectral characteristics. These target stimuli were preceded by six different precursor sentences representing different speaking rates, speakers, and contexts. Participants were asked for their tone judgments. Analyses show interactions between precursor type and target manipulation. Specifically, normalization primarily occurs when precursor and target vary along the same acoustic dimension. However, variation in coarticulatory context violated this pattern. These experiments will be discussed in terms of the use of temporal and spectral cues and its interaction with listener normalization of speaking rate and talker. [Research supported by NSF.]

3aSCb19. Perceptual shifting versus perceptual learning: How listeners accommodate accented speech. Ed King and Meghan Sumner (Dept. of Linguist., Stanford Univ., Bldg. 460, Rm. 127, Stanford, CA 94305, etking@stanford.edu)

Variation is ubiquitous in speech. Differences among individuals ensure that no two speakers pronounce the same sound the same way. Listeners nonetheless understand variable pronunciations as instances of the same category, adapting to familiar variation (e.g., accents they have heard before) or learning novel variants (e.g., unfamiliar accents). Perceptual learning, in which listeners hear features ambiguous between two phonemes and subsequently shift their perceived phoneme boundary, is claimed as a mechanism for accent learning (Eisner and McQueen, 2005). Work examining adaptation to familiar variation has found that listener expectations about speaker dialect influence speech perception (Hay *et al.*, 2010). This study investigates adaptation to a familiar accent, British English, by testing American listeners' identification of "er" on an [e]-[3] continuum before and after hearing a series of British-accented words without the phoneme /3/. The perceptual learning paradigm predicts that training should shift the 50% "er"-identification crossover toward [3], with listeners accepting more [3]-like sounds as "er" after hearing British English. This approach is novel because none of the training words contain [3], suggesting that, unlike traditional perceptual learning, adaptation to a phoneme in a familiar accent does not require immediate exposure to that phoneme.

3aSCb20. Non-symmetric time resolution for spectral feature trajectories. Stephen A Zahorian, Jiang Wu, and Montri Karnjanadecha (Dept. of Elec. and Comput. Eng., Binghamton Univ., Binghamton, NY 13902)

In a study presented at the fall 2010 meeting of the Acoustical Society of America (Zahorian *et al.*, "Time/frequency resolution of acoustic features for automatic speech recognition"), we demonstrated that spectral/temporal evolution features which emphasize temporal aspects of acoustic features, with relatively low spectral resolution, are effective for phonetic recognition in continuous speech. These features are computed using discrete cosine transform coefficients for spectral information from 8 ms frames and discrete cosine series coefficients (DCSCs) for their temporal evolution, over overlapping intervals (blocks) longer than 200 ms. These features are presented as an alternative to mel-frequency cepstral coefficients, and their delta terms, for automatic speech recognition (ASR). In the present work, it is shown that these features are even more effective for ASR, using a non-symmetric time window which is tilted toward the beginning of each block when computing DCSCs. This non-symmetry can be implemented by combining two Gaussian windows with different standard deviations. This work also supports the hypothesis that the left context is somewhat more informative to phonetic identity than is the right context. Experimental results for automatic phone recognition are given for various conditions using the TIMIT and NTIMIT databases.

3aSCb21. Listening to a novel foreign accent, with long lasting effects. Neil P. Bardhan and Andrea Weber (Adaptive Listening Group, P.O. Box 310, 6500AH Nijmegen, The Netherlands, neil.bardhan@mpi.nl)

In conversation, listeners frequently encounter speakers with foreign accents. Previous research on foreign-accented speech has primarily examined the short-term effects of exposure and the relative ease that listeners have with adapting to an accent. The present study examines the stability of this adaptation, with seven full days between testing sessions. On both days, subjects performed a cross-modal priming task in which they heard several minutes of an unfamiliar accent of their native language: a form of Hebrew-accented Dutch in which long /i:/ was shortened to /I/. During this task on Day 1, recognition of accented forms was not facilitated, compared to that of canonical forms. A week later, when tested on new words, facilitatory priming occurred, comparable to that seen for canonically produced items tested in both sessions. These results suggest that accented forms can be learned from brief exposure and the stable effects of this can be seen a week later.

3aSCb22. Hearing vocalizer race from voiced laughter sounds. Anais F. Stenson, Noel B. Martin, R. Toby Amoss, and Michael J. Owren (Dept. of Psych., Georgia State Univ., P.O. Box 5010, Atlanta, GA 30302-5010, anaisfern@gmail.com)

Previous experiments testing whether an individual's race can be identified from speech sounds have produced inconsistent results. Current work examined this question using spontaneous laughter sounds to help isolate voice quality from dialect effects. In Experiment 1, six laughs from each of six black and six white vocalizers were edited to create 72 bouts of three to four contiguous bursts. These bouts were then further edited to create 72 single bursts. Fourteen undergraduate participants heard both sets of laughs in separate blocks. Sounds were presented twice on each trial and categorized as "Black" or "White." Mean percentage-correct was 61.1 (SD = 10.1) with multi-burst versions, and 55.8 (SD = 12.7) with single bursts. Both

outcomes were statistically higher than chance performance in *t*-test comparisons. Experiment 2 tested 12 new listeners in a balanced, same-different design with paired, bout-length stimuli from same- or different-race vocalizers. There were 120 stimuli in all, combining 5 laughs from each of 5 laughers of each race. Mean d' was 0.92 (SD = 0.44), which was statistically different from chance. Taken together, these experiments suggest modest, but reliable sensitivity to vocalizer race from voice-quality alone.

3aSCb23. Compensation for altered feedback is sensitive to speaking style. Shira Katseff (NZ Inst. of Lang., Brain, and Behaviour, Univ. of Canterbury, Private Bag 4800, Christchurch 8140, New Zealand, shira.katseff@canterbury.ac.nz) and John F. Houde (UCSF, HSE800, San Francisco, CA 94143-0444)

Many speakers oppose alterations to auditory feedback, using a higher pitch when they hear their pitch lowered or a lower first formant when they hear that formant raised. But there is substantial variation in compensation within and across individuals. This case study asks whether a speaker's individual vowel space influences compensation for altered auditory feedback. A generalized linear regression model was used to investigate the influence of vowel density on trial-to-trial formant movement. First, two density maps, showing "hot spot" regions where the subject produced many vowels and "cold spot" regions where the subject produced few vowels, were constructed for (1) casual speech (from a 30-min mock interview) and (2) citation form speech (from 360 HVD words). Maps were produced for one non-phonetician speaker of California English. Afterward, the speaker participated in Five altered feedback experiments on Five separate days. The analysis showed that the subject's between-trial formant changes were not correlated with spontaneous speech vowel density but were positively correlated with citation form vowel density. This pair of results suggests that speech motor decisions are influenced by speech style, accessing citation form vowels more readily when producing citation form speech.

3aSCb24. Perception of speaker sex in children's voices. Peter F. Assmann (School of Behavioral and Brain Sci., Univ. of Texas at Dallas, Richardson, TX 75083-0688) and Terrance M. Nearey (Univ. of AB, Edmonton, AB T6G 2E7, Canada)

To determine how accurately adult listeners can identify speaker sex in children's voices, we presented vowels in /hVd/ syllables produced in isolation and in a brief carrier sentence by five boys and five girls from each of four age groups (5–8, 9–12, 13–16, and 17–18 yr). Speaker sex identification improved with increasing age of the speaker, but even in the youngest group performance was significantly above chance. Strong biases were noted, with older boys more accurately classified than older girls, while the reverse pattern was found at younger ages. Listeners sometimes reported difficulty distinguishing the older girls from younger pre-adolescent boys. Recognition of speaker sex from isolated syllables was more accurate and listeners were more confident of their responses when informed of the age of the speaker. Performance was also higher when syllables were embedded in a carrier phrase, and in this condition the advantage provided by knowledge of the speaker's age was reduced. The results indicate that the perception of speaker sex can be informed by knowledge of the speaker's age, and that sentence context provides additional cues to speaker sex in children's speech.

3aSCb25. Fundamental frequency and perceptual adaptation to gender and emotion. Daniel J. Hubbard and Peter F. Assmann (School of Behavioral and Brain Sci. GR4.1, Univ. of Texas at Dallas, P.O. Box 830688, Richardson, TX 75083, dhubbard@utdallas.edu)

Previous work has documented perceptual adaptation to nonlinguistic properties in speech using voice gender and emotion categories. Exposure to voice gender and expressive speech adaptors produced a response shift away from the adaptation category. This project extends those findings by examining the contribution of fundamental frequency (F0) to perceptual speech aftereffects following adaptation to VCV syllables. Voice recordings from six speakers were processed using the STRAIGHT vocoder and an auditory morphing technique to synthesize gender (experiment 1) and expressive (experiment 2) speech sound continua ranging from one category endpoint to the other (female to male; angry to happy). Continuum endpoints served as adaptors for F0 present and F0 removed conditions. F0 removed stimuli were created by replacing the periodic excitation source with broadband noise. Aftereffects were found only in the F0 present condition, resulting in a decreased likelihood to identify test stimuli as part of the adaptation category. Aftereffects did not appear when F0 was removed. The findings highlight the important role F0 plays in perceptual adaptation to gender and expressive properties of speech, and further identifies a common acoustic basis for speech adaptation among several different classes of stimuli.

3aSCb26. Investigating the roles of vocal tract size and phoneme content in context effects. A. Davi Vitela and Andrew J. Lotto (Dept. of Speech, Lang., and Hearing Sci., Univ. of Arizona, 1131 E. 2nd St., Tucson, AZ 85721, adv1@email.arizona.edu)

In this series of experiments, we examined the role of specific vocal tract and phoneme contextual information on vowel categorization. An articulatory synthesizer was used to create vowels produced by specific vocal tract lengths (VT) of different sizes [Story, J. Acoust. Soc. Am. **117**, 3231-3254 (2005)]. Contexts of repeated vowels (/a/ or /i/) produced by long and short VTs preceded a target series that changed perceptually from /bit/-/bet/. If listeners are extracting information about the VT itself, similar context effects should be elicited by either vowels produced by the same VT. If listeners are comparing the ambiguous target to the placement of a known vowel, one might predict similar results for the /aa/ condition or the /ii/. However, a complete cross-over interaction was obtained such that the effect of the context could not be predicted by the VT size or the vowel identity, but instead on the acoustic characteristics of F1. Follow-up experiments used contexts consisting of cross-cutting vowels from both VTs or two different vowels from the same VT. The results suggest that the order of the appended stimuli matters, demonstrating that listeners are not averaging information equally across the entire context. [Work supported by NIH-NIDCD.]

3aSCb27. Free classification of dysarthric speech. Kaitlin L. Lansford, Rebecca E. Norton, Julie M. Liss, and Rene L. Utianski (Dept. of Speech and Hearing Sci., Arizona State Univ., P.O. Box 870102, Tempe, AZ 85287, kaitlin.lansford@asu.edu)

The "gold standard" for classification of motor speech disorders, based on the work of Darley *et al.* [J. Speech Lang. Hear. Res. **12**, 246–269 (1969)], presumes that the underlying pathophysiology of each dysarthria subtype is responsible for the associated perceptual features. However, this approach to differential diagnosis suffers significant limitations in that (1) there is substantial overlap of perceptual characteristics associated with the dysarthria classes and (2) characteristics of a given dysarthria vary with severity. It is therefore difficult to perceptually classify speakers reliably without knowledge of the underlying etiology. In the present investigation, 20 speech pathology graduate students grouped 33 speakers with dysarthria into clusters according to the speakers' perceptual similarity using an unconstrained free classification task. The resulting similarity data underwent cluster analysis to identify subsets of perceptually similar speakers. Multidimensional scaling (MDS) revealed the dimensions salient to judgments of perceptual similarity. To interpret the dimensions derived by the MDS, correlation analyses comparing the speakers' spatial distances along the dimensions to obtained segmental and suprasegmental acoustic metrics was conducted. The results of these analyses and how they support the establishment of a non-etiology based, perceptually relevant taxonomy of dysarthric speech will be discussed. [This investigation was supported by 1 F31 DC010093-01A2 awarded to K.L.]

3aSCb28. Temporal and spectral characteristics of fricatives in dysarthria. Heejin Kim, Mark Hasegawa-Johnson (Beckman Inst., Univ. of Illinois at Urbana-Champaign, 405 N. Mathews Ave., Urbana, IL 61801, hkim17@illinois.edu), and Adrienne Perlman (Univ. of Illinois at Urbana-Champaign, Champaign, IL 61820)

Imprecise production of fricatives has been noted in various types of dysarthria, but their acoustic properties have not been fully investigated. Using durational measures and spectral moment analysis, this study examines the acoustic characteristics of alveolar and post-alveolar voiceless fricatives produced by speakers with cerebral palsy (CP) in three different intelligibility levels. The following questions are addressed: (1) Are fricatives longer in CP-associated dysarthric speech, compared to controls? (2) What is the evidence of reduced acoustic distinctions between alveolar vs. post-alveolar fricatives? and (3) Are the intelligibility levels associated with acoustic measures of fricatives? Duration and the first three spectral moments were obtained from word initial fricatives produced by 18 American English native speakers (9 speakers diagnosed with spastic CP and 9 controls). Results showed that speakers with CP exhibited significantly longer duration of fricatives and a reduced distinction between alveolar versus post-alveolar fricatives compared to control speakers. A reduced place distinction in dysarthric speech was mostly due to lower first moments and higher third moments compared to normal speech. The group difference was greater for alveolar fricatives than for post-alveolar fricatives. Furthermore, as the intelligibility level decreased, durational increase and the degree of place overlap were consistently greater.

3aSCb29. The role of linguistic and indexical information in improved recognition of dysarthric speech. S. A. Borrie, M. J. McAuliffe (Dept. of Commun. Disord., Univ. of Canterbury, Private Bag 4800, Christchurch, New Zealand, steph.borrie@gmail.com), J. M. Liss (Arizona State Univ., Tempe, AZ 85257-0102), G. A. O.Beirne (Univ. of Canterbury, Christchurch, New Zealand), and T. Anderson (Van der Veer Inst. for Parkinson's and Brain Res., Christchurch, New Zealand)

This investigation examined perceptual learning of dysarthric speech. Forty listeners were randomly assigned to one of two identification training tasks, aimed at highlighting either the linguistic (word identification task) or indexical (speaker identification task) properties of the neurologically degraded signal. Immediately following familiarization, all listeners completed an identical phrase transcription task. Analysis of post-training listener transcripts revealed remarkably similar intelligibility improvements for listeners trained to attend either the linguistic or the indexical properties of the signal. Perceptual learning effects were also evaluated with regard to underlying error patterns indicative of segmental and suprasegmental processing. Comparisons revealed no significant difference at either level of perceptual processing for the two training groups. The findings of this study suggest that elements within both the linguistic and indexical properties of the dysarthric signal are learnable and interact to promote improved processing of this type and severity of speech degradation. Furthermore, error pattern analysis indicates that similar cognitive–perceptual mechanisms may underlie the processing of indexical and linguistic information. Thus, the current study extends support for the development of a model of perceptual processing in which the learning of indexical properties is encoded and retained alongside linguistic properties of the signal.

3aSCb30. Automated rhythmic discrimination of dysarthria types. Rene L. Utianski, Visar Berisha, Julie M. Liss, and Kaitlin Lansford (Dept. of Speech and Hearing Sci. Arizona State Univ., P.O. Box 870102, Tempe, AZ 85287, rutiansk@asu.edu)

This study examines whether features of rhythm, extracted via an automated program, can match the success of hand-extracted metrics in discriminating among control and different types of dysarthric speech. Previously, Liss *et al*. [J. Speech, Language, and Hearing Res., **52**(5), 1334–1352 (2009)] demonstrated that rhythm metrics could successfully separate individuals with perceptually distinct rhythm patterns, with the majority of classification functions more than 80% successful in classifying dysarthria type. However, the hand extracted rhythm measurements are labor intensive and require expertise in acoustic analysis to achieve valid and reliable measures. In this study, digitized speech was automatically segmented into vocalic and voiceless intervals using an autocorrelation-based algorithm, and, from these intervals, rhythm metrics were computed. Discriminant function analyses were performed and revealed the majority of functions were more than 90% successful in classification, matching, or exceeding the success of the hand- extracted measurements. Removing subjectivity, automating this procedure, and verifying its comparability with traditional methods improve the viability of rhythm metrics as a clinical tool in speech therapy intervention. Additionally, the ease of manipulation of the automated program will allow for development of metrics that capture severity and individual speaker differences that will inform models of speech perception.

3aSCb31. The effect of listener age on the recognition of dysarthric speech. Megan J. McAuliffe, Elizabeth M. R. Gibson, Sarah E. Kerr (Dept. of Commun. Disord. and New Zealand Inst. of Lang., Brain & Behaviour, Univ. of Canterbury, Christchurch, New Zealand, megan.mcauliffe@canterbury.ac.nz), and Tim Anderson (Van der Veer Inst. for Parkinson's and Brain Res., Christchurch, New Zealand)

A growing number of studies have attempted to address the cognitive–linguistic source of intelligibility deficits in dysarthria. These have focused predominantly on the ability of young adults with normal hearing to decipher dysarthric speech. However, dysarthria is commonly associated with aging; older listeners often form primary communication partners. It is also known that older adults exhibit difficulty understanding speech that has been temporally or spectrally degraded. It follows that the spectral and temporal degradations present in dysarthric speech may pose a greater perceptual challenge for older, as opposed to younger, listeners. Twenty younger listeners (mean age = 20 yr) and 15 older listeners with good hearing for their age (mean age = 65 yr) transcribed the speech of individuals with moderate hypokinetic dysarthria. Percent words correct (intelligibility) was calculated and underlying error patterns at the suprasegmental and segmental levels of processing were examined. While the younger and older listener groups achieved similar intelligibility scores, the younger group showed greater reliance on syllabic strength cues to inform word boundary decisions. Similar levels of attention to segmental cues were observed across both groups. These results suggest that the recognition of dysarthric speech may be comparable across younger and older listeners.

3aSCb32. Movements of the tongue, lips, and jaw for selected vocalic nuclei in speakers with dysarthria. Gary Weismer (Dept. Commun. Disord., UW-Madison, Goodnight Hall, Madison, WI 53706)

Speech movement data in persons with dysarthria are relatively rare, and much of this sparse data is restricted to the lips and jaw. The purpose of this presentation is to provide a survey of speech movements for selected vocalic nuclei, in speakers with dysarthria due to Parkinson's disease and Amyotrophic Lateral Sclerosis, and to compare these movements to those of healthy controls. The speech materials used in this analysis are those collected over a ten-year period at the x-ray microbeam facility at the University of Wisconsin-Madison. The selected vocalic nuclei are extracted from connected speech samples, including both read sentences and an extended reading passage; multiple repetitions of sentence data are available to estimate the variability of the selected movements. Both qualitative and quantitative analyses of these speech movements, with emphasis on lingual motions, will be employed to generate hypotheses concerning the speech movement deficit in the dysarthrias. This inductive analysis approach will pay special attention to across-speaker variability, among both healthy controls and speakers with dysarthria. A working hypothesis is that the lingual motions for speakers with dysarthria are no more variable across repetitions than those in healthy controls, but show deficits in scale and form.

3aSCb33. A continuing study of the temporal structure of the speech of a person with dementia. Linda Carozza (Dept. of Commun. Sci. and Disord., St. John's Univ., 300 Howard Ave., Staten Island, NY 10301), Margaret Quinn, Julia Nack, and Fredericka Bell-Berti (St. John's Univ., Queens, NY 11439)

The question of motor speech degeneration in the course of dementing illness is a relatively unexplored area; rather, extensive research has focused on cognitive and language processes in dementia. The potential for early dissociation of motor functions of language at the level of speech production has not been explored; an interaction between motor speech and language production and perception changes should inform our understanding of the deterioration in dementia. We have previously reported results of a preliminary study of the temporal structure in the speech of persons with dementia. In a continuing longitudinal study of one of our subjects, we found that after 6 months, utterance-level timing patterns were more disrupted than segment- and syllable-level patterns in his speech. This report explores changes over time in the temporal structure of speech of that participant, who was recorded 12 months after the second recording session. The results of this third recording session, in which we will examine utterance-level timing patterns (phrase-final lengthening and compensatory shortening) and segment- and syllable-level timing patterns (VOT and the effects of final consonant voicing on vowel duration) will be presented.

3aSCb34. Prosodic features and evaluation of pitch controllable electrolarynx. Minako Koike, Satoshi Horiguchi (School of Medicine, Kitasato Univ., 1-15-1 Kitasato, Minami-ku, Sagamihara, Kanagawa, 252-0374, Japan, minako@kitasato-u.ac.jp), Hideki Kasuya (Utsunomiya Univ., Utsunomiya 321-8585, Japan), Yoshinobu Kikuchi (Int. Univ. of Health and Welfare, Otawara 324-8501, Japan), Makihiko Suzuki, and Makito Okamoto (Kitasato Univ., Sagamihara 252-0374, Japan)

The prosodic features and perceptual evaluation of an electrolarynx (EL) with pitch-control function were examined. We previously constructed a prototype electrolarynx of which F0 (hereafter "pitch") can be adjusted by up-down or left-right thumb movement, and it was recently developed into a commercial product (Yourtone II, manufactured by DENCOM). Users can choose the pitch-fixed type with jitter (PF-EL) or the pitch-controllable type without jitter (PC-EL). One laryngectomized male speaker, who was an EL user and whose mother tongue was Japanese, served as the subject. He practiced controlling the pitch of PC-EL in Japanese (Tokyo dialect) for two weeks, and then his utterances when using PF-EL and PC-EL were recorded. Twenty normal listeners evaluated his EL speech tokens and were asked to rate on a visual analog scale how close the speech tokens were to normal speech. The results indicated that although the pitch range of PC-EL was narrower than that of normal speech and the production of Tokyo-dialect Japanese was not perfect, most of the PC-EL speech was rated more highly than PF-EL speech. [Work supported by KAKENHI (20500163 and 22500147).]

3aSCb35. A principal components analysis of tongue motion patterns in normal and glossectomy speakers. Julie Langguth (Dept. of Orthodontics, Univ. of Maryland Dental School, 650 W. Baltimore St., Baltimore, MD 21201, juliemikhail@gmail.com), Jonghye Woo, Maureen Stone (Univ. of Maryland Dental School, Baltimore, MD 21201), Hegang Chen (Univ. of MD School of Medicine, Baltimore, MD 21201), and Jerry L. Prince (Johns Hopkins Univ., Baltimore, MD 21218)

This study determines whether a small excision of the tongue, due to removal of a lateral T1 tumor, will cause a change in tongue motion patterns during speech. Principal components analysis (PCA) compared the motion patterns of ten control subjects to those of three glossectomy patients. The patients were 7–39 months post-glossectomy removal of T1N0M0 tumors and had primary closures. Cine- and tagged- magnetic resonance imaging (MRI) data included three sagittal slices, which were used to compare the midsagittal and bi-lateral motions of the tongue during speech. The data were internal tissue point motions extracted from the tagged MRI images using harmonic phase MRI (HARP-MRI) (NessAiver and Prince, 2003). The motion patterns of the tongue (velocity field) were observed during the motion from "ee" to "s" which may be difficult for glossectomy patients as it requires subtle changes in the tongue tip. The results demonstrated normal variability across the control subjects. The three patients retained some components of normal tongue motion; however, there were also notable differences from the controls. The glossectomy patients plotted at the extremes of the PC loadings, and the details revealed how they differed from the controls, from each other, and used individual compensatory strategies. [Work supported by NIH-R01CA133015.]

3aSCb36. Acoustic characteristics of diphthongs in Parkinson's disease. Kris Tjaden (Dept. of Commun. Disord. & Sci., Univ. at Buffalo, 3435 Main St., Buffalo, NY 14214, tjaden@buffalo.edu)

The second formant frequency (F2) is sensitive to dysarthria. For example, vowel F2 slope has been shown to correlate with intelligibility, such that persons who are less intelligible or who have more severe speech impairment also tend to have shallower F2 slopes (Weismer *et al*., 1992). This result is consistent with studies suggesting the perceptual importance of transitions (Hillenbrand and Nearey, 1999), although the extent to which F2 slope explains reduced intelligibility in dysarthria is unclear. F2 slope also may serve as an acoustic metric of effort (Moon and Lindblom, 1994; Wouters and Macon, 2002). Therapeutic techniques for dysarthria, including Clear and Loud speech, are thought to be beneficial because they involve a scaling up of effort. Clear speech emphasizes hyperarticulation or increased articulatory effort, while Loud speech focuses on increasing respiratory–phonatory effort, although it has been suggested that increased loudness involves a system-wide scaling up of effort (Fox *et al*., 2006). Empirical evidence that Clear or Loud speech is associated with increased articulatory effort in dysarthria is limited. The current study sought to further evaluate this suggestion by investigating F2 slope characteristics for vowels produced in Habitual, Clear and Loud conditions by individuals with dysarthria secondary to Parkinson's disease.

3aSCb37. Music masking speech in cochlear implant simulations. Shaikat Hossain and Peter F. Assmann (Behavioral and Brain Sci., Univ. of Texas at Dallas, Richardson, TX 75083)

While the question of how speech is segregated among competing talkers has received quite a lot of attention in the cochlear implant (CI) literature, an equally important question has been largely neglected: To what extent does music mask speech? Recent studies have shown that bimodal stimulation, using a hearing aid to boost low frequency acoustic information in the ear with residual hearing, can markedly improve CI users' ability to separate multiple talkers, recognize speech in noisy conditions and discriminate between melodies. In this study, normal hearing listeners completed a speech recognition task in musical maskers under two different processing conditions: bimodal and normal CI simulations. Results showed that listeners performed significantly better under the bimodal simulation in all three SNR conditions (0, -3, -6 dB). It is thought that the low frequency acoustic information provided listeners with a release from masking as compared to the normal CI condition, particularly at low SNRs. The amount of masking release varied as a function of instrument type and SNR. The findings of this study may be especially relevant for the perceptual separation of instrumentation and lyrics in popular music which remains to be a highly challenging task for CI users.

3aSCb38. Envelope and temporal fine structure perceptual weights for sentences: Effect of age, hearing loss, and amplification. Daniel Fogerty and Larry E. Humes (Dept. of Speech and Hearing Sci., Indiana Univ., 200 S. Jordan Ave., Bloomington, IN 47405, dfogerty@indiana.edu)

The speech signal may be divided into spectral frequency bands, each band containing temporal properties of the envelope and fine structure. This study measured the perceptual weighting strategies for the envelope and fine structure in each of three frequency bands for sentence materials in young normal-hearing listeners, older normal-hearing listeners, aided older hearing-impaired listeners, and noise-matched young normal-hearing listeners. A novel processing method was designed to vary the availability of each acoustic property independently through noisy signal extraction. Thus, the full speech stimulus was presented with noise used to mask the different auditory channels. Perceptual weights were determined by correlating a listener's performance with the signal-to-noise ratio of each acoustic property on a trial-by-trial basis. Preliminary results demonstrate that fine structure perceptual weights remain stable across the four listener groups. However, a different weighting typography was observed between the listener groups for envelope cues. Preliminary results suggest that spectral shaping used to preserve the audibility of the speech stimulus may alter the allocation of perceptual resources. The relative perceptual weighting of envelope cues may also change with age. These effects were largely limited to envelope weights. [Work supported by NIA R01 AG008293.]

3aSCb39. Contributions of static and dynamic spectral cues to vowel identification by cochlear implant users. Gail S. Donaldson, Catherine L. Rogers, Emily S. Cardenas, Benjamin A. Russell, and Nada H. Hanna (Dept. of Commun. Sci. and Disord., Univ. of South Florida, 4202 E. Fowler Ave., Tampa, FL 33620, gdonalds@usf.edu)

Relatively little is known about cochlear implant (CI) users' ability to make use of static versus dynamic spectral cues in vowel perception tasks. The present study measured vowel identification in CI users and young normal hearing (YNH) listeners using naturally produced /dVd/ stimuli (*deed, did, Dade, dead, dad, dud,* and *Dodd*). Vowel identifcation was tested for (1) the unmodified syllables, (2) syllables modified to retain only 60 or 80 ms of the vowel center (*center-only* conditions), and (3) syllables modified to retain only 30 or 40 ms of the initial and final vowel transitions, with vowel duration neutralized (*edges-only* conditions). YNH listeners achieved high levels of performance for the unmodified stimuli (avg. 99.8%) and for the center-only stimuli (90.8%); their performance dropped to more moderate levels (68.1%) for the edges-only stimuli. CI users demonstrated moderate performance for the unmodified stimuli (avg. 72.0%) but demonstrated substantially poorer performance for both the center-only (41.1%) and edges-only stimuli (27.8%). Findings suggest that CI users (1) have difficulty identifying vowels in syllables when one or more cues are absent and (2) rely more strongly on quasi-static cues from the vowel center as compared to dynamic cues at the syllable edges.

3aSCb40. Spectral contrast analyses of vowels processed through a multichannel simulated hearing aid. Amyn M. Amlani (Dept. of Speech and Hear Sci, Univ. of North Texas, P.O. Box 305010, Denton, TX 76201, amlaniam@unt.edu), Sneha V. Bharadwaj (Texas Woman's Univ., Denton, TX 76204), Shirin Jivani, and Jody Pogue (Univ. of North Texas, Denton, TX 76201)

Hearing aids are designed to separate the audible frequency response into independent channels so that gain can be modulated correspondingly, based on the degree and configuration of the hearing loss. However, increasing the number of channels and employing fast-acting compression might lead to the spectral flattening of speech segments. In this study, we performed spectral contrast analyses on three vowels /i, oo, u/ spoken by two male and two female talkers in a /CVC/ context. The initial consonant consisted of /p, b, t, d, k, g, s, sh/ and the final consonant consisted of /t/. Each /CVC/ word was spoken in the phrase, “I said /CVC/, again” Each phrase was then processed by a simulated hearing aid having 2, 4, 8, and 16 independent channels according to an amplification scheme (linear, compression [fast-fast, slow-slow, fast-slow]). Findings revealed that spectral contrast decreased (1) as the number of channels increased, (2) when a fast attack time was employed, and (3) when the talker was a male. Further, significant spectral flattening and shifts in vowel formants occurred for /i, oo/ compared to /u/. We discuss these findings relative to the speech-intelligibility performance of impaired listeners.

3aSCb41. An additive blending sound presentation system with air and bone conducted sound. Kenzo Itoh (Dept. of Comput. Sci. and Information System, Iwate Prefectural Univ., Iwate 020-0193, Japan, itoh@iwate-pu.ac.jp)

The sound of our voice reaches the inner ear by two paths: the first being air and the second being sound conducted by bones. When you speak, what you hear as your voice is the blending of two signals: one signal comes from the auditory canal and the other signal comes from bone conduction. This paper proposes a new sound presentation system with an additive blending effect with air and bone conducted signals. The system is composed of normal headphones and/plus a bone conduction transducer. The sound quality is evaluated by subjective tests using three kinds of musical sources: Piano, Vocal, and Orchestra. Two experimental conditions are used. One condition is normal headphones as air only (AIR), and the other is normal headphones and a bone conduction transducer simultaneously and in phase (BONE). The subjects select the better quality after comparing AIR and BONE conditions. The result of the experiment shows that the most subjects chose BONE. The proposed system can be used for aged persons or persons with hearing loss.

3aSCb42. Perceptual deficits in Thai sensorineural hearing loss patients. C. Tantibundhit, C. Onsuwan, N. Rueangwit, N. Saimai, T. Saimai (Dept. of Elec. and Comput. Eng., Thammasat Univ., Thailand, tchartur@engr.tu.ac.th), N. Thatphithakkul, K. Kosawat, S. Thatphithakkul, and P. Chootrakool (Natl. Electron. and Comput. Technol. Ctr., Thailand)

This study explored differences in CVV perception in two groups of Thai listeners: with normal hearing and with sensorineural hearing loss (with/without hearing aids). All participants chose one response in each of 210 Thai stimulus rhyming pairs, e.g., /taa/-/naa/. The rhyming monosyllabic words share an /aa/ vowel and mid tone, but differ in their initial phonemes (symmetrically distributed across 21 phonemes). While all stimuli for the normal hearing group were embedded in 4 signal-to-noise ratio levels, clean stimuli were presented to the patients. Comparisons of confusion patterns and perceptual distance were made. In both groups, /r/ is the most confusable phoneme, while /w/ is among the least. Perceptual representations of initial phonemes show five individual clusters: glide, glottal constriction, nasality, aspirated obstruent, and a combination of liquid and unaspirated obstruent. Patients’ perceptual difficulty could be attributed to the nasality grouping, which is normally well separated, shifting closer to the glottal constrictions and aspirated obstruents. Hearing aids seem to improve perception of all phonemes by 10%, with /kh/ and /h/ showing the highest improvement rate, and /d/ the lowest. The instruments are beneficial in moving the nasality cluster further away from the nearby groupings.

3aSCb43. Living sound identification system using smartphone for persons with hearing loss. Ashita Sarudate and Kenzo Itoh (Dept. of Comp. Sci. and Info. Sys., Iwate Prefectural Univ., Iwate 020-0193, Japan, g236g003@s.iwate-pu.ac.jp)

In today’s advanced information society that overflows with a variety of sounds, people with hearing loss find it very difficult to obtain information on sounds within the home. Although there are many systems to aid handicapped persons, support systems for the hearing impaired do not provide adequate performance. In light of this, the various living sounds identification system for persons with hearing loss was proposed. This system adopts the method of pre-storing signal characteristics so that it can discriminate important living sounds in the home with a high degree of accuracy. In order to construct a real-time processing system, basic signal processing was subjected to frame-by-frame analysis. In addition, so as to be able to detect signals precisely in a noisy environment, signal part detection—based on auditory perception—was adopted. In an experiment in a simulated life space, the system was able to discriminate ten living sounds with almost 100% accuracy. Therefore, the proposed method was very applicable. In addition, the method used by us to present the sound identification results using a mobile phone is considered to be of value. This paper proposed the living sound identification system for use with a smartphone and discussed the utility of this system.

3aSCb44. Talker and gender effects in induced, simulated, and perceived stress in speech. James Harnsberger, Christian Betzen, Kristen Perry, and Harry Hollien (Dept. Linguist., Univ. of Florida, Gainesville, FL 32611, jharns@ufl.edu)

Prior modeling on the speech correlates of psychological stress have typically reported averaged data in which the problem of quantifying degree of stress was not adequately addressed. Harnsberger, Kahan, and Hollien (2006) reported a speech database in which stress was quantified both with (two) physiological measures and (two) self-report scales. In this study, these materials were assessed in two stress perception experiments, involving ratings and classification, and the subsequent analysis was organized by gender and individual talker. Several observations were made. First, both induced and simulated stress samples were perceptibly different from baseline, unstressed samples. Second, up to one third of talkers in a given gender group differed from the overall trend, showing instead a propensity to be rated as less stressed during high degrees of physiological arousal (reflected also in self-report scales); essentially, sounding calmer under stress than the baseline condition. Third, female talkers showed the expected positive correlation between the degree of physiological shift under stress versus the shift in self-reported stress ratings while, in male subjects, higher degrees of physiological arousal were underreported in self-report (a “tough guy” response). Finally, acoustic modeling efforts were conducted by gender and stress response type, for induced, simulated, and perceived stress.

3aSCb45. Reliability and validity of a simple user interface (EarPrint™) for the personalization of mobile audio. Meena Ramani, Nick Michael, and Chaslav Pavlovic (Sound ID and SoundHawk 2595 East Bayshore Rd., Palo Alto, CA 94303)

People hear differently and have different aesthetic criteria. Due to a number of these reasons, individuals may prefer different processing parameters for both speech and music. For example, some individuals may prefer more low frequency emphasis and longer reverberation time while listening to music. Alternatively, certain individuals with hearing loss may prefer higher compression ratios at high frequencies. To perform this “personalization,” a number of psychometric procedures can be used. They vary in terms of accuracy, complexity, testing time, etc. For consumer devices, such as headsets, personal amplifiers, personal music players, etc., we have developed a new methodology, called EarPrint™, which is fast, intuitive, and requires minimum instructions. The procedure is based on reducing a multidimensional parametric space into a 2-D surface representation where the user perceives a consistent quality increment as he/she moves his/her finger along a straight line. In this paper, we evaluate how valid and reliable the EarPrint™ results are when compared to more rigorous psychometric procedures (e.g., category scaling or magnitude estimates.) Fifteen subjects were tested and the results were analyzed in terms of validity, dependence on listening material and reliability.

Session 3aSPa

Signal Processing in Acoustics: Array Processing and Source Localization Techniques

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Contributed Papers

8:00

3aSPa1. Broadband model-based tracking. Edmund Sullivan (Prometheus Inc., 46 Lawton Brook Ln., Portsmouth, RI 02871)

Joint range and bearing estimation using a Kalman-based recursive processor has already been achieved for the narrow band case. The signal is modeled as having a circular wavefront curvature, and the range is estimated as the radius of this curvature. The bearing is also jointly estimated along with the range. Since there is a direct estimation of the range, a maneuver of the towed array receiver is not necessary. To extend this procedure to the broadband case, a preprocessed measurement is computed using a sequence of contiguous Fourier transforms. The phases of the associated frequency components are then aligned, providing one effective frequency. This allows an effective time delay to be computed for each Fourier transform, thus allowing a sequential estimation of the bearing and range to be carried out. A further advantage of this approach is that, since the array is in motion, the Doppler of the effective frequency can be included in the signal model, thereby allowing the additional bearing information in the Doppler to be exploited. This extra bearing information significantly decreases the variances on the estimated parameters, providing a synthetic aperture effect. This permits the use of a relatively short array aperture.

8:15

3aSPa2. Tracking system developed using an optical fiber-based distributed acoustic sensor. Georgios Efstathopoulos, Daniel Finfer, Yousif Kamil, Sergey Shatalin, Tom Parker, and Mahmoud Farhadiroushan (Silixa Ltd., Silixa House, 230 Centennial Park, Elstree, Hertfordshire WD6 3SN, United Kingdom)

This work will present a system for acoustic source tracking and imaging developed using a novel distributed acoustic sensor. This sensor, the intelligent Distributed Acoustic Sensor (iDAS), uses standard off-the-shelf optical fiber as a transducer to capture the audio-frequency acoustic field (both phase and amplitude) simultaneously along the entire length of a fiber with a spatial resolution of about 0.5 m. iDAS was used to track the bearing of acoustic sources in a reverberant environment, with localization being performed in two stages. First, the received signal was preprocessed for the mitigating effects of calibration errors and multipaths. Then tracking was performed via particle filters. These filters can track multiple acoustical sources and are efficient enough to run in nearly real time. The output from this system is presented in the form of an acoustic camera-type video.

8:30

3aSPa3. Tracking of small vessels with passive acoustic sensors. Helen H. Ou, Pasang Sherpa, and Lisa M. Zurk (Dept. of Elec. and Comput. Eng., Portland State Univ., 1900 SW 4th Ave., Portland, OR 97201, ohui@pdx.edu)

This paper introduces a small boat vessel tracking method in marine environment using autonomous, low-complexity acoustic sensors. A cross-correlation of time series between sensors generates a coarse localization, and an application of the extended Kalman filter (EKF) gives the vessel track. Since each sensor has a local clock that operates asynchronously, the time series received by multiple sensors are first synchronized with one another by measuring a sequence of impulses played at the beginning of the recording. The algorithm detects a moving boat by the increase in the broadband sound level and confirms it by extracting the amplitudes at harmonic frequencies due to the propeller movement. To calculate the time delay of arrival, the boat-present signals are divided into small segments that are cross-correlated with signals received by other sensors. Thresholding and clustering are introduced to extract multiple tracks from the cross-correlation. The EKF is trained using the estimated time delays to provide the vessel track. The algorithm has been successful with Bellhop simulated boat signals and field data collected on the Willamette River and Columbia River in Oregon. Limiting factors such as network topology and sensor movement during deployment are also investigated. [Work sponsored by the Nature Conservancy.]

8:45

3aSPa4. Double multiple signal characterization active large band (D-MUSICAL) for extraction of observable in acoustic tomography. Le Touzé Grégoire, Nicolas Barbara (Gipsa-Lab, 961 rue de la Houille Blanche, BP 46 F - 38402 Grenoble Cedex, gregoire.le-touze@gipsa-lab.grenoble-inp.fr), Roux Philippe (ISTerre, BP 53 38041 Grenoble Cedex 9, France), and Mars Jérôme (Gipsa-Lab., 961 rue de la Houille Blanche, BP 46 F - 38402 Grenoble, Cedex)

In the ocean, the acoustic propagation is characterized by different arrivals. Each arrival is characterized by its observables: amplitude, arrival time, emission, and reception angles. To achieve ocean acoustic tomography, a method to separate arrivals and extract observables has been developed. Two vertical arrays are used: a source and a receive array. Until now, the double-beamforming algorithm has been used to extract observables. It allows to go from the initial (time–source depth–receiver depth) domain to the beamformed domain (time–source angle–receive angle). This algorithm is efficient thanks to the 3-D domain of estimation but shows resolution limitation in the angle domain. To improve wave separation, we develop and apply a high-resolution algorithm based on MUSIC principle, in the same 3-D domain. The proposed method, called Double-MUSICAL, is the extension on the emission angle of the 2-D MUSIC Acitve Large band method (MUSICAL). Its principle is to separate the recording space onto the two orthogonal signal and noise spaces. We present results on real data recorded in an ultrasonic tank and show that the developed method better detects and separates the first arrivals than double beamforming.

9:00

3aSPa5. A direction-selective filter with a rational beam pattern. Dean J. Schmidlin (El Roi Analytical Services, 2629 US 70 East, Unit E-2, Valdese, NC 28690-9005, djschmidlin@charter.net)

Presented in this paper is a direction-selective filter whose beam pattern is a rational function of a direction cosine. First, a plane-wave sinusoidal pressure function is converted from a 4-D function to a 2-D one by restricting the spatial points to lie on a radial line extending out from the origin in a prescribed look direction. The 2-D pressure function is then input into a linear filter characterized by a second-order partial differential equation with constant coefficients. The natural and forced responses are determined from which expressions for the beam pattern of the filter and the time constant of the natural response are found. The beam pattern is the reciprocal of a second-degree polynomial function of the direction cosine of the plane wave. It is shown that the frequency range over which the integrity of the beam pattern is maintained is a function of the natural response of the filter. An example is presented which illustrates the directivity index that is achievable in contrast to that of a vector or dyadic sensor, both of which have beam patterns that are polynomial functions of a direction cosine.

9:15

3aSPa6. Multiple sources localization with microphone array: A subarray approach. Kai-Chung Tam (Dept. of Bldg. Services Eng., The Hong Kong Polytechnic Univ., Hung Hom, Kowloon, Hong Kong), Siu-Kit Lau (Construction Univ. of Nebraska-Lincoln), and Shiu-Keung Tang (The Hong Kong Polytechnic Univ., Hung Hom, Kowloon, Hong Kong)

The problems of location and distance estimations with planar microphone array for multiple point sources are addressed in the present study. Though traditional array processing methods show the reliability in estimation of direction-of arrival (DOA) of the sources, such techniques are incapable to provide source distances which requires for three dimensional (3-D) target coordinates. With planar array processing, azimuth and elevation angles are always obtainable but no longer to extend into more details in 3-D analysis. A new approach to deal with this problem is proposed by estimating multiple sources' DOA with multiple planar sub-arrays. A least square solution based on the estimated DOA of the sources is derived to give the source distances with minimum two sub-arrays in arbitrary microphone configurations. The benefits of the proposed method include a extension of the capability of planar array processing to a higher dimensional analysis. Moreover, its adaptability with any DOA estimation methods makes it convenient to use. The new approach is examined with conventional delay and sum beamforming and MUSIC methods under noisy conditions by both simulations and experiments. [K.C.T. is supported by a studentship from the Hong Kong Polytechnic University.]

9:30

3aSPa7. A comparative analysis of numerical optimization algorithm for array pattern synthesis. Yingmin Wang, Xiao Wei, and Yucai Wang (College of Marine Eng., Northwestern Polytechnical Univ., 127 You Yi Xi Lu, Xi'an 710072, People's Republic of China)

The problem of array pattern synthesis with desired shapes has received much attention over years. There is a number of intelligent optimization methods that were adopted for pattern synthesis, and to some content, several successful design cases are reported. In this paper, based on a special sensor array, a comparative study of those methods is given. Those methods include simulated annealing, genetic technique, and adaptive searching procedures. The effectiveness of the algorithms is demonstrated by the sidelobe level and the width of the main beam. For a given array, those techniques are investigated on their convergence speed and residue error energy. A set of design results are given in details and conclusions are included.

9:45

3aSPa8. Analysis of truncation and sampling errors in the spherical harmonic representation of soundfields. Stefanie Brown and Deep Sen (Acoust. Lab., School of Elec. Eng. and Telecommun., Univ. of New South Wales, Sydney, Australia, stefanie@unsw.edu.au)

The use of microphone arrays to record soundfields for subsequent immersive reproduction, real time or off-line beamforming applications are becoming increasingly prevalent. The representation and analysis of such recorded soundfields using finite-order spherical harmonic coefficients is elegant and has a number of advantages. The focus of this paper is to quantify the accuracy of such a representation which is limited by errors introduced by several types of non-idealities: manufacturing non-idealities of the array such as microphone positional errors, coupling between microphones, etc.; physical non-idealities due to the discrete spatial sampling of the soundfield which introduces spatial aliasing and the inherent truncation of the modal representation of the soundfield to a finite order; and mathematical errors due to the computational technique used to derive the modal coefficients. This paper investigates the latter two classes of errors and quantifies the significant advantages in accuracy achievable when the physical structure of the microphone array allows the use of the orthogonal properties of the spherical harmonic basis functions to extract the modal coefficients. In addition, we extend previously published analysis of the truncation error, which was limited to incident plane waves, to spherical waves.

10:00

3aSPa9. Analysis on novel tangible acoustic interfaces approaches and new thoughts. Xu Wang (School of Honors, HIT, Harbin, 150001 China, yiwang20071@hotmail.com) and Hong Jun Yang (Univ. of INHA, Incheon)

To solve the problems in novel tangible acoustic interfaces approaches [time delay of arrival (TDOA) and location pattern match (LPM)], a new method (which is amplify contrast method) of setting threshold so as to determine the arrival point will be applied, and three new thoughts on LPM, which are improved LPM, time match, and amplitude attenuation match, would be proposed. Traditional TDOA is based on measuring time difference and wave velocity. Traditional LPM is based on time reversal theory and cross correlation. New thoughts are based on amplification on signal, classification of signal source, match time, and uniqueness of damping. Amplify contrast method improves the accuracy of determining the arrival point.

Session 3aSPb

Signal Processing in Acoustics, Underwater Acoustics, and Biomedical Acoustics: Fusion of Acoustic Signals with Data from Various Sensor Modalities

Grace A. Clark, Cochair
Electronics Engineering, Lawrence Livermore National Lab., 7000 East Ave., Livermore, CA 94550

Colin W. Jemmott, Cochair
Applied Operations Research, 420 Stevens Ave., Ste. 230, Solana Beach, CA 92075

Chair's Introduction—10:30

Invited Papers

10:35

3aSPb1. Multiview, multimodal fusion of acoustic and optical data for classifying marine animals. Paul L. D. Roberts, Jules S. Jaffe (Marine Physical Lab., Scripps Inst. of Oceanogr., La Jolla CA 92093-0238), and Mohan M. Trivedi (Univ. of California at San Diego, La Jolla, CA 92093-0434)

Multiview, multimodal acoustic and optical sensor data can be used to improve our ability to remotely classify marine fish and zooplankton. A key advantage of these data is that by capturing more views of a subject, they are more robust to unknown orientation or pose. In this regard, these data offer significant advantages over singleview, unimodal sources. However, they require additional processing to combine data sources together to render a final classification. Here, a fusion algorithm based on combining data sources at the classification probability level (decision fusion) using confidence-weighting with feedback is investigated. This algorithm uses support vector machines as the underlying classifiers can incorporate any number of views or modalities on the fly and is in general independent of the underlying data source. In addition, the algorithm is able to favor views or modalities that are more discriminant without any *a priori* knowledge. The algorithm is tested on an array of problems related to classifying fish from broadband acoustic scattering and classifying zooplankton from broadband acoustic scattering and optical images. In all cases, the algorithm yields significant (more than 50%) reductions in error by fusing multiview, multimodal data.

10:55

3aSPb2. Exploiting multiple sensor modalities for classification of underwater munitions: A Dempster–Shafer theoretic approach. Thanuka L. Wickramarathne, Kamal Premaratne, Shahriar Negahdaripour (Dept. of Elec. and Comput. Eng., Univ. of Miami, Coral Gables, FL 33146), Lisa N. Brisson, and Pierre P. Beaujean (Florida Atlantic Univ., Dania Beach, FL 33004)

Detection and classification of underwater UXOs (UneXploded Ordnance) is a task that is receiving considerable attention from the DoD and related agencies that are involved in management of military munitions. Achieving a reasonable accuracy in detection and classification of these objects is extremely difficult mainly due to the harsh cluttered underwater environment. It is now an accepted fact that no single sensing technology can be both accurate and cost-effective. Low visibility in underwater exposes a significant limitation to optical cameras which are usually better suited for identifying the physical shape of objects. While acoustic imaging is a good alternative, the characteristics of imaging and physical system artifacts make the object recognition task non-trivial. Multi-sensory fusion provides an avenue to exploit the strengths of individual sensors and modalities while mitigating their weaknesses. We address the problem of fusion of multiple sensory information for the task of underwater UXO detection via the use of a Dempster–Shafer (DS) theoretic evidence fusion scheme. The DS theoretic foundation of our fusion algorithm also enables it to incorporate domain expert opinions and knowledge from databases to assist in the detection and classification tasks. We illustrate this method via the use of real data obtained at a test site located in the Florida Atlantic University premises.

Contributed Papers

11:15

3aSPb3. Applications of rhythm algorithm for periodic broadband signals. Alexander Ekimov (NCPA, Univ. of Mississippi, 1 Coliseum Dr., Oxford, MS 38677)

Rhythm algorithm, recently published in J. Acoust. Soc. Am. **129**(3), 2011 was developed for any periodic broadband signals. This algorithm was applied in human and animal footsteps studies for signals from a node of orthogonal sensors and for underwater signals of sperm mammals clicks measured with hydrophones. Some new applications of this rhythm algorithm are presented and discussed. The rhythm algorithm was applied for water breaking waves on a lake shore. The signals were from a node of two orthogonal sensors placed on the lake shore. This node had a narrow band ultrasonic microphone and a Doppler sonar unit. The narrow band ultrasonic microphone showed detectable levels of ultrasonic signals for breaking waves while Doppler sonar showed maximum wave velocities. Signals from this node were recorded and processed simultaneously with the rhythm algorithm. Another application of this rhythm algorithm for music periodic

sound is presented also. In this case, it was possible to see the music rhythms and variations of these rhythms with time using short term Fourier transform.

11:30

3aSPb4. Acoustically sensing the presence of methane and carbon dioxide in underground coal mine. A. Singh and M. Radhakrishna (Indian Inst. of Info. Tech., CSED, Allahabad, 211012 India, ajitsingh@iiita.ac.in)

A WSN can be used to continuously sense, monitor, and transmit data to a centralized control station in a under ground coal mine. A fact limiting the possibility is the presence of highly humid condition in UG coal mines. Current sensors cannot work continuously over prolonged period in UG coal mine environment. This paper describes a multi-aspect data fusion approach for acoustical sensors, which make it possible to measure the build up of methane and carbon dioxide in UG coal mine environment. Suggested approach takes time of flight, phase, and attenuation of sonic pulses to determine the build up of methane and carbon dioxide. Suggested approach is more power efficient in comparison to existing sensors. A temperature sensor is used to accommodate change in characteristics of sonic pulses.

11:45

3aSPb5. Development of three-dimensional time reversal algorithm to locate tinnitus-related hyperactive neurons. Na Zhu, Sean Wu, and Jinsheng Zhang (Dept. of Otolaryngol., School of Medicine, Wayne State Univ., Detroit, MI 48201)

This paper aims at developing 3-D time reversal (TR) algorithm to locate the positions of tinnitus-related neural activities inside the brain auditory pathways, including the auditory cortex (AC), inferior colliculus (IC), and dorsal cochlear nucleus (DCN), which have been implicated in the etiology of tinnitus. Spontaneous neural network signals will be measured by using multichannel electrode arrays that are implanted on the AC, IC, and DCN of adult Sprague Dawley rats and be taken as input to this TR algorithm. To specify the speeds at which neural signals are traveling through the brain auditory pathways, benchmark tests will be conducted, in which electrical stimulations at pre-determined positions in the AC, IC, and DCN will be used, and the corresponding responses be measured. Once this is done, the precise locations of hyperactivity that is responsible for tinnitus perception can be determined by using TR algorithm to scan through the brain auditory pathways. Note that this 3-D time reversal algorithm will allow for tracing neurons into deeper layers of the AC, IC, and DCN, provided that the electrode arrays are implanted in strategically placed in space. Numerical simulations and preliminary experimental results will be presented.

WEDNESDAY MORNING, 2 NOVEMBER 2011 PACIFIC SALON 3, 7:40 A.M. TO 12:00 NOON

Session 3aUW

Underwater Acoustics, Acoustical Oceanography, and Signal Processing in Acoustics: Inversion Methods and Uncertainty Estimation in Ocean Environments

Stan E. Dosso, Cochair
School of Earth and Ocean Science, Univ. of Victoria, P.O. Box 1700, Victoria, B. C., V8W 3P6, Canada

David P. Knobles, Cochair
Applied Research Lab., Univ. of Texas, 10000 Burnet Rd., Austin, TX 78713

Chair's Introduction—7:40

Invited Papers

7:45

3aUW1. Sequential geoacoustic inversion using frequency coherent processing. Caglar Yardim, Peter Gerstoft, and William S. Hodgkiss (Scripps Inst. of Oceanogr., Univ. of California San Diego, La Jolla, CA 92093-0238, gerstoft@ucsd.edu)

Insufficient sampling in frequency domain results in range aliasing that affect geoacoustic inversion. Range aliasing and its effects on source localization and environmental parameter inversion are demonstrated on data collected during the MAPEX2000 experiment. Using a source and geoacoustic particle filter mitigates range aliasing caused by using just a few frequencies. Particle filters are sequential Bayesian techniques that provide dynamic estimation of both the geoacoustic parameters and their evolving uncertainties. This approach allows for side lobe suppression due to prior information supplied by the particle filter and hence improved source localization that results in better geoacoustic inversion.

8:05

3aUW2. Geoacoustic inversion with sequential Bayesian filtering and multipath arrivals. Zoi-Heleni Michalopoulou (Dept. of Math. Sci., New Jersey Inst. of Technol., Newark, NJ 07102, michalop@njit.edu)

When sound propagates in the ocean, multipath arrival receptions at spatially separated hydrophones provide a wealth of information on geometry parameters, including source location and water column depth, and sound speed in medium layers. Additionally, attenuation can be extracted using arrival amplitudes. We extend a method previously developed for arrival time estimation in space using particle filters, which now employs backward filters that smooth arrival time and amplitude estimates. The new approach is applied to synthetic and real data for source localization and geoacoustic inversion. Results are compared with estimates obtained using traditional arrival time estimation methods. The comparison reveals that the new technique provides more accurate results with reduced uncertainty in comparison to other approaches; uncertainty is expressed via posterior probability density functions. It is shown that the smoother plays a significant role in the quality of the time estimates and, subsequently, in geoacoustic inversion. [Work supported by ONR.]

8:25

3aUW3. Examining meso-scale variability: Sequential geoacoustic posterior probability estimation for autonomous underwater vehicle (AUV) towed array data. Jan Dettmer (School of Earth and Ocean Sci., Univ. of Victoria, Victoria BC, Canada V8W 3P6), Charles W. Holland (The Penn State Univ. State College, PA), and Stan E. Dosso (Univ. of Victoria, Victoria BC, Canada V8W 3P6)

This paper applies a sequential trans-dimensional Monte Carlo method to data recorded on an autonomous underwater vehicle towed array. The seismo-acoustic recordings are processed in terms of reflection coefficients as a function of frequency and angle, resulting in a sequence of data sets with small seabed footprints. The sequential particle filter applies advanced Markov chain Monte Carlo steps in combination with importance sampling techniques to carry out consecutive geoacoustic inversions along the track. In particular, the typically high information content of reflection-coefficient data is addressed by heuristic interpolating distributions implemented using annealed importance sampling to bridge between adjacent data sets with distant/disjoint high-likelihood regions. The geoacoustic parametrization, including changes in the number of sediment layers, is estimated along the track with trans-dimensional partition modeling. This trans-dimensional approach intrinsically estimates the amount of structure consistent with the data-information content and accounts for the limited knowledge about the parametrization in the geoacoustic posterior probability density estimates. The algorithm is applied to a sequence of data sets collected on the Malta Plateau with small seabed footprints (~10 m), which facilitates examination of meso-scale seabed variability.

Contributed Papers

8:45

3aUW4. Geoacoustic inversion from ambient noise data using a trans-dimensional Bayesian approach. Jorge E. Quijano, Stan E. Dosso, Jan Dettmer (SEOS, Univ. of Victoria, Bob Wright Ctr. A405, 3800 Finnerty Rd., Victoria, BC V8P 5C2, jorgeq@uvic.ca), Martin Siderius, and Lisa M. Zurk (Portland State Univ., Portland, OR 97201)

Geoacoustic inversion of seabed parameters from ambient noise in shallow water is a promising technique, with potential advantages over active survey methods such as low environmental impact, easier deployment procedures, and less restrictive hardware requirements. Bayesian inversion provides a framework to estimate the posterior probability density (PPD) of geoacoustic parameters. Parameter and uncertainty estimates can be obtained from PPD moments, such as the maximum a posteriori model, means, correlations, and marginal distributions. A fundamental step in the inversion is the selection of a model parametrization (i.e., number of seabed layers) consistent with the data information content. Recent developments in Bayesian inversion of seismic and active-source acoustic data have considered a trans-dimensional approach to model selection, where the number of model parameters is treated as unknown. Different models are sampled according to their support by the data, accounting for parametrization uncertainty in the geoacoustic parameter uncertainty estimates. This work applies a trans-dimensional reversible-jump Markov chain Monte Carlo algorithm to ambient noise reflection coefficient data. The approach is demonstrated with data collected on the Malta Plateau using a vertical line array. [Work supported by ONR.]

9:00

3aUW5. Sequential geoacoustic inversion for mobile and compact source–receiver configuration. Olivier Carriére and Jean-Pierre Hermand (Environ. Hydroacoustics lab., Université libre de Bruxelles, Ave. Fr. Roosevelt 50 CP 194/05, 1050 Brussels, Belgium, ocarrier@ulb.ac.be)

The development of light instrumentation for geoacoustic characterization requires higher-frequency transmissions and inversion methods suitable for mobile configurations that efficiently combine any *a priori* knowledge about the environment or the time-varying source–receiver geometry and the acoustic transmission measurements. Sequential filtering methods provide a framework to achieve these objectives. In this paper, repeated short-range acoustic transmissions (between 750 and 1500 m) acquired on a drifting 4-hydrophone array that spans a small part of the water column, are sequentially inverted in nonlinear Kalman filters. The sequential filtering approach is demonstrated on actual data from the MREA/BP07 sea trial, with a space-coherent processing of multitone signals and a phase-coherent processing of linearly frequency modulated signals, in low (250–800 Hz) and medium (800–1600 Hz) frequency bands. The sequential inversion of repeated acoustic transmissions shows a good agreement with hydrographic and geophysical data with more stable estimates than conventional meta-heuristic inversion results and a reduced computational cost. The effect of filter covariance tuning is also examined and monitored with statistical tests. For large propagation modeling uncertainty, extended Kalman filter and ensemble Kalman filter are of comparable accuracy in parameter tracking, but the ensemble method should be preferred to get reliable associated uncertainty estimates.

9:15

3aUW6. Dynamic tomography of a gravity wave at the interface of an ultrasonic fluid waveguide. Lenaic Bonneau, Christian Marandet, Philippe Roux (ISTerre, Maison des Gosciences, 1381 rue de la piscine, 38041 Grenoble, France), Barbara Nicolas, and Jérôme Mars (Gipsa Lab., BP 46, Cedex 9, 38402 Grenoble, France)

Based on single-scattering effects, the diffraction-based sensitivity kernels which make the link between the acoustic perturbations and the medium fluctuations have been extensively studied. This research has been recently extended to perturbations at the air–water interface in an ultrasonic waveguide that scales down with a 1-km-long, 50-m-deep ocean acoustic channel in the kilohertz regime. Using array processing between two source-receiver arrays, the sensitivity kernel for both time and amplitude has been experimentally measured and theoretically calculated for each acoustic echo reverberated in the waveguide in response to a small and local surface elevation. In the present work, an ultrasonic experiment is performed during which the waveguide transfer matrix between the source-receiver arrays is recorded 50 times per second while a low-amplitude gravity wave is generated at the water–air interface. The acoustic inversion performed from the surface sensitivity kernels permits to follow the gravity wave dynamics at any point of the surface between the source and the receiver arrays.

9:30

3aUW7. Estimation of seabed parameters in the presence of highly nonlinear internal waves. Jason D. Sagers and David P. Knobles (Appl. Res. Labs., Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758)

Nonlinear internal waves (NLIWs) can cause significant scattering of acoustic energy within the modal spectrum. Increased sensitivity to seabed parameters may occur in cases where the primary direction of modal scattering is from lower-angle modes into higher-angle modes, where the mode functions extend deeper into the sediment. In this work, the sensitivity of acoustic propagation to seabed parameters when NLIWs are present is investigated. Comparisons are made between experimental acoustic data from Shallow Water 2006 and modeled acoustic data. The challenge of specifying the temporo-spatial behavior of the water column is addressed so that seabed parameters can be explored.

9:45

3aUW8. Validation of statistical inference using towed source data. Steven A. Stotts, Robert A. Koch, and Jason D. Sagers (Appl. Res. Labs., Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758)

An idealized concept of inversion modeling assumes for a given model that environmental and geometrical parameter values can be determined to match similar measured parameter values. In practice, data-model mismatch

is inherent due to noise and the inability of the model to provide a one-to-one correspondence with the real environment. Several approaches have been devised to apply statistical inference to the inversion geoacoustic parameter values, but the uncertainties calculated for these values cannot be validated directly with ground truth measurements. However, tones with calibrated source levels in data received from a towed source can provide a direct assessment of whether the estimated source level uncertainties encompass the actual error in the estimated source levels. Examples from the Shallow Water '06 experiment will be discussed to illustrate the procedure.

10:00–10:15 Break

10:15

3aUW9. Inversion of shear properties in ocean sediments from numerical Scholte wave. Zhengliang Cao and Hefeng Dong (Dept. of Electron. and Telecommun., Norwegian Univ. of Sci. and Technol., O.S. Bragstads Plass 2B, 7491 Trondheim, caozhengliang@yahoo.com)

Shear properties in marine environment are important to be measured for the stability of sediments, the tomography of shallow layers, and the conversion of acoustic energy from water-borne into sea bottom. Although surface wave methods are widely used to determine the shear wave velocity profiles at sites, inversion methods of shear attenuation have to be tackled with other difficult problems. So, an inversion method by dispersion curves of surface wave is developed and tested with the numerical Scholte wave data. The method includes three steps: extract of dispersion curves from data, calculation of forward dispersion model, and linear inversion by match of them. To simultaneously estimate dispersion curve of velocity and attenuation, Prony's method is applied for multi-modes of surface wave. Based on the methods of finding complex roots, forward model of dispersion curves is calculated with so accuracy as to get reasonable Jacoby of inversion. Different to inversion of shear velocity, the effect of spread loss is considered and investigated for Scholte wave. The performance of inversion shear wave velocity and attenuation profiles in the upper sediment layers is also examined for multi-mode dispersion curves.

10:30

3aUW10. Broadband geoacoustic inversion on a horizontal line array. A. Tolstoy (1538 Hampton Hill Circle, McLean, VA 22101, atolstoy@ieee.org)

Shallow water bottom properties can be very difficult to estimate and are usually done so by means of vertical line arrays over a range of frequencies. This work will examine the behavior of the minimization broadband processor at low frequencies (25–100 Hz) for a variety of horizontal line arrays at endfire in simulated Shallow Water 2006 environments. Do we ever have enough sensitivity and resolution to estimate any of the bottom properties when using a horizontal array?

10:45

3aUW11. Remote-sensing of hydrodynamics in Tokyo Bay and Bakan Strait. Tokuo Yamamoto (Appl. Marine Phys. Div. RSMAS, Univ. of Miami, 4600 Rickenbacer Cswy, Miami, FL 33149, tyamamoto@bellsouth.net)

In the inertia range, turbulent energy is concentrated in eddies. The eddies interact, dissipate the kinetic energy and reduces the eddy size. The turbulent velocity fields are measured by inverting the acoustic wave attenuation by volume scattering. The eddy viscosity ϵ is the rate of kinetic energy dissipation of the turbulence. Acoustic wave dissipation is roughly proportional to (velocity intensity fluctuations) ϵ, 2/3. The Reynolds stresses are also induced by the velocity intensity fluctuations ϵ, 2/3. The Reynolds stresses take various forms, such as Rip currents, wave set-ups, etc. The acoustic wave volume scattering, transmission loss by volume scattering, turbulent fields, Reynolds stresses, and eddy viscosity fields are inverted from eight point transmission lines in two completely different turbulent flow fields. They are Tokyo Bay ($\epsilon \sim 0.02$) and Bakan Strait ($\epsilon \sim 1.2$). The Reynolds stresses are in Tokyo Bay ($\sim$33 kPa) and in Bakan Strait ($\sim$1200 kPa). The side by side comparisons of the two data sets reviles the quite complicated differences in their responses to turbulent current field which will be presented in this paper.

11:00

3aUW12. Efficient acoustic uncertainty estimation for uncertain source depth. Kevin R. James (2010 Lay Auto Lab., Univ. of Michigan, 1231 Beal Ave., Ann Arbor, MI 48109) and David R. Dowling (Univ. of Michigan, Ann Arbor, MI 48109)

Uncertainty in source depth is a prevalent source of uncertainty in underwater acoustic measurements and predictions. This uncertainty can arise from limited information about a source, or the movement of a source between measurement intervals, such as the bobbing of a ship's propeller near the ocean surface. This presentation describes the effect of source depth uncertainty on transmission loss predictions at frequencies from a few hundred hertz to a few kilo-hertz in both range-independent and range-dependent environments, as well as the relative contribution of source depth uncertainty compared to other likely sources of environmental uncertainty. Several approaches to estimating the resulting transmission loss uncertainty are considered, including direct simulation with Taylor series approximations, the field-shifting method, polynomial chaos expansions, and a reciprocity-based solution. In particular, it will be shown that a reciprocity-based algorithm provides exact results in range-independent environments with a single field calculation but can require additional calculations to obtain transmission loss versus range information in a range-dependent environment. Strategies for limiting the computational effort and possibilities for implementing the reciprocity approach within the US Navy's existing transmission loss uncertainty estimation technique, the Uncertainty Band algorithm, are presented. [Work sponsored by the Office of Naval Research, Code 322OA.]

11:15

3aUW13. Estimates of the water column sound-speed field in three dimensions using a linearized perturbative technique. Christopher M. Bender, Megan S. Ballard (Appl. Res. Labs., Univ. of Texas at Austin, Austin, TX 78713-8029), and Preston S. Wilson (Univ. of Texas at Austin, Austin, TX 78712-0292)

A perturbative inversion scheme [S. D. Rajan *et al*., J. Acoust. Soc. Am. **82**, 998–1017 (1987)] is applied to estimate the water-column sound-speed field in a three-dimensional volume of the shallow ocean. The input data to the inversion are estimates of modal travel time data calculated from measurements from a distributed network of sources and receivers. The differences in the data are used as the basis for updating, or perturbing, a background model through the inversion algorithm to arrive at a solution. According to this technique, the horizontal plane is divided into a grid of range-independent regions and an estimate of the depth-dependent sound-speed profile within each region is obtained. The linearized perturbative technique is particularly well-suited for the three-dimensional inverse problem as the solution typically obtained in less than 20 iterations. During this talk, the method will be validated using synthetic data which are representative of oceanographic range-dependence in shallow water environments. The effect of the density and distribution of sources and receivers on the resolution of the inversion results will be discussed. [Work supported by ARL:UT IRD.]

11:30

3aUW14. A new broadband matched field processor? A. Tolstoy (1538 Hampton Hill Circle, McLean VA 22101, atolstoy@ieee.org)

It is well known that the use of multiple frequencies helps to reduce uncertainty and to improve resolution for matched field processing. This can be extremely important for the estimation of geoacoustic (bottom) parameters and usually means averaging ambiguity surfaces either incoherently or coherently (if the source spectrum is known) for the linear or minimum variance processors. However, a new approach requires that only the minimum processor values be retained over all the frequencies considered. This presentation will show how this method improves resolution and sidelobe reduction (for the simple incoherent method) both for the linear and for the minimum variance processing. Of course, one never gets something for nothing. This improvement comes at the cost of using only those frequencies which show high-matched field values, usually the low frequencies (25–75 Hz), for geoacoustic inversion. If some frequency component is strongly degraded, then the method will not work. The behavior of this new

processor (to be called the "minimization broadband processor") is demonstrated on simulated shallow water data.

11:45

3aUW15. Bottom reflections from rough topography in the long-range ocean acoustic propagation experiment (LOAPEX). Ilya A. Udovydchenkov, Ralph A. Stephen, Timothy F. Duda (Woods Hole Oceanograph. Inst., Woods Hole, MA 02543), Peter F. Worcester, Matthew A. Dzieciuch (Univ. of California at San Diego, La Jolla, CA 92093), James A. Mercer, Rex K. Andrew (Univ. of Washington, Seattle, WA 98105), and Bruce M. Howe (Univ. of Hawaii at Manoa, Honolulu, HI 96822)

Data collected during the 2004 long-range ocean acoustic propagation experiment form a unique set of measurements containing information about absolute intensities and absolute travel times of acoustic pulses at long ranges. This work primarily focuses on sound reflected from the seafloor at the shortest transmission range of the experiment, approximately 50 km. At this range, bottom-reflected energy interferes with arrivals refracted by the sound channel. Stable bottom-reflected arrivals are seen from broadband transmissions from acoustic source at 800 m depth with 75 Hz center frequency, and from a source at 350 m depth with 68.2 Hz center frequency. Standard acoustic propagation modeling tools such as parabolic equation solvers can be successfully used to predict acoustic travel times and intensities of the observed bottom-reflected arrivals. Inclusion of range-dependent bathymetry is necessary to get an acceptable model-data fit. Although there is no direct ground truth for actual sub-bottom properties in the region, a good fit can be obtained with credible sub-bottom properties. The potential for using data of this type for geoacoustic property inversion in the deep ocean will be discussed. [Work supported by ONR.]

WEDNESDAY MORNING, 2 NOVEMBER 2011 DOVER, 8:30 TO 9:45 A.M.

Meeting of Accredited Standards Committee (ASC) S3 Bioacoustics

C. A. Champlin, Chair ASC S3
Univ. of Texas, Department of Communication Sciences & Disorders, CMA 2-200, Austin, TX 78712

G. J. Frye, Vice Chair ASC S3
Frye Electronics, Inc., P.O. Box 23391, Tigard OR 97281

Accredited Standards Committee S3 on Bioacoustics. Working group chairs will report on the status of standards under development. Consideration will be given to new standards that might be needed over the next few years. Open discussion of committee reports is encouraged.

People interested in attending the meeting of the TAGs for ISO/TC 43 Acoustics and IEC/TC 29 Electroacoustics, take note–those meetings will be held in conjunction with the Standards Plenary meeting at 9:00 a.m. on Tuesday, November 1, 2011.

Scope of S3: Standards, specifications, methods of measurement and test, and terminology in the fields of psychological and physiological acoustics, including aspects of general acoustics which pertain to biological safety, tolerance and comfort.