Meeting of Accredited Standards Committee (ASC) S12 Noise

R. D. Hellweg, Chair, S12
Hellweg Acoustics, 13 Pine Tree Road, Wellesley, MA 02482

W. J. Murphy, Vice Chair, S12
NIOSH, 4676 Columbia Parkway, Mail Stop C27, Cincinnati, OH 45226

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

People interested in attending the meeting of the TAG for ISO/TC 43/SC 1 Noise, take note - that meeting will be held in conjunction with the Standards Plenary meeting at 9:15 a.m. on Tuesday, 11 November 2008.

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

TUESDAY AFTERNOON, 11 NOVEMBER 2008 CARISSA ROOM, 2:15 TO 3:30 P.M.

Meeting of Accredited Standards Committee (ASC) S1 Acoustics

P. Battenberg, Chair, S1
Quest Technologies, Inc., 1060 Corporate Center Drive, Oconomowoc, WI 53066-4828

R. J. Peppin, Vice Chair, S1
Scantek, Inc., 7060 #L Oakland Mills Road, Columbia, MD 21046

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

People interested in attending the meeting of the TAGs for ISO/TC 43 Acoustics and IEC/TC 29 Electroacoustics, take note - those meetings will be held in conjunction with the Standards Plenary meeting at 9:15 a.m. on Tuesday, 11 November 2008.

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

Meeting of Accredited Standards Committee (ASC) S2 Mechanical Vibration and Shock

R. L. Eshleman, Chair, S2
Vibration Institute, 6262 Kingery Highway, Ste. 212, Willowbrook, IL 60527

A. T. Herfat, Vice Chair, S2
Emerson Climate Technologies, Inc., 1675 West Campbell Road, PO Box 669, Sidney, OH 45365 0669

Accredited Standards Committee S2 on Mechanical Vibration and Shock. Working group chairs will report on the status of various shock and vibration standards currently under development. Consideration will be given to new standards that might be needed over the next few years. Open discussion of committee reports is encouraged.

People interested in attending the meeting of the TAG for ISO/TC 108, Mechanical vibration, shock and condition monitoring, and its five subcommittees, take note - that meeting will be held in conjunction with the Standards Plenary meeting at 9:15 a.m. on Tuesday, 11 November 2008.

Scope of S2: Standards, specification, methods of measurement and test, and terminology in the field of mechanical vibration and shock, and condition monitoring and diagnostics of machines, including the effects of exposure to mechanical vibration and shock on humans, including those aspects which pertain to biological safety, tolerance and comfort.

TUESDAY AFTERNOON, 11 NOVEMBER 2008 LEGENDS 9, 1:00 TO 5:20 P.M.

Session 2pAA

Architectural Acoustics: Acoustics of Retrofitted Performance Spaces

Dana S. Hougland, Cochair
Shen Milsom Wilke, Inc., 1822 Blake St., Ste. 2A, Denver,CO 80202

David S. Woolworth, Cochair
Oxford Acoustics Inc., 356 CR 102, Oxford, MS 38655

Chair's Introduction—1:00

Invited Papers

1:05

2pAA1. Adding art to an artless space: case studies in refitting existing buildings as performance facilities. Gregory Miller, Evelyn Way, Byron Harrison, and Richard Talaske (TALASKE, 1033 South Boulevard, Oak Park, IL 60302)

Performing artists will find ways to present their work, often in spaces that they find and adapt for their own use. This paper will provide a number of historic examples of the use of "found spaces" for theatrical performance (specifically referencing the work of Peter Brook and CIRT), as well as contemporary case studies of buildings that have undergone significant renovations. Contemporary projects will include the Lookingglass Theatre (a former municipal pumping station), the Steppenwolf Garage (a parking structure), and the Lincoln Park Music Theatre (a former bank building).

1:25

2pAA2. Barons, barns, bombs: Innovative unlikely homes for the arts. Scott Pfeiffer (Threshold Acoust. LLC, 53 West Jackson Blvd. Ste., 1734, Chicago, IL 60604)

The School of the Art Institute, The Weston Playhouse, and The Peck School of the Arts all have made use of existing structures as augmentations of their facilities. Sustainability is served in these projects regardless of their pursuit of the U.S. Green Building Council Leadership in Energy and Environmental Design Certification. Adaptive reuse preserves the energy spent in the creation of the materials and the construction of the original building, as long as modern systems are introduced that allow the buildings to function efficiently.

In each case, the unique qualities that made the buildings successful in their original use were exploited in their redevelopment. Issues of adapting structure, mechanical systems, and adapting to the local environment all guide the setting of priorities in the acoustic design. Special attention is paid to developing flexible infrastructure since clearly the second (or sometimes the third) use of the building will not be its last. The special problems overcome in retrofitting the existing space for the performing arts are discussed.

1:45

2pAA3. Four buildings converted to performance spaces. David Woolworth (Oxford Acoust., 356 CR 102, Oxford, MS 38655, dave@oxfordacoustics.com)

This paper documents the conversion of two metal buildings and two brick buildings into performance spaces: a metal seafood restaurant into a church sanctuary, a cotton gin into an amplified music venue, a 1928 power house into a black box and gallery, and an office complex and restaurant into a 1200-person capacity amplified venue and movie theater. The challenges and trade offs will be discussed, and the coordination of room acoustics, mechanical noise, and sound systems. Methods of control of amplified low end are addressed for each venue in terms of isolation and absorption.

2:05

2pAA4. Sala São Paulo Concert Hall, 1998 to 2008. Jose Nepomuceno (Acustica Sonica, R. Fradique Coutinho, 955 sala 01, 05416-011 São Paulo, Brasil)

Sala São Paulo is the house of Orquestra Sinonica do Estado de São Paulo (OSESP), and it is an adaptive conversion of a portion of the Julio Prestes railway terminal in São Paulo, Brazil into a concert hall of 1509 seats. Estao Julio Prestes dates from 1938 and the renovation project began in 1997. The concert hall construction started in 1998, and it was opened in July 1999. Ten years after the construction started, Sala São Paulo is a great success. A moveable ceiling with 15 panels weighing 7.5 tons each provides unique sound adjustments. The system calls attention from musicians, from the public, and from researchers. This paper presents a discussion about Sala São Paulo achievements from 1998 to 2008. For example, before Sala São Paulo, acoustics used to start as soon as the architecture design was finished. After Sala São Paulo, this is no longer the rule. Items designed for this project were further developed and installed in other venues and are now commercially available. Moveable ceiling adjustments changed since the project, strongly influenced by listening experiences, showing limitations of judgment based solely on measurements. Construction issues are discussed.

2:25

2pAA5. Rugby Hall. Marshall Long (Marshall Long Acoust., 13636 Riverside Dr., Sherman Oaks, CA 91423)

Rugby Hall began as a flat floored rectangular room with a low ceiling, which had false wooden beams at right angles to the main axis of the room. There was a low-raised platform on the short wall at one end of the room which served as a makeshift stage. The room was used for lectures, plays, and musicals and performed poorly for all uses. After much puzzling it was decided to excavate the floor to yield raked seating and an expanded stage area. The false beams were removed and the ceiling was raised. The new ceiling had to be designed around the large steel beams above and a unique sidewall design was developed. The resulting hall which seats 386 has served the school for over 20 years as a mixed use facility.

2:45

2pAA6. A new performance space for Temple University in the oldest building on campus. David Greenberg (Creative Acoust., LLC, 5 Inwood Ln., Westport, CT 06880) and Felicia Doggett (Metropolitan Acoust., LLC, Philadelphia, PA 19118)

Construction of the Baptist Temple on North Broad Street in Philadelphia was begun in 1899 and completed the following year. Temple University was founded within its walls and consequently where it got its name. The University purchased the building in 1972 where it remained vacant for several decades. In 2007, Temple University announced plans to adapt the structure as a multipurpose venue for orchestra and choral performances, jazz ensembles, drama, lectures, and convocations. The building includes the main worship space located above the Chapel of the Four Chaplains in the basement, both of which will be renovated into performance spaces. How does one renovate a wood-framed, volume-limited, noisy street side building with deep wraparound balconies and less-than-ideal mechanical equipment locations into a performance venue for a high-quality music department on a limited budget with demanding users and an ultra-cost-conscious owner while maintaining the historic fabric of the building? This presentation explores not only the details of the renovations but also the complicated relationships between the owner, the user, the architect, the acoustical consultant, and the peer reviewer.

3:05—3:20 Break

3:20

2pAA7. Retrofitting a performance space: Everyman Theatre. Julie Fischer (Shen Milsom Wilke, 3300 N Fairfax Dr., Ste. 302, Arlington, VA 22201)

The new home of the Everyman Theatre company opened in 1910 as a Vaudeville house and has had many reincarnations throughout its lifetime, including a parking garage and 1550-seat movie theater. In its latest reincarnation, the space under renovation was a large movie theater. A challenge for the project team has been to take this configuration and turn it into something that fits the needs of the Everyman Theater company. The current renovation is underway and will include a large studio theater, with approximately 250 seats and several different configurations including a typical end stage, thrust stage, and arena stage, a Blackbox theater, and support spaces such as classrooms, offices, and rehearsal space. One of the main issues related to the pre-existing space is its current structure. The current design has two theaters stacked vertically and there is concern that noise and vibration from activities from the upper theater

could disturb occupants in the lower theater. Upgrading the structure is complicated by the location of the building which sits directly above a metro tunnel and therefore has strict weight requirements. The approach currently proposed is to include a floating concrete floor and sound barrier ceiling. [Chris Pollock (SMW), Rima Namek and Diane Cho (Cho Benn Holback), Millie Dixon and John Coyne (Theatre Projects), and Vincent Lancisi (Everyman Theatre).]

3:40

2pAA8. 939 Cafe at Berklee College of Music: From an old office to a performing space. Ioana Pieleanu (Acentech Inc., 33 Moulton St., Cambridge, MA 02138)

Long due for the world renowned jazz and contemporary music school, and open in October 2007, the 939 Cafe is the first club-type performing space at Berklee College of Music in Boston. Given the urban location of the college and the shortage of available real estate in the area, the Cafe is a retrofit of an old architect's office space. Among the Cafe's neighbors in the building there is the Bostons Tennis and Racket Club, a building tenant since 1902, and an independent restaurant/bar. All aspects of the acoustical design, including the room acoustics, sound isolation, and mechanical system noise control, presented challenges in this limited volume historical setting. This presentation discusses these challenges, limitations, and the proposed acoustical solutions.

4:00

2pAA9. Acoustic challenges of converting a historical hall to multipurpose use. Anthony Bontomase and Kristin Bleedorn (Shen Milsom Wilke, 417 Fifth Ave., New York, NY 10016, abontomase@smwinc.com)

The Great Hall of the historical Cunard Building in New York City was originally used for the Cunard passenger ship lines ticketing beginning in 1921 and later converted to a US Post Office branch. After years of vacancy, a hospitality company proposed to take advantage of the attractive Renaissance Revival inspired hall for events ranging from speeches to weddings and concerts. The Great Hall features a huge vaulted ceiling and is surrounded by occupied office spaces, with original windows between. The acoustic goals were to reduce sound transmission into offices and to tame the reverberation time to accommodate various programs. Several acoustic tests were performed to establish the reverberation time and the noise isolation class between the hall and offices. Further testing with windows covered overyielded the achievable noise isolation without modification of walls within office spaces. Only nonpermanent interior acoustic treatments were permitted as preserving the visual appeal of the interior is critical. Insulating glass additions on the interior of offices is expected to provide a noise isolation class above 50. The use of carpet and suspended drapery is expected to reduce the reverberation time to be more manageable for a variety of programs.

Contributed Papers

4:20

2pAA10. Recording studio rehabilitation. Marshall Long (Marshall Long Acoust., 13636 Riverside Dr., Sherman Oaks, CA 91423)

From time to time projects are designed and built, which do not perform in the expected manner. Sometimes outside events interfere with the design intent. In other cases there is a fundamental misunderstanding of the technical principles involved. This is an example of one of these cases. A recording studio was designed (by a well known studio designer, who shall remain nameless) and built on the first floor of a multistory high rise in Los Angeles, CA. It consisted of a studio approximately $20\times18\times10$ ft^3 high and an adjacent control room. A 3-in. concrete floor was isolated on 1/2-in. rubber and the walls and ceiling were built independently on top of the floor. After completion both footfall and carts being pushed along a corridor approximately 15 ft away were clearly audible inside the studio. This paper describes the steps taken to analyze and correct the problem.

4:35

2pAA11. Multithread implementation for calculating room impulse responses. Wolfgang Ahnert, Stefan Feistel, and Alexandru Miron (Ahnert Feistel Media Group Berlin, Arkonastr. 45-49, 13189 Berlin, Germany, wahnert@ada-acousticdesign.de)

Modern simulation programs are committed to calculate full impulse responses instead of doing simple mapping presentations. The main problem here is how to obtain such IRs quickly but accurately. The usual algorithm simulates sound propagation by tracing a statistical ensemble of acoustic rays. In this contribution, we deal with a new ray-triangle intersection algorithm, based on a three-dimensional grid scheme, that provides significant performance improvement for ray tracing. We report about the introduction of a uniform grid ray-tracing algorithm, which is optimized for acoustic models and can be up to five times faster (standard Pentium 4) than hierarchical space decomposition. Furthermore, the algorithm has been parallelized, taking advantage of the inherently independent propagation of distinct rays. On multicore processors, the computation employs a variable number of thread tracing groups of rays in parallel. The speed increase is almost linear with the number of cores. An overview is given to explain this approach. To support this fast calculation, an additional algorithm derives a suggested particle number and length from the mean free path properties of the room and the desired detection rate at the receiver. So time-consuming guesses about the selecting particle numbers will be avoided.

4:50

2pAA12. Swept sine against maximum length sequences in room acoustics with music signals as background noise. Joel Paulo (DEETC, ISEL -Tecnical Inst. of Lisbon, Rua Conselheiro Emídio Navarro 1, Lisbon, 1959-007, Portugal, jpaulo@deetc.isel.ipl.pt) and José Bento Coelho (CAPS - Instituto Superior Tcnico, Lisbon, Portugal)

The Swept Sine and the MLS techniques are very popular in room acoustic measurement set-ups. Advantages and disadvantages of both methods have been well investigated and can be found in the literature. However, information regarding the performance of these techniques in the presence of high background music levels is scarce. Since the estimation of the room impulse response is based on the correlation between signals, the likelihood between the test signal and the music contents plays an important role on the results' accuracy. This paper explores these issues by taking into account the semantic information of the music signals when used as disturbance. The method used for the assessment of the gain between the two techniques consists of splitting each frame into segments and applying a weighting function depending on a likelihood function. The features used for the likelihood function are the rms value of each segment, spectral energy envelope relation, bandwidth and harmonic structure. Several examples are presented for comparison of the performance of the Swept Sine and the MLS techniques. Advantages and disadvantages of each technique are discussed for music signals as noise.

5:05

2pAA13. Predictions of sound energy flows in coupled spaces using a diffusion equation model. Yun Jing and Ning Xiang (Graduate Program in Architectural Acoust., School of Architecture, Rensselaer Polytechnic Inst., Troy, NY 12180)

In this work, a diffusion equation model [J. Xiang, J. Acoust. Soc. Am., **123**, 145–153 (2008)] is applied to two coupled spaces to study the time-dependent sound energy flows through the coupling area (aperture). The energy feedback is found when the primary room is less reverberant than the secondary room. The Bayesian framework is applied to quantify double-slope characteristics of sound-energy flow decays obtained from the diffusion equation numerical simulations. This work also reveals that the turning point on a double-sloped energy decay is highly correlated to the time instant when the energy flow direction flips over.

TUESDAY AFTERNOON, 11 NOVEMBER 2008 LEGENDS 2, 1:30 TO 4:55 P.M.

Session 2pAB

Animal Bioacoustics: Marine Mammal Acoustics in Honor of Sam Ridgway II

James J. Finneran, Chair
SPAWARSYSCEN San Diego, 53560 Hull St., San Diego, CA 92152

Invited Papers

1:30

2pAB1. Prey capture by harbor porpoises. Lee Miller (Inst. of Biology, Univ. of Southern Denmark, Campusvej 55, DK-5230 Odense M., Denmark, lee@biology.sdu.dk)

The harbor porpoise (*Phocoena phocoena*) is a small toothed whale living mostly in coastal waters. There are large, but unknown, numbers in the inner Danish waters. Four are in captivity at Fjord Belt Center, Kerteminde, Denmark, one of which was born there in 2006. Harbor porpoises use their ultrasonic clicks as biosonar for orientation and detection of prey (mostly smaller pelagic and bottom dwelling fish), and for communication. For studying wild animals, hydrophone arrays [Villadsgaard *et al.*, J. Exp. Biol. **210** (2007)] and acoustic (time/depth) tags [Akamatsu *et al.*, Deep Sea Res. **2** (2007)] have been used. For studying captive animals, arrays and video techniques [Verfuss *et al.*, J. Exp. Biol. **208** (2005)], as well as miniature acoustic-behavioral tags [Deruiter *et al.*, J. Acoust. Soc. Am **123** (2008)], have been used. While searching for prey, harbor porpoises use clicks at long intervals (50 ms) that progressively decrease when closing an object. After detecting the prey, the click interval stabilizes and then becomes progressively shorter while approaching the prey. The sequence ends in a terminal high-repetition rate buzz (500 clicks/s) just before capturing the prey (a video will be shown). The temporal sequence differs from that of beaked whales but is similar to that of bats.

1:50

2pAB2. Backscatter measurements of three species of salmon using simulated killer whale echolocation signals. Whitlow W. L. Au (Hawaii Inst. of Marine Biology, Univ. of Hawaii, P.O. Box 1106, Kailua, HI 96734), John K. Horne and Christopher D. Jones (Univ. of Washington, Seattle, WA 98105-6698)

The resident ecotype of killer whales (*Orcinus orca*) in the waters of British Columbia and Washington State has a strong preference for Chinook salmon even in months when Chinook are 5%–10% of the salmon population. The foraging behavior of killer whales suggests that they rely on echolocation to detect and recognize their prey. In order to determine possible cues in echoes from salmon, a series of backscatter measurements was made at the Applied Physics Laboratory, University of Washington Facility on Lake Union, on three different salmon species using simulated killer whale echolocation signals. The fish were tied to a monofilament net and rotated while echoes were collected, digitized, and stored on a laptop computer. Three transducer depths were used; same depth, 22^{o}, and 45^{o} above the horizontal plane of the fish. Echoes were collected from five Chinook, three Coho, and one Sockeye salmon. Radiograph images of all the specimens were also obtained to examine the swim bladder shape and orientation. Results show that the echo structure from similar sized but different species of salmon were different and probably recognizable by foraging killer whales. The results also suggest that a broadband echo-sounder pointing downward could be used to discriminate salmon species.

2:10

2pAB3. Progress in clinical hearing evaluation of bottlenose dolphins (*Tursiops truncatus*). James J. Finneran (U.S. Navy Marine Mammal Program, Space and Naval Warfare Systems Ctr., San Diego, Code 71510, 49620 Beluga Rd., San Diego, CA 92152) and Dorian S. Houser (Biomimetica, Santee, CA 92071)

It has long been recognized that the need to train experimental subjects is a major obstacle to large scale behavioral hearing studies of marine mammals. The time and level of access required for behavioral audiometry have not only prevented hearing assessment in wild individuals but have also limited attempts to assess auditory system fitness in captive populations. Ridgway and co-workers of electrophysiological methods were quick to realize that auditory evoked potential (AEP) measurements could be used to assess marine mammal hearing without lengthy training. Ridgway performed some of the earliest work to characterize dolphin evoked potentials and to develop fast techniques for threshold measurement. In addition to hearing assessment in larger whales not routinely kept in captivity, Ridgway has also strived to develop techniques that would allow routine clinical hearing assessment of captive animals. This talk reviews the development of AEP techniques for rapid large-scale hearing assessment in dolphins at the US Navy Marine Mammal Program. Topics range from Ridgway's early work on dolphin AEP threshold measurements to recent efforts to acquire longitudinal audiometric data from a large population of dolphins. [Work supported by ONR.]

2:30

2pAB4. Application of the evoked-potential method to the study of the odontocete biosonar. Alexander Supin (Inst. of Ecology and Evolution, Russian Acad. of Sci., 33 Leninsky Prosp., 119071 Moscow, Russia)

In a false killer whale *Pseudorca crassidens*, auditory evoked potentials (AEPs) were recorded in conditions of echolocation using both real targets and electronically synthesized and played-back (phantom) echoes. The electronic echoes were triggered by emitted sonar pulses, their spectra were similar to that of the emitted sonar clicks of the subject, and their intensities were proportional to the level of the emitted sonar pulse. AEPs to both the emitted pulse and real or electronic echoes were well detectable. AEPs related to emitted sonar clicks displayed the regular amplitude dependence on click level: The higher the level, the higher the amplitude. The echo-related AEP amplitudes depended on both echo attenuation and delay with a trade of around 9-dB attenuation per delay doubling. When the echo attenuation was varied in conjunction with delay keeping a rate of 9-dB deeper attenuation per delay doubling, the echo-related AEPs were nearly invariant. This result is well explainable by a hypothesis implying that partial forward masking of echoes by preceding emitted sonar pulses serves as a mechanism of automatic gain control in the auditory system of echolocating odontocetes.

2:50

2pAB5. Perception of echoes with FM1-FM2 delay disparities: Bats have selective direction-of-gaze high-resolution imaging. James A. Simmons (Dept. of Neurosci., Brown Univ., Providence, RI 02912, james-simmons@brown.edu), Mary E. Bates, Sarah A. Stamper and Douglas Benedicto (Brown Univ., Providence, RI 02912)

Echolocating big brown bats (*Eptesicus fuscus*) emit frequency-modulated (FM) biosonar sounds containing two prominent downsweeping harmonics, FM1 from 55 to 23 kHz and FM2 from 100 to 45 kHz. Acoustics of echoes in air ordinarily keep FM1 and FM2 aligned in time; however, if FM1 and FM2 are segregated artificially with low-pass and hig-hpass filters and delivered to bats at different delays, accuracy of delay perception is disrupted. As a direct confirmation of delay-resolution results from echo jitter experiments, harmonic delay offsets of 1–2 us are sufficient for the bat to reject echoes from accurate delay determination. Echoes ordinarily do arrive with differentially greater attenuation of FM2 relative to FM1 for longer ranges (FM2/FM1 is −3 dB for a range of 2 m) and for off-axis targets (FM2/FM1 added −3 dB for angles of 15 deg at a 2-m range). The resulting greater neuronal amplitude-latency trading for FM2 relative to FM1 activates the harmonic delay-offset effect to prevent accurate determination of delay. Noncoherence of harmonics in auditory neuronal spectrograms shunts information about the off-axis and far away targets to a different display of range and crossrange than for targets located in the direction of gaze. [Work supported by the ONR and the NIMH.]

3:10—3:25 Break

Contributed Papers

3:25

2pAB6. Underwater sound measurements of the Hawaii super ferry and the potential impact to humpback whales in the main Hawaiian Islands. Alison K. Stimpert, Alexis Rudd, and Whitlow W. L. Au (Inst. of Marine Biology, Univ. of Hawaii, P.O. Box 1106, Kailua, HI 96734)

Underwater measurements of the Hawaii Super Ferry were made with a vertical array of four hydrophones. The depths of the hydrophones were 3, 6.5, 10, and 20 m. The Alakai is a 107 m catamaran passenger and automobile ferry that has a waterjet propulsion system with no exposed propellers. The Alakai made three close proximity passes at speeds of 37, 24, and 12 kn on tracks that allowed for acoustic measurements of sounds radiating from the bow, broadside, and stern aspects. The track of the Alakai was recorded on global positioning system. The measurements were conducted while the ferry proceeded on normal scheduled trips, allowing for three measurements a day. All the sound pressure levels were referenced to 91 m (100 yards), the minimum legal approach distance of boats to humpback whales. Sound intensity increased with speed and was the highest at 37 kn in the broadside aspect (160 dB) and was 14 dB lower for both the bow and stern aspects. The sounds had a broadband noise quality with spectra that had a low frequency peak below 100 Hz and decreased continuously with frequency. The effects of the sounds will be discussed from a humpback whale perspective.

3:40

2pAB7. Patterns of coastal use by Hawaiian spinner dolphins (*Stenella longirostris*) observed using passive acoustic monitoring. Marc O. Lammers (Hawaii Institue of Marine Biology, P.O. Box 1106, Kailua, HI 96734, lammers@hawaii.edu), Simona Bernasconi (CIBRA, Universita Degli Study di Pavia, Pavia 27100, Italy), Whitlow W. L. Au (Hawaii Inst. of Marine Biology, Kaneohe, HI 96744), Kevin Wong, and Russell E. Brainard (Pacific Islands Fisheries Sci. Ctr., Honolulu, HI 96822)

Spinner dolphins are the most commonly sighted cetacean in coastal Hawaiian waters and are an important higher-level trophic component of the near-shore ecosystem. Establishing their long-term patterns of occurrence is important both for their conservation and for a better understanding of the relationships between dolphins, their prey, and other members of the ecosystem. Here we report on an effort to use ecological acoustic recorders (EARs) to monitor the presence of spinner dolphins along the leeward coast of the island of Oahu, Hawaii. The EAR is a bottom-moored recorder with a bandwidth of 30 kHz, which allows the detection of both dolphin whistles and echolocation clicks. Eight EARs were deployed along the coast at depths ranging from 15 to 50 m. Five units were deployed in an array, providing information about the EAR's detection range. The other three units were placed at sites along the coast commonly frequented by spinner dolphins. The results reveal distinct patterns of preference in habitat use, both for daytime resting behavior and night-time foraging. The recordings also provide a measure of vessel traffic at locations important to spinner dolphins. Combined, these data demonstrate the value of passive acoustic methods for monitoring cetacean populations and their habitat over extended periods.

3:55

2pAB8. Temporal and geographic patterns in the occurrence and distribution of minke whale (*Balaenoptera acutorostrata*) boings in the central and western North Pacific. Julie Oswald (Hawaii Inst. of Marine Biology, Univ. of Hawaii, 46-007 Lilipuna Rd., Kaneohe, HI 96744, jnoswald@hawaii.edu), Tom Norris (Bio-Waves Inc., Encinitas, CA 92024), Whitlow Au (Hawaii Inst. of Marine Biology, Kaneohe, HI 96744), and Fred Duennebier (Univ. of Hawaii, Honolulu, HI 96822)

Minke whales are elusive and difficult to study using visual methods. The source of the "boing" sound was recently linked to North Pacific minke whales, allowing passive acoustics to be used to study this species. The seasonal occurrence of minke whales was examined using data collected at the Station ALOHA Cabled Observatory, an ocean bottom hydrophone 100 km north of Oahu. Preliminary analysis of data collected between February and June 2007 indicates that boings occur during all of these months, peaking in early April. No diurnal variation was evident. Towed hydrophone-array surveys were conducted in the offshore waters of the islands of Oahu, Kauai and Ni'ihau (February 2005) and off Guam and the Northern Mariana Islands (January–April 2007). Although rarely observed visually, the prevalence of boings detected in these areas indicates that minke whales are common. Distribution patterns from both studies suggest that minke whales prefer deep but not the deepest waters. Boings recorded from Guam and the Northern Mariana Islands appear to be more similar to the "central" boing (which includes the Hawaiian Islands) than the "eastern" boing [which includes those recorded east of 138°W, Rankin and Barlow (2005)]. This has important implications for North Pacific minke whale stock structure.

4:10

2pAB9. Contextual sound production by tagged humpback whales (*Megaptera novaeangliae*) on a feeding and breeding ground. Alison K. Stimpert, Whitlow W. L. Au (Hawaii Inst. of Marine Biology Marine Mammal Res. Program, Dept. of Zoology, Univ. of Hawaii at Manoa, P.O. Box 1106, Kailua, HI 96734, stimpert@hawaii.edu), David N. Wiley (Stellwagen Bank Natl. Marine Sanctuary, Scituate, MA 02066), and David Mattila (Hawaiian Islands Humpback Whale Natl. Marine Sanctuary, Kihei, HI 96753)

Humpback whales are amongst the best studied of the baleen whales. The species is also known for its flexibility and variety of behavior on both feeding and breeding grounds. DTAGs were attached to 19 whales on the Hawaiian breeding grounds and to 24 whales on the northwestern Atlantic feeding grounds between the years of 2004 and 2008 to describe the variety of acoustic behavior in each location. Sounds were analyzed using ADOBE AUDITION, XBAT, and custom programs in MATLAB 7, and acoustic records from the tags showed differences in sound production between the two populations engaged in different activities. Recordings from the feeding grounds showed higher social sound production rates and also contained the sounds with the highest signal-to-noise ratios (10–20 dB for 1 μPa higher than on the breeding grounds). Differences in ambient noise may contribute to this: song chorusing is present in Hawaii, but commercial shipping traffic is higher in the northwest Atlantic feeding areas. Some similar sounds were recorded from the two areas, and exemplars of sound types from each location will be described. Overall, the feeding whales appeared to use sound more frequently, and perhaps for longer range communication than sounds produced within competitive groups in Hawaii.

4:25

2pAB10. The presence of low-frequency narrowband phonations in the wild bottlenose dolphins' acoustic repertoire. Natalija Lace and Stan Kuczaj (Dept. of Psych., Univ. of Southern Mississippi, P.O. Box 5025, 118 College Dr., Hattiesburg, MS 39406, kodzaks@yahoo.com)

Bottlenose dolphins' phonations are commonly divided into three major categories: whistles, echolocation clicks, and burst pulses. Other categories are often mentioned and described as yelps, squawks, barks, and low-frequency narrowband sounds. Here, we report the occurrence of low-frequency tonal phonations with fundamental frequency within the 500–2000 Hz range, accompanied by numerous harmonics. Recordings were made using 48 and 192 kHz sampling rates and a 100 Hz high-pass filter in the presence of free-ranging bottlenose dolphins in Mississippi Sound, MS. Auditory sensitivity of the bottlenose dolphins has been studied extensively, and the results indicate that dolphins have their best hearing sensitivity in the higher-frequency range (15–110 kHz). Low-frequency sounds (above 75 Hz) can be detected as well, and it has been suggested that this detection mechanism may be entirely different from that used for higher frequencies and may even include mechanoreception. The occurrence of low-frequency phonations in wild dolphins indicates that low-frequency detection may play an important role in the animals' everyday activities.

4:40

2pAB11. Spatial distribution of right whale "gunshot" sound displays in the Bay of Fundy, Canada. Susan E. Parks, Cara F. Hotchkin (Appl. Res. Lab., The Penn State Univ., P.O. Box 30, State College, PA 16804, sep20@psu.edu), and Christopher W. Clark (Cornell Univ., Ithaca, NY 14850)

North Atlantic right whales (*Eubalaena glacialis*) produce a distinctive loud broadband signal referred to as the "gunshot" sound. Current hypotheses for the function of this signal include an agonistic threat signal between males, a male reproductive advertisement signal to female right whales, or a combination of both. This sound has been documented both in a social context in mixed sex surface active groups and by lone individual males in a stereotyped display. An array of five archival bottom mounted acoustic recorders was deployed in the Center of the North Atlantic Right Whale Conservation Area in the Bay of Fundy, Canada in August 2005. The five recorders were spaced 3–7 nm apart, allowing for localization of whales producing gunshot sounds within or near the array. These recordings were used to describe the regional distribution, spacing, movement patterns, and timing of gunshot sound displays produced by right whales over a two week period to investigate the potential function of this sound. Further investigations included both the diel trends in the sound production and evidence for interactions between individuals simultaneously producing these displays.

Session 2pBB

Biomedical Ultrasound/Bioresponse to Vibration: Microbubble Response and Modeling

Tyrone M. Porter, Chair
Boston Univ., Aerospace and Mechanical Eng., 110 Cummington St., Boston, MA 02215

Contributed Papers

1:30

2pBB1. Coupled two-phase model of a gas microbubble containing suspended light absorbing nonoparticles. Elisabetta Sassaroli, Brian E. O'Neill, and King C.P. Li (The Methodist Hospital Res. Inst., 6565 Fannin St, Houston, TX 77030, esassaroli@comcast.net)

A mathematical model of a micrometer size gas bubble containing nanometer size light absorbing nanoparticles is presented. A description of such a system can be obtained in terms of a coupled two-phase model with the solid particles in suspension in the gas phase. It is assumed that the suspension is diluted so that particle-particle interaction can be ignored. Because the heat exchange between the particles and the gas is of main interest in this calculation, it is assumed in first approximation that the gas and the particles have the same velocity and pressure. The pressure is assumed to be a function of time only. In this case, only the equations of continuity and energy for both the particulate phase and the gas phase are needed. The two-phase model is then solved in a linear approximation, and a system of four differential equations with constant coefficients is obtained. The system is diagonalized and the general solution of the coupled system is obtained. The general solution is a combination of exponential decaying oscillatory functions for the temperature of the two phases, the pressure, and the microbubble radial oscillations. It was found that the oscillatory behavior takes place in the megahertz range.

1:45

2pBB2. The effect of scatterer size distribution on the backscatter coefficient. Michael King (Bioacoustics Res. Lab., Dept. of Elec. and Comput. Eng., Univ. of Illinois at Urbana-Champaign, Urbana, IL 61801, mrking2@uiuc.edu), Ernest L. Madsen, Timothy J. Hall (Univ. of Wisconsin-Madison, Madison, WI 53706), Alexander Haak, Michael L. Oelze, and William D. O'Brien, Jr. (Univ. of Illinois at Urbana-Champaign, Urbana, IL 61801)

The bias and variance of mean scatterer size estimates made from backscatter coefficients might be reduced by incorporating an estimate of scatterer size distribution into the parameter estimation scheme. Current estimation schemes generally assume a single scatterer size representing the volume-weighted mean diameter of a distribution. When the mean interrogation frequency has a ka value near 0.8 for the mean scatterer size, and the size distribution is relatively narrow, the results are unbiased. However, consideration of the distribution of sizes may improve scatterer size estimation, especially at frequencies for which $ka > 1$. A phantom of 2% (dry weight) agar spheres (with diameters between 90 and 125 μm) in a 3.4% agar background was scanned over a broad frequency range using several transducers, and the backscatter coefficients were estimated from these data. When accounting for the size distribution, the mean square error between the measured and theoretical backscatter coefficients over the frequency range of 3–14 MHz was reduced by at least 50% in the size range of interest. These results suggest that accounting for the size distribution in backscatter models will significantly improve the ability to parametrize the backscatter coefficient. [Work supported by NIH CA111289]

2:00

2pBB3. Modeling of the dynamics of a coated microbubble confined in a blood vessel. Sergey Martynov, Eleanor Stride, and Nader Saffari (Dept. of Mech. Eng., Univ. College London, Torrington Pl., London WC1E 7JE, UK, e.stride@ntlworld.com)

Coated microbubbles have been extensively investigated as contrast agents for diagnostic ultrasound imaging and more recently for therapeutic applications such as targeted drug delivery. However, theoretical models for microbubble dynamics have previously been developed either for encapsulated bubbles in an infinite volume of fluid or for uncoated bubbles in a confined volume. In the present study, a numerical model is developed to explore the effects of both encapsulation and confinement in a blood vessel upon the microbubble response in the time domain. Both surfactant-coated microbubbles and polymeric microspheres are examined for a range of vessel:bubble diameter ratios and mechanical properties. The model is validated against other theoretical models of oscillating bubbles, and the theory of buckling of a thin spherical shell. It will be shown that even at low acoustic pressures (20 kPa) the radial oscillations of the bubble and the amplitude and spectrum of the radiated pressure field can be significantly modified as a result of confinement and that these effects are sensitive to the viscoelastic properties of the coating and the vessel. The implications for diagnostic and therapeutic applications of microbubbles will be discussed.

2:15

2pBB4. Definition of a cavitation index for real time monitoring during *in vitro* liposomal drug release. Lucie Somaglino, Guillaume Bouchoux, Jean-Louis Mestas, Adrien Matias, Jean-Yves Chapelon, and Cyril Lafon (Inserm, U556, 151 cours Albert Thomas, 69424 Lyon, cedex 03, France and Universit de Lyon, Lyon F-69003, France, lucie.somaglino@inserm.fr)

Drug release from liposomes can be activated by ultrasound and inertial cavitation is assumed to be the main involved mechanism. Broadband noise is known as a good indicator for such cavitation. The feasibility of assessing the extent of drug release by acoustic noise measurements has been investigated. A 1.17-MHz focused transducer (50-mm diameter and 50-mm focal length) was excited with tone-bursts (duty cycle: 10%–100%, PRF: 100 Hz–10 kHz, and Ispta $< 800\ \mathrm{W/cm^2}$). Emitted noise was registered with a broadband needle hydrophone and bandpass filtered between the fundamental and the first harmonic frequency for assessing the instantaneous inertial cavitation activity. A cavitation index (CI) was computed by integrating the filtered signal over time. This index was validated experimentally using terephthalate oxidized to fluorescent hydroxyterephthalate under the action of free hydroxyl radicals generated by inertial cavitation. The hydroxyterephthalate fluorescence and the CI were well correlated ($R^2 \simeq 0.92$). Experiments with liposomes were performed showing very good correlation between CI and drug release extent from liposomes. The definition of this CI allowed applying consistent doses of cavitation and performing comparative tests between different liposome formulations. [This work was funded by a grant from the Norwegian Research Council (NANOMAT programme). Epitarget AS, Norway is acknowledged for the supply of liposome samples.]

2:30

2pBB5. Time reversal acoustic receiver. Laurent Fillinger (Artann Labs., 1459 Lower Ferry Rd., Trenton, NJ 08618 and Stevens Inst. of Technologies, Hoboken, NJ 07030), Yegor Sinelnikov (ProRhythm, Ronkonkoma, NY 11779), Alexander Sutin, and Armen Sarvazyan (Artann Labs., Trenton, NJ 08618)

The time reversal acoustics (TRA) principle has been employed for focusing ultrasonic waves in various biomedical and industrial applications. TRA-based ultrasonic transmitters have numerous advantages over conventional methods of ultrasound focusing such as an ability to effectively focus and steer focal spot with a transmitter comprising even a single piezoelectric transducer. In this study it is shown that such TRA focusing systems with a small number of transducers can also work in the receiving mode and detect both active and passive targets with a spatial resolution close to a half wavelength. The feasibility tests were conducted using two types of TRA receivers. The first TRA receiver comprising a water-filled reverberator with a few 600 kHz piezoceramic elements attached was used for the detection of the signal radiated by a miniature (1 mm) emitter in a water tank. It was shown that the TRA receiver can detect the position of this point emitter with accuracy better than 1 mm. The second test was conducted with an acoustic reverberator made of series of thin randomly shaped membranes and a single disk transducer with a frequency of 3.3 MHz. This TRA receiver allowed the localization of a reflecting target illuminated by a plane acoustic wave with resolution also better than 1 mm.

2:45

2pBB6. Identifying the inertial cavitation threshold in a vessel phantom using focused ultrasound and microbubbles. Yao-Sheng Tung, James Choi, Shougang Wang (Dept. of Biomedical Eng., Columbia Univ., 351 Eng. Terrace, 1210 Amsterdam Ave., New York, NY 10027, YT2235@columbia.edu), Jameel Feshitan, Mark Borden and Elisa Konofagou (Columbia Univ., New York, NY 10027)

Unveiling the mechanism behind the blood brain barrier using focused ultrasound (FUS) and microbubbles is essential for brain molecular delivery. Here, *B*-mode imaging and rf signals were acquired to pinpoint the threshold for inertial cavitation during FUS with microbubbles. A cylindrical hole of 800 μm in diameter was generated inside a polyacrylamide gel to simulate a brain arterial vessel. Definity® (Lantheus Medical Imaging, USA) microbubbles with a 1.1–3.3 μm-diameter and (1,2-distearoyl-sn-glycero-3-phosphocholine) (DSPC) shelled microbubbles with a 1–2-μm-diameter were injected prior to sonication (frequency: 1.525 MHz; pulse length: 100 cycles; PRF: 1 kHz; pulse duration: 40 ms). The cavitation response was passively detected using a 7.5-MHz single-element transducer, confocal with the FUS transducer, and a one-dimensional linear array transducer placed perpendicular to the FUS beam. The broadband spectral response of the acquired rf signals and the *B*-mode images detected the occurrence and location of inertial cavitation, respectively. Findings indicated that the rarefactional pressure threshold was approximately 0.53 MPa for the definity and 0.69 MPa for the DSPC-shelled microbubbles. Further studies will vary the vessel diameter and microbubble size to determine their role in the pressure threshold. [This work was supported by NIH Grant No. R21EY018505 and NSF CAREER 0644713. The authors thank Jennifer Hui for bubble characterization.]

3:00

2pBB7. Single transmitter time reversal focusing characterization using cross-correlation method. Yegor Sinelnikov and Andrey Vedernikov (ProRhythm Inc., 105 Comac St., Ronkonkoma, NY 11770)

Acoustic focusing with reverberating acoustic cavities based on the time reversal acoustic principle has been demonstrated in a range of medical applications from three-dimensional imaging to the generation of high pressure therapeutic pulses for the destruction of kidney and gallbladder stones in the human body. We investigate acoustic focusing in a system comprised of a single transmitter and a layered membrane reverberator. The system is capable of achieving focal peak acoustic pressures up to 0.4 MPa with less than 1 mm focal spot and steering range up to 20 mm. Focusing in the frequency range between 0.5 and 4 MHz is investigated. The parametric investigation of various membrane configurations is performed by a conventional time reversal technique with a hydrophone. In addition, we propose the cross-correlation of hydrophone signals as a method of evaluating the system focusing properties without conducting the actual time reversal focusing experiment. The cross-correlation method predicts the focal spot dimensions for the layered reverberator single transmitter system that compares well with experiment and theory. An important implication is the possibility to use inverse filtering to construct signals of the desired waveform with a low spatial peak pressure amplitude that can enhance the cell's permeability toward macroparticle uptake.

3:15

2pBB8. Information theoretic approaches to ultrasonic system design: Beamforming with interative spatial filters. Nghia Nguyen (Dept. of ECE and Beckman Inst., Univ. of UIUC, 405 N. Mathews, Urbana, IL 61801, nnguyen6@uiuc.edu), Craig Abbey (Univ. of California at Santa Barbara, Santa Barbara, CA 93106), and Michael Insana (Univ. of UIUC, Urbana, IL 61801)

First-principles approaches to the design of medical ultrasonic imaging systems for specific visual tasks are being explored. The study focuses on breast cancer diagnosis and is based on the ideal observer concept for visual discrimination tasks, whereby clinical tasks are expressed mathematically as likelihood functions. Realistic approximations to the ideal strategy for each visual task are proposed to maximize the diagnostic information content of the images. Based on previous studies [Abbey *et al.*, IEEE Trans. Med. Imaging **25**, 198–209 (2006)], it is known that the Wiener filter approach to beamforming, derived as a stationary approximation of the ideal observer, is limited for low-contrast, large area lesion discrimination. In this study an adaptive iterative Wiener filter coupled to a segmentation algorithm is developed to optimally beamform and process radio-frequency (rf) signals. The segmentation is achieved by applying a Markov random field approach. Performance is measured by comparing proportion correct in human-observer studies with the ideal observer and its approximations as required for an estimate of discrimination efficiency. It is found that by adapting to local statistical properties of the rf signal the iterative Wiener filter increases visual discrimination efficiency where the Wiener filter and conventional beamformers fail. [Work supported by the NCI/NIH.]

Session 2pEA

Engineering Acoustics: Acoustic Engineering of Materials and Techniques

Thomas R. Howarth, Cochair
NAVSEA Newport, 1176 Howell St., Newport, RI 02841

Kim C. Benjamin, Cochair
NAVSEA Newport, Newport, RI 02841

Contributed Papers

1:30

2pEA1. Coupling coefficient of segmented stack piezoelectric transducers using high-coupling materials. Stephen C. Thompson, Richard J. Meyer, and Douglas C. Markley (Appl. Res. Lab., The Penn State Univ., P.O. Box 30, State College, PA 16804)

High-power piezoelectric transducers normally use a segmented stack of piezoelectric pieces. The pieces are arranged with alternating polarization and are driven by electrodes of alternating voltage polarity. This allows long stacks to be built from easily manufactured pieces and reduces the voltage required for full power operation. The common understanding among transducer designers is that the mechanical properties of the stack of n pieces are equivalent to those of a single piezoelectric piece of equal total length, while the electrical impedance is reduced by the factor n^2. For conventional designs with operating bandwidth less than a half octave, this is quite accurate. However, as new materials with high-electromechanical coupling make it possible to design high-power transducers with bandwidths greater than an octave, the use of segmented stacks must be reexamined. With high coupling materials, there is a significant reduction in effective coupling coefficient caused by stack segmentation. The effect is small for materials such as hard PZT, but is larger in at least some transducers built from materials with coupling coefficient above 80%. This paper will address the effects of stack design based on modeled and measured results. [Work supported by the Office of Naval Research.]

1:45

2pEA2. Galfenol material properties and impact on transducer design. Ryan S. Scott (Knowles Electronics, LLC, 1151 Maplewood Dr., Itasca, IL 60143), Richard J. Meyer, Jr., and Stephen C. Thompson (The Penn State Univ., State College, PA 16804)

A simple measurement technique to accurately characterize magnetostrictive materials such as Terfenol-D and Galfenol is required as they become more widely used. A technique was developed that measures the magnetostrictive d_{33}, $\mu_{33}{}^{T}$, $s_{22}{}^{H}$, and Q_m constants using a pair of Helmholtz coils, a laser vibrometer, and a sense coil. Unlike previous methods, this technique does not require that the magnetostrictive sample be built into a transducerlike apparatus. A matrix of magnetostrictive material properties as a function of magnetic bias and ac drive amplitude can be created using the developed method. To demonstrate the usefulness of this technique, stress annealed laminated Fe–18.4%Ga rods were tested. Property trends observed for these samples will be reported as well as their impact on transducer performance. [This work was sponsored by the Office of Naval Research (Contract No. N00014-06-1-0530) and by Etrema Products, Inc.]

2:00

2pEA3. Verification of a method for measuring magnetostrictive parameters for use in transducer design modeling. Scott Porter, Ryan Scott, Richard Meyer, Jr., and Stephen Thompson (Appl. Res. Lab., The Penn State Univ., State College, PA 16804, scott.porter@psu.edu)

Previous studies have initiated an investigation into a new method of measuring the d_{33}, $\mu_{33}{}^{T}$, and $s_{33}{}^{H}$ magnetostrictive constants of a given sample using a pair of Helmholtz coils, a laser vibrometer, and a sense coil. These parameters are calculated from measurements made as a function of dc magnetic bias and ac drive amplitude. This method contrasts with the traditional measurement technique which involves building the sample into a transducerlike device. The new method was demonstrated for laminated stress-annealed Galfenol rods of high aspect ratio. Measurements obtained by this method appear credible and demonstrate good repeatability. For verification, the authors have returned to the established approach and designed a simple transducer using one of the measured Galfenol rods. The transducer was driven in air as a function of dc magnetic bias and ac drive amplitude. Data obtained from this transducer are compared to an analytical model that uses the material properties obtained experimentally with the new method. [This work was sponsored by the Office of Naval Research (Contract N00014-06-1-0530) and by Etrema Products, Inc.]

2:15

2pEA4. Estimating the mechanical properties of acrylic plates via acoustic transmission experiments and modeling. Natasha A. Chang (Dept. of Mech. Eng., Univ. of Michigan, 1231 Beal Ave., 2010 Autolab, Ann Arbor, MI 48109) and David R. Dowling (Univ. of Michigan, Ann Arbor, MI 48109)

Clear acrylic [polymethylmethacrylate (PMMA)] is commonly used to separate instruments from the test-section flow in water tunnel experiments. Thus, for some hydroacoustic studies, knowledge of the sound transmission properties of acrylic plates may be essential. Unfortunately, the actual mechanical properties of PMMA plates cover a relatively large range (e.g., the elastic modulus can vary from 2 to 5.5 GPa) and these mechanical properties may be frequency dependent. In this presentation, numerical and experimental results for acoustic pulse transmission through submerged flat PMMA plates are compared to identify plate properties and to calibrate an approximate plate transmission model. The experiments were conducted in a fish tank at sound incidence angles of 0–35 deg with a spherical wave source and two receivers. The transmitted sounds were short pulses with a nominal frequency range of 40–200 kHz. Several plate thicknesses, d, were tested for kd values (k is the acoustic wave number in the water) from 0.25 to 16. A sound propagation code was used to calculate the transmission characteristics of the plate. By minimizing the difference between the calculated and measured waveforms, the mechanical properties of the PMMA plates were estimated. [Work sponsored by ONR.]

2:30

2pEA5. Two ultrasonic transducer through-wall communication system analysis. Henry A. Scarton (Dept. of Mech., Aerosp., Nuclear Eng., Rensselaer Polytechnic Inst., Troy, NY 12180)

The use of ultrasound to convey data from one side of a metallic wall to the other side is presented. A communication channel is established by attaching two ultrasonic crystals to either side of the wall. The outside transducer injects a continuous ultrasonic wave into the wall. The inside transducer operates as an energy harvester and signal modulator. The outside transducer also receives the modulated signal reflected back from the wall containing the inside transducer. A sensor on the inside provides analog data

2p TUE. PM

(e.g., temperature) that is then digitized. The digitized bits are used to vary the electrical load applied to the electrical terminals of the inside transducer by changing its acoustic impedance in accordance with the data bits. The impedance changes, in turn, modulate the amplitude of the reflected ultrasonic signal. This modulated signal is detected at the outside receiving (as well as transmitting) transducer, where it is then demodulated to recover the data. Additionally, some of the acoustic power received at the inside transducer is harvested to produce the electrical power needed to operate the communication and sensor system on the inside. Digital data communication rates exceeding 50 000 bits/s are achieved.

2:45

2pEA6. Measurement uncertainties caused by tapping machines in floor sound insulation tests. Valentin Buzduga (Scantek Inc., 7060 Oakland Mills Rd., Ste. L, Columbia, MD 21045, buzdugav@scantekinc.com)

This paper illustrates how the parameters of the tapping machine may influence the measurement accuracy when testing the impact sound insulation of floors. The analysis connects the specifications for tapping machines given in ISO 140-6 and ASTM E 492-4 standards with the requirements for repeatability and reproducibility of the sound insulation measurements given in ISO 140-2. The paper also discusses calibration and testability aspects for tapping machines and presents the method developed at Scantek for measuring the impact velocity of the hammers. The paper gives modeling equations, preliminary calculations for measurement uncertainties, and experimental results obtained at Scantek on calibrating the Norsonic tapping machine N-211.

3:00

2pEA7. Ultra-wide-band filter for noise control. Manvir Kushwaha (Inst. of Phys., Univ. of Puebla, P.O. Box J-45, 72570 Puebla, Mexico)

Extensive band structures are performed for two-dimensional periodic arrays of rigid stainless steel cylinders in air, with Bloch vector being perpendicular to the cylinders. We embark on the opening up of a complete acoustic band gap—independent of the polarization of the wave and of the direction of propagation. In addition, we propose the fabrication of a multiperiodic system in tandem that could create a huge hole in sound within the human audible range of frequencies.

3:15

2pEA8. Effect of sound amplitude on the acoustic attenuation characteristics of Helmholtz resonators. Asim Iqbal (Dept. of Mech. Eng. and The Ctr. for Automotive Res., The Ohio State Univ., 201 W. 19th Ave., Columbus, OH 43210, iqbal.27@osu.edu) and Ahmet Selamet (The Ohio State Univ., Columbus, OH 43210)

Influence of sound amplitude on the acoustic attenuation performance of a Helmholtz resonator is investigated computationally. Time-dependent and compressible flow field is determined by solving two-dimensional unsteady, turbulent, and compressible Navier–Stokes equations through an implicit and noniterative pressure-implicit-splitting-of-operators algorithm. The solution of full Navier–Stokes equations remedies the constraints in small-amplitude-wave treatments, thereby allowing a proper modeling of nonlinearities due to large amplitudes. Transmission loss of the Helmholtz resonator is then calculated for discrete inlet amplitudes using the pressure obtained from the flow field. Higher amplitudes are shown in this study to reduce the peak transmission loss significantly.

3:30

2pEA9. Experimental characterization of a biomimetic differential microphone diaphragm. Ronald Miles, Weili Cui (Dept. of Mech. Eng., SUNY at Binghamton, Binghamton, NY 13902-6000, miles@binghamton.edu), and Mihir Shetye (Solteras, City of Industry, CA 91748)

The identification of mechanical properties for a microphone diaphragm based on the coupled ears of the fly *Ormia ochracea* and fabricated out of polycrystalline silicon is described. An acoustic test setup using a laser vibrometer has been developed that facilitates the characterization of the diaphragm on a bare die level. A major problem with using a laser vibrometer for acoustic measurement of microstructures is the close working distance between the sensor head and test device, resulting in measurements corrupted by reflections. Time select windowing procedures are often used to obtain anechoic response estimates from measurements taken in reverberant environments, but are not effective for characterization at lower frequencies or testing of lightly damped structures where the time window length needs to be more than the reverberation time. It is shown that the reflections from the measured response of the biologically inspired diaphragm can be reduced through comparison methods for the calibration of the sound field using a closely placed probe microphone and a commercially available pressure differential microphone. Equivalent mechanical parameters for the diaphragm are estimated with a least squares identification procedure. Characterization results for a diaphragm with two different back volume configurations are compared. [Work supported by NIH.]

3:45

2pEA10. Testing microphones in small acoustical enclosures. Mariana Buzduga and Valentin Buzduga (Scantek, Inc., 7060 Oakland Mills Rd., Ste. L, Columbia, MD 21046, buzdugam@scantekinc.com)

For both instrument calibration and testing, there is a need for faster and more economical test methods, while keeping the test accuracy within acceptable limits. We analyze the use of a small acoustical enclosure that is not strictly "anechoic" but provides a test environment that can be used to make measurements with a predictably high degree of repeatability for testing the frequency characteristics of the microphones or the influence of their accessories (windscreen and grids). First, the characteristics of the acoustical field in the enclosure are discussed. Then the sensitivity of two test methods to specific influencing parameters is analyzed. The first method is the successive comparison using a reference microphone. The second method is a simultaneous comparison based on the constant divergence of the sound pressure in the field [V. Buzduga and M. Buzduga, "Acoustical test methods based on the constant divergence of the sound pressure level," Noise Control Eng. J., **51**(6), 343–347 (2003)]. The accuracy of the tests and the frequency domain of possible use for each method are discussed.

Session 2pMU

Musical Acoustics: Telematic Music Technology

Jonas Braasch, Cochair
Rensselaer Polytechnic Inst., School of Architecture, 110 8th St., Troy, NY 12180

Pauline Oliveros, Cochair
Rensselaer Polytechnic Inst., School of Architecture, 110 8th St., Troy, NY 12180

Invited Papers

1:30

2pMU1. An automated calibration system for telematic music applications. Jonas Braasch (CA^{3}RL, School of Architecture, Rensselaer Polytechnic Inst., Troy, NY 12180)

The complex polyphonic nature of music typically has stricter requirements for sound-pressure level calibration than back-and-forth natured speech dialogs. An onsite sound engineer typically controls the levels for electroacoustically enhanced music performances. In telematic applications, however, this task can easily become difficult because the sound engineer only has restricted access to the remote sites. This makes it difficult to balance the overall sound. For this reason, an automatic calibration system was developed. The system works with closely captured microphone signals, which are needed to avoid echo-feedback between two or more remote telepresence sites. For calibration purposes, a small-aperture pyramidal five-microphone array is mounted in the center of each remote site. The microphone array tracks the positions and sound-pressure levels of the individual instruments, based on an algorithm that can extract this information in the presence of concurrent sound sources [Braasch and Tranby, 19th ICA, Madrid, Spain (2007)]. The extracted data are then transmitted via OpenSound Control to the remote site(s) for accurate spatial reproduction of the original sound-source positions. The signals are calibrated such that they evoke the same sound-pressure levels at the microphone array of each site.

1:55

2pMU2. Synchronization and acoustics in network performance. Juan-Pablo Caceres (CCRMA, Dept. of Music, Stanford Univ., The Knoll, 660 Lomita Dr., Stanford, CA 94305, jcaceres@ccrma.stanford.edu)

The limitations that transmission delays impose on telecommunications are explored and used to the advantage of music making in network performances. This presentation discusses several strategies and practical applications that the SoundWIRE group at Stanford University has implemented in recent years. Performances with Belfast, New York, and Beijing are showcased. Strategies to synchronize remote musicians show how one performance can generate separate instances of a piece in different locations. The Internet's acoustic path (i.e., the delay in the sound transmissions) is used as part of musical effects and physical models that are embedded in the network. These processes make the distance "audible" for performers and support (and sometimes extend) musical ideas that otherwise would be impractical. Finally, a project with an application to attest the quality of service (QoS) in an audio connection is presented. Remote clients can "sound-ping" to a server running the JACKTRIP application and get an idea of its quality. Different physical model instruments can be used. The longer the distance, the lower the pitch. With higher variability in the connection, vibrato is heard. This technique enables musicians to tune a link using the most appropriate and simplest tool, their ears.

2:20

2pMU3. Haptic communication and colocated performance. Curtis Bahn, Jonas Braasch, Dane Kouttron, Kyle McDonald, and Pauline Oliveros (Dept. of the Arts, Rensselaer Polytechnic Inst., 110 8th St., Troy, NY, 12180, crb@rpi.edu)

How can experimental computer-mediated performance attain the very high levels of interpersonal intimacy and temporal intensity attained over millennia of artistic evolution involving traditional analog instruments and affordances? Current practice is lamentably missing key components of a hypothetical possibility space combining enhanced sensory perception with embodied skills inherited from traditional art forms. This presentation examines the development of the "Vibrobyte," a small inexpensive wireless interface for the exploration of haptic communication in computer mediated and colocated performance. The Vibrobyte was developed through a research seminar at Rensselaer Polytechnic Institute and is in its first stages of artistic application. We will present the technical capabilities of the new device and examples of its first applications. Future possibilities and plans for large-scale musical compositions incorporating the Vibrobyte will be introduced.

2:45

2pMU4. Creating systems for collaborative network-based digital music performance. Doug Van Nort (Music Technol., McGill Univ., 555 Sherbrooke St. West, Montreal, QC H3A 1E3, Canada)

The internet has proven to be an important catalyst in bringing together musicians for remote collaboration and performance. Existing technologies for network audio streaming possess varying degrees of technological transparency with regard to allowable bandwidth, latency, and software interface constraints, among other factors. In another realm of digital audio, the performance of "laptop music" presents a set of challenges with regard to human-computer and interperformer interaction—particularly in the context of improvisation. This paper discusses the limitations as well as newfound freedoms that can arise in the construction of musical performance systems that merge the paradigms of laptop music and network music. Several such systems are presented from personal work created over the past several years that consider the meaning of digital music collaboration, the experience of sound-making in remote physical spaces, and the challenge of improvising across time and space with limited visual feedback. These examples include shared audio processing over high-speed networks, shared control of locally generated sound synthesis, working with artifacts in low-bandwidth audio chat clients, and the use of evolutionary algorithms to guide group improvisations.

3:10—3:30 Break

3:30

2pMU5. Historical perspectives and ongoing developments in telematic performances. Scot Gresham-Lancaster (Media and Technol. Services, California State Univ. Hayward, 25800 Carlos Bee Blvd., Hayward, CA 94542)

This paper presents historical perspective on development 8232, and new technology applications for performances in which the author collaborated with several other dancers, musicians, and media artists to present synchronized colocated concerts at two or more sites. This work grew out of the author's participation in the landmark computer music ensemble, "The HUB." Each of the various performances was made possible by an evolving array of videoconferencing hardware and software. These will be discussed, as well a look back at the early history of "telematic" performance. The problems and interesting side effects presented by latency and dropouts are a unique part of this performance practice. This work leverages the concepts of shared space, and video and audio feedback generate with evolving forms created by the combinations of the space, the sounds, and the movements of the participants. The ubiquity of broadband Internet connections and the integration and constant improvement of videoconferencing software in modern operating systems make this unique mode of performance an essential area of research and development in new media performance. These new techniques will be discussed as well.

3:55

2pMU6. Telematic performance in mixed realities: Virtual and acoustic spaces. Pauline Oliveros (Arts Dept., Rensselaer Polytecnic Inst., 110 Federal St., Troy, NY 12680, olivep@rpi.edu)

Performers are always dealing with latencies. Some of the smaller latencies are long forgotten and integrated into performance practice. Spaces new to performers always present challenges. Halls can be too dry giving the performer insufficient feedback to create a warm instrumental sound or in the other extreme too wet so that the clarity of the sound is blurred in the reverberations. Networked performances between two or more locations present new challenges and latencies. The mixed realities of networked performance spaces will be explored in this paper. How can performers adjust to receiving from and projecting sound into more than one space simultaneously? Where is the focus point when two or more spaces meet in a network? How can the ambiguity of simultaneous spaces provide creative inspiration for new acoustical forms?

Contributed Papers

4:20

2pMU7. Constructing an integrated system for the practice of playing the guitar. Mamoru Ichise (Graduate School of Sci. and Technol., Ryukoku Univ., Japan), Norio Emura (Doshisha Univ., Japan), Masanobu Miura (Ryukoku Univ., Japan), and Masuzo Yanagida (Doshisha Univ., Japan)

Constructed here is an integrated system for the practice of playing the guitar for novice guitar players. Novice guitarists for popular music often give up playing it because they cannot keep practicing for the difficulty of realizing finger positions. An algorithm YG for giving optimal chord-form sequence for given chords was proposed [Emura *et al.*, AST, 104–107 (2006)]. Authors construct an integrated system for guitar practicing by implementing YG. The new system deals with thousands of MIDI files combined with names of chords for them. The proposed system shows chord sequences as the TAB score, with corresponding fingerings generated by YG as the chord diagram. By using the system, users are able to play excerpts using not the difficult but simple pattern of chord-form. The system interacts with the user playing the MIDI guitar by showing the performed notes with an accompanying playback for current excerpt. The user is expected to be supported on playing it under a performance evaluation, where errors of fingerings are simultaneously shown with accompanying playback stopping or slowing down. The performance of the system is confirmed as better than the usual style of practicing in terms of occurrence frequency of errors. [Work supported by HRC, Ryukoku University.]

4:35

2pMU8. New musical interface for playing the guitar solo. Sangjin Cho (School of Elec. Eng., Univ. of Ulsan, Ulsan 680-749, Korea, sjcho75@ulsan.ac.kr), Myeongsu Kang, and Uipil Chong (Univ. of Ulsan, Ulsan 680-749, Korea)

This paper describes the development of a new musical interface to play guitar solo with the nonstringed guitar. The previous work of authors is the implementation of the nonstringed guitar comprised of laser strings and a chord glove [J. Acoust. Soc. Am. **122**, 3055 (2007)]. It is hard to express guitar solo, so voltage dividers are set up on the neck to represent frets. In the case of the chord glove, the same voltage source of the TMS320F2812 is supplied instead of attaching force sensing registers to the fingertip. Consequently, there are three essential components for the proposed interface: voltage dividers, laser strings, and the chord glove. Voltage dividers replace the six strings on the neck and send fret information to DSP, laser strings represent strokes, and the chord glove expresses playing chords and solo. This proposed interface represents many kinds of playing styles. In the case of hammering on, pulling off, and sliding, it needs fret information from voltage divider to express these things. On the other hand, bending that varies the pitch in a fret needs to detect how much flex sensors are bent to play. [Work supported by the Korea Research Foundation Grant funded by the Korean Government(MOEHRD)(KRF-2006-521-H00002)]

Session 2pPA

Physical Acoustics: Topics in Nonlinear Acoustics

Zhiqu Lu, Cochair
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James M. Sabatier, Cochair
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Contributed Papers

1:00

2pPA1. Higher-order statistical analysis of nonlinearly propagated broadband noise. Micah R. Shepherd and Kent L. Gee (Dept of Phys. and Astronomy, Brigham Young Univ., N283 ESC, Provo, UT 84602)

Nonlinearity indicators have recently been used to identify and characterize nonlinear effects present in high-amplitude noise propagation. Although frequency-domain approaches have many benefits, a time-domain approach may be a more natural fit for the time-domain phenomenon of wave form steepening and shock formation. Standard skewness and kurtosis metrics are computed for propagated Gaussian noise at both low and high amplitudes. The same higher-order statistics are also computed for the first time derivative of the propagated noise in order to accentuate non-Gaussian variation in the wave form. The results appear to reveal an asymptotic behavior of the statistical quantities as the time wave form becomes more shock dominated.

1:15

2pPA2. Ultrasonic wave generation by means of highly nonlinear waves' sensor technology. Piervincenzo Rizzo (Dept. of Civil and Environ. Eng., Univ. of Pittsburgh, 942 Benedum Hall, 3700 O'Hara St., Pittsburgh, PA 15261, pir3@pitt.edu), Devvrath Khatri, and Chiara Daraio (California Inst. of Technol., Pasadena, CA 91125)

This paper describes an innovative approach to generate and sense pulse waves in structural materials. The approach is based on the generation of highly nonlinear solitary waves (HNSWs). HNSWs are stress waves that can form and travel in highly nonlinear systems (i.e., granular, layered, fibrous, or porous materials) with a finite spatial dimension independent of the wave amplitude. Compared to conventional linear waves, the generation of HNSWs does not rely on the use of electronic equipment (such as an arbitrary function generator) and on the response of piezoelectric crystals or other transduction mechanism. HNSWs possess unique tunable properties that provide a complete control over tailoring: (1) the choice of the waves' width (spatial size) for defects' investigation, (2) the composition of the excited train of waves (i.e., number and separation of the waves used for testing), and (3) their amplitude and velocity. HNSWs are excited onto concrete samples and steel rebar. The characteristics of the pulses traveling along simple waveguides such as metallic plates and rods are experimentally studied.

1:30

2pPA3. Nonlinear acoustic phenomena in viscous thermally relaxing fluids: Shock bifurcation and the emergence of diffusive solitons. Pedro Jordan (U.S. Naval Res. Lab., Stennis Space Ctr., Code 7181, MS 39529)

In this talk, we will consider the propagation of finite-amplitude acoustic waves in fluids that exhibit both viscosity and thermal relaxation. Under the assumption that the thermal flux vector is given by the Maxwell–Cattaneo law, which is a well known generalization of Fourier's law that includes the effects of thermal inertia, we derive the weakly nonlinear equation of motion in terms of the acoustic potential. We then use singular surface theory to determine how an input signal in the form of a shock wave evolves over time and for different values of the Mach number. Then, numerical methods are used to illustrate our analytical findings. In particular, it is shown that the shock amplitude exhibits a transcritical bifurcation; that a stable nonzero equilibrium solution is possible, and that a Taylor shock (i.e., a diffusive soliton), in the form of a "tanh" profile, can emerge from behind the input shock wave. Finally, an application related to the kinematic-wave theory of traffic flow is noted. [Work supported by ONR/NRL funding (PE 061153N)].

1:45

2pPA4. Intensity and temperature variance in sonoluminescence. Lyric Gillett (1821 Harvard St., Houston, TX 77008, mjcgillett@comcast.net)

Sonoluminescence is the process in which light is produced through ultrasonic pressure waves causing the expansion and subsequent collapse of a gas/vapor bubble, held in an ultrasonic standing wave pattern, in water. This study measured sonoluminescent bubble light intensity output utilizing a photomultiplier tube and a lock-in amplifier to determine the relationship of the sonoluminescent intensity to variance in water temperature. The results could be indicative of the optimal temperatures at which to conduct sonoluminescence research, and assist in facilitating and advancing sonoluminescence research ventures relating to further study of energy production and nuclear reactions. Though not part of the experimental design, the initial experimentation suggested a positive correlation between the drive level of the transducer and the light intensity from the sonoluminescent bubble. Further experimentation was conducted in which the transducer drive signal and frequency were continuously optimized as the water temperature rose. After data were graphed and analyzed, it was determined that there was a decrease in sonoluminescent intensity as the water temperature increased. It is believed that this experimentation resulted in a greater plasma density inside the sonoluminescent bubble. To that end, future experimentation, as well as potential military and medical applications, interests this researcher.

2:00

2pPA5. Nonlinear resonance frequency shifts in acoustical resonators with varying cross sections. Yurii A. Ilinskii, Mark F. Hamilton, and Evgenia A. Zabolotskaya (Appl. Res. Labs., The Univ. of Texas, P.O. Box 8029, Austin, TX 78713-8029)

The frequency response and nonlinear resonance frequency shift for an acoustical resonator with losses and having a varying cross section were investigated previously using Lagrangian mechanics for resonator shapes that are close to cylindrical [J. Acoust. Soc. Am. **110**, 109 (2001)]. The same approach is extended to include resonators having any shape for which a one-dimensional Webster-type equation is a valid model in the linear approximation. Admissible shapes include cones and bulbs proposed for acoustical compressors. The approach is appropriate for approximate but rapid parameter estimations for resonators with complicated shapes, requiring 100 times less computer time than for direct numerical solution of the one-dimensional model equation ordinarily used for these resonators [Ilinskii *et al.*, J. Acoust. Soc. Am. **104**, 2664 (1998)]. Results for cone and bulb shaped resonators with losses are compared with results from the previous one-dimensional theory and with experimental data. It is shown that the direction of the resonance frequency shift is determined by the efficiency of second-harmonic generation in modes having natural frequencies below versus above the frequency of the second harmonic, and how the net effect of this coupling compares with the frequency shifts due to cubic nonlinearity and static deformation.

2:15

2pPA6. Prediction and measurement of particle velocities in ultrasonic standing waves. Bart Lipkens, Jason Dionne, Alex Trask, Brian Szczur (Dept. of Mech. Eng., Western New England College, 1215 Wilbraham Rd., Springfield, MA 01119), and Ed Rietman (Physical Sci. Inc., Andover, MA 01810)

A numerical model has been developed to predict particle trajectories in ultrasonic standing waves. The model includes an electroacoustic model that calculates the characteristics of the one-dimensional standing wave as a function of the input voltage to the piezoelectric transducer driving the cavity. Next, the acoustic radiation force is calculated for particles residing within the water filled cavity. Finally, the particle trajectories are calculated through integration of the equations of motion of the particles. Particle translation is achieved through a periodic sweeping of the excitation frequency. Translational velocities of 6-μm-diameter polystyrene spheres are calculated for a 2-MHz standing wave driven by a PZT-4 transducer. In the experiment a cavity is filled with water and polystyrene particles. A PZT-4 transducer operates near its resonance frequency of 2 MHz. Through a periodic sweeping of the frequency the particles are translated away from the transducer face and ultimately clump together near the rigid reflector at the opposite end. The particle translational velocity is a function of the sweep parameters, i.e., frequency range and sweep period. The calculated particle velocities are then compared to the measured velocities. Needle hydrophone measurements are used to characterize the acoustic standing wave.

2:30

2pPA7. Computation of three-dimensional, pulsed, nonlinear acoustic wavefields from medical phased array transducers in very large domains. Jacob Huijssen, Martin D. Verweij (Lab. of Electromagnetic Res., Fac. of Elec. Eng., Mathematics and Comput. Sci., Delft Univ. of Technol., Mekelweg 4, 2628 CD Delft, The Netherlands, j.huijssen@tudelft.nl), and Nico De Jong (Erasmus Medical Ctr., 3015 GR Rotterdam, The Netherlands)

For the optimization and development of medical ultrasound transducers and imaging modalities, the iterative nonlinear contrast source (INCS) method has been developed. This numerical method predicts the nonlinear acoustic pressure field generated by a pulsed plane source with an arbitrary aperture and propagating in a three-dimensional tissuelike medium that extends over a very large domain of interest. The INCS method obtains the acoustic pressure from the nonlinear acoustic wave equation by treating the nonlinear term as a contrast source. The full nonlinear wave field is then found by iteratively solving the linearized wave problem using a Green's function method. By employing the filtered convolution method discussed in a companion paper, accurate field predictions are obtained at a discretization approaching two points per wavelength or period of the highest frequency of interest. In this paper, very large-scale nonlinear field profiles are presented for transducers with cylindrical as well as phased array geometries excited with a pulsed waveform having a center frequency of 1–2 MHz. Comparison with the results obtained from models of reduced complexity shows that in all cases the INCS method accurately predicts the nonlinear field. [Work supported by the STW and the NCF.]

2:45

2pPA8. A filtered convolution method for the numerical evaluation of very large-scale ultrasound problems. Martin Verweij, Jacob Huijssen (Lab. of Electromagnetic Res., Fac. of Elec. Eng., Mathematics and Comput. Sci., Delft Univ. of Technol., Mekelweg 4, 2628 CD Delft, The Netherlands, m.d.verweij@tudelft.nl), and Nico de Jong (Erasmus Medical Ctr., 3015 GR Rotterdam, The Netherlands)

The pulsed ultrasound field in nonlinear biomedical tissue may be computed by the recurrent evaluation of the field of a distributed contrast source in a linear homogeneous background medium. For human organs, the computational domain may easily measure hundreds of wavelengths or periods in each spatiotemporal dimension. Even with today's supercomputers, the evaluation of the transient field from the contrast sources in these very large inhomogeneous domains is a big challenge, and is only feasible if the applied numerical method is extremely efficient in terms of memory usage and computational speed. Using a Green's function method, the critical operation is a spatiotemporal convolution over the entire computational domain. It is shown how a systematic filtering and windowing of the involved functions yield accurate numerical convolutions with a discretization approaching the Nyquist limit of two points per wavelength or period of the highest frequency. This minimizes the storage requirement. Application of fast Fourier transforms for evaluation of the discrete convolution sum renders a method that is also computationally fast. The performance of a Neumann method equipped with these numerical convolutions is shown by presenting the computed transient wave field that traverses a very large inhomogeneous domain. [Work supported by the STW and the NCF.]

Session 2pPP

Psychological and Physiological Acoustics: Complex Sound Perception

Lawrence L. Feth, Chair
Ohio State Univ., Speech and Hearing Sci., 110 Pressey Hall, 1070 Carmack, Columbus, OH 43210-1372

Contributed Papers

3:00

2pPP1. Spectral weights and the transition between two auditory processes. Bruce Berg (Dept. of Cognitve Sci., Univ. of California, Irvine, 3151 Social Sci. Plaza, Irvine, CA 92697-5100, bgberg@uci.edu)

Evidence of broad auditory filters for processing temporal information was obtained with a molecular psychophysical technique. The standard in the 2IFC discrimination task consisted of n equal intensity sinusoids, with frequency separations ranging from 10 to 40 Hz. The signal was an intensity increment of the central sinusoid, which on average was 1000 Hz. Pitch cues were degraded by randomly selecting the center frequency over a 100 Hz range on each stimulus presentation. Single channel intensity cues were degraded with a roving-level procedure. As tones were added symmetrically to the ends of the complex, thresholds increased until a certain bandwidth was reached, beyond which thresholds decreased. Previous results showed that transition bandwidths ranged from 100 Hz to more than 1000 Hz across individuals [B. G. Berg, J. Acoust. Soc. Am. **121**, 3639–3645 (2007)]. This discontinuity presumably represents a change in the underlying auditory processes, with envelope cues at bandwidths narrower than the breakpoint and across channel level comparisons for wider bandwidths. Findings show that the pattern of spectral weight estimates before the breakpoint is consistent with an envelope processing model, whereas the weights for bandwidths greater than the transition point were consistent with an across-channel comparison. [Work supported by ONR].

3:15

2pPP2. Matching the temporal window and the fine-structure. William M. Hartmann (Dept. of Phys. and Astronomy, Michigan State Univ., East Lansing, MI, hartman2@msu.edu)

Research in psychological and physiological acoustics requires careful stimulus generation. This is especially true for research in binaural hearing because of the importance of interaural phase relationships. Knowing the stimulus precisely becomes even more important when the experiment employs reproducible noise because there is less averaging over stimulus variations. The discrete Fourier transform can be used to generate noise bursts with precise timing. Although the bursts are not band limited, the amplitude and phase spectra are well characterized within any chosen band if the fine-structure is matched to the temporal window. For the example, with a 100-ms rectangular window, the components are taken to be integer multiples of a fundamental frequency of 10 Hz. With this ideal fundamental frequency, the long-term spectrum is predictable. If the window is not rectangular, the ideal fundamental frequency depends on the temporal parameters of the window. The ideal frequency can be found from the Fourier transform of the window itself. For trapezoidal and Hann (raised-cosine) symmetrical windows, the ideal frequency turns out to be the reciprocal of the total (nonsilent) duration minus the rise time (only). Windows that are not symmetrical about their center lead to phase shifts and should be avoided.

3:30

2pPP3. Rapid efficient encoding of correlated complex acoustic properties. Christian E. Stilp, Timothy T. Rogers, Stephanie L. Jacobs, and Keith Kluender (Dept. of Psych., Univ. of Wisconsin, 1202 W. Johnson St., Madison, WI 53706, cestilp@wisc.edu)

Early sensory processing is long thought to extract and exploit redundancy and regularity in the input, but perceptual evidence supporting this hypothesis has been limited. Experiments are reported demonstrating efficient encoding of multiple acoustic attributes resulting in the reduction of perceptual dimensionality. Stimuli varied across two complex independent dimensions: attack/decay (AD) and spectral shape (SS). A subset of 24 stimuli captured a near-perfect correlation between these dimensions (r^2= 0.95). Participants passively listened to 25 presentations of these correlated stimuli over 7.5 min before discriminating sound pairs in AXB trials without feedback. For trials where both SS and AD varied, listeners successfully discriminated sounds consistent with the correlation of SS and AD during exposure, but they were at chance discriminating pairs varying orthogonal to the correlation presented in exposure. After exposure, participants were also significantly poorer at detecting differences among stimuli varying in only SS or AD. Thus, passive listening to stimuli with correlated attributes resulted in the collapse of two physically and perceptually distinct stimulus dimensions onto a single dimension that efficiently encoded exposure covariance. Data are consistent with Hebbian models of cortical adaptation but contradict models derived from anti-Hebbian and competitive-learning principles. [Work supported by NIDCD.]

3:45

2pPP4. Timbre perception. Lydia Hennig (Inst. of Musicology, Univ. of Hamburg, Neue Rabenstr. 13, 20354 Hamburg, Germany, lydia_muwi@web.de)

An electroencephalographic study dealing with timbre perception has been performed. Timbre is next to tone and chroma is the basic element of sound and is perceived as a multidimensional space in which perceptual coordinates correlate with acoustic parameters. To show these correlations and their associations with different brain areas, sounds all constant with respect to their fundamental pitch of 311 Hz were varied in timbre. Due to these variations of acoustical parameters clear differences could be found concerning the perceptional pathway and corresponding brain areas. The modification applied to the sounds included variations of attack time, spectral center of gravity, and even harmonic attenuation. Next to other findings a hierarchy of perception was found where variations of the attack time was overwritten by the even harmonic attenuation and the spectral centroit variations, but only if all three were varied simultaneously. In all other cases the attack time variations caused clear perceptional delays in the prefrontal lobe.

4:00—4:15 Break

4:15

2pPP5. Assessment of a computational model for auditory perception for rotary-wing aircraft using human responses. Evelyn Hoglund, Nandini Iyer, Douglas Brungart, Frank Mobley, and John Hall (Air Force Res. Labs., 2610 Seventh St., Bldg 441, Area B, Wright Patterson Air Force Base, OH 45433, evelyn.hoglund@wpafb.af.mil)

Traditional auditory perceptual models for detection of complex signals against complex ambient soundscapes are based on the human audibility threshold imposed upon computed representations of auditory critical band filters. Such models attempt to locate a positive signal-to-noise ratio in any critical band and then apply classic signal detection theory to derive detectability measures (d') and probability of detection values for the event. One limitation to these models is the limited experimental validation against human sound jury performance, especially using very low-frequency target

signals such as helicopters. This study compares computational auditory detection model predictions against a corresponding large sample of human sound jury data points obtained in the laboratory. Helicopter and ambient soundscape signals were obtained from high-sensitivity recordings in the field. Playback in the laboratory was achieved under high-fidelity headphones calibrated to accommodate helicopter primary rotor frequencies with minimal distortion above human sensation level. All listeners completed at least 12 000 trials detecting helicopters against rural and urban soundscapes to represent the spectrum of potential environments involved in a real world scenario. Analysis compares the human sound jury performance against a contemporary computational auditory detection model, called "AUDIB," developed by the U.S. Army and NASA.

4:30

2pPP6. Simulating the precedence effect by means of autocorrelation. Jonas Braasch (CA^{3}RL, School of Architecture, Rensselaer Polytechnic Inst., Troy, NY 12180)

A number of algorithms have been developed to simulate the robust localization performance of humans in reverberant conditions, the most successful of which are based on (contralateral) inhibition. While these models can demonstrate the precedence effect in general, their nonlinear behavior makes it difficult to optimize their settings. A linear algorithm has now been developed to overcome this limitation. An autocorrelation algorithm determines the delay between lead and lag and their amplitude ratio for both channels. An inverse filter is then used to eliminate the lag signal before it is localized with a standard localization algorithm. Interestingly, the filter contains both inhibitory and excitatory elements, and the filter's impulse response looks somewhat similar to the response of a chopper cell. The algorithm operates robustly on top of a model of the auditory periphery (gammatone filterbank and halfwave rectification). Due to its linear nature, the model performs better if the full waveform is reconstructed by subtracting a delayed version of the halfwave-rectified signal, with a delay time that corresponds to half the period of each frequency band's center frequency. The model is able to simulate a number of experiments with ongoing stimuli and performs robustly with onset-truncated and interaural-level-difference based stimuli.

4:45

2pPP7. Binaural loudness summation by cochlear implant users. Monika Kordus and Richard S. Tyler (Dept. Otolaryngol.-Head Neck Surgery, Univ. of Iowa Hospital Clinics, 200 Hawkins Dr., Iowa City, IA 52246, monika-kordus@uiowa.edu)

The loudness summation in people with bilateral and unilateral cochlear implants (CIs) was investigated. The stimuli were 1/3-octave bands of noise centered at frequencies of 0.25, 1, and 4 kHz presented frontally from a loudspeaker in an ascending or random sequence of level. Subjects rated different intensity levels between the threshold of hearing and level of discomfort on scales from 0 to 100. Eight intensity levels were used for normally hearing subjects who served as a control group. For the CI users, results showed difference in loudness between monaural and binaural conditions with substantial amount of binaural loudness summation. The shape of the intensity-loudness function varied between monaural and binaural conditions, depending on the listener. The power function of control group exhibited an exponent of near 0.5. In binaural condition, stimuli of the same sound pressure level were about 1.8 louder than in monaural condition. For 0.25- and 4-kHz bands of noise, no consistent differences between monaural and binaural thresholds were observed. For a 1-kHz band, the binaural loudness ratings were about 3% and 9% lower for mid- or high-intensity levels as compared to the monaural condition. The results suggest future application for fitting bilateral devices. [Work supported by NIH 5 P50 DC00242.]

TUESDAY AFTERNOON, 11 NOVEMBER 2008 LEGENDS 7, 1:30 TO 4:30 P.M.

Session 2pSC

Speech Communication: Prosody (Poster Session)

Rahul Shrivastav, Chair

Univ. of Florida, Comm. Sci. and Disorders, Dauer Hall, Gainesville, FL 32611

Contributed Papers

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

2pSC1. Passive exposure to speech stimuli facilitates within-category discrimination. Laurent Bonnasse-Gahot (Ctr. d'Analyse et de Mathématique Sociales (CAMS, UMR 8557 CNRS-EHESS), Ecole des Hautes Etudes en Sci. Sociales, 54 Bd. Raspail, Cedex 06, F-75270 Paris, France) and Jessica Maye (Northwestern Univ., Evanston, IL 60208)

Phonetic categories are known to induce good discrimination of stimuli from different categories and poor discrimination within category [Liberman *et al.*, J. Exp. Psych. **54**, 358–368 (1957)]. In addition, passive exposure to a phonetic continuum improves discrimination if stimuli are presented according to a bimodal distribution of occurrence frequency, whereas a unimodal frequency distribution results in decreased discriminability [Maye *et al.*, Dev. Sci. **11**, 122–134 (2008)]. The present study examined whether passive exposure to bimodal versus unimodal frequency distributions results in a categorical pattern of perception. Stimuli were tokens of the syllable [ma] with rising pitch contour, where pitch slope varied across the continuum. Participants completed an AX discrimination task both before and after passive familiarization to either a unimodal or bimodal distribution of the continuum stimuli. At post-test participants in the bimodal condition showed a significant improvement in discrimination and outperformed participants in the unimodal condition. However, contrary to the predictions of categorical perception, the bimodal group's discrimination improved most for stimuli that did not cross the category's boundary. These data suggest that short-term changes in speech perception after passive exposure may not reflect phonetic category learning but rather enhanced encoding of frequently occurring sounds.

2pSC2. Temporal structure and syntactic disambiguation across lexical biases. Yoon-Shil Jeon, Amy J. Schafer, and Victoria B. Anderson (Dept. of Linguist., Univ. of Hawaii, 1890 East-West Rd., Honolulu, HI 96822, yoonshil@hawaii.edu)

The present study investigates prosodic cues across three lexical biases in the production of syntactically ambiguous sentences such as "Walkers and runners with dogs use the outer portion of the park," in which "with dogs" can modify either the second conjunct alone (low association) or both conjuncts (high association). A phrase-combination task induced speakers to

produce high- and low-association tokens for each of 48 test sentences, split across three pretested lexical biases (high bias, equibias, and low-association bias) that were controlled for the number of syllables in the critical region. Results showed a significant lengthening difference between association conditions in the second noun, while the lengthening effect was minor in the first noun. In the high-association condition, the second noun region was significantly longer than the same region in the low-association condition. This is consistent with a stronger prosodic boundary at the end of the second conjunct for high-association productions than for low-association ones. The interaction of lexical bias and association was not statistically significant, suggesting that speakers' tendency to disambiguate through durational differences was not significantly affected by lexical bias.

2pSC3. On the reliability of rhythm metrics. Amalia Arvaniti, Tristie Ross, and Naja Ferjan (Dept. of Linguist., UCSD, 9500 Gilman Dr., La Jolla, CA 92093-D108, amalia@ling.ucsd.edu)

In the past decade or so, various metrics of vocalic and consonantal variability have been used to quantify linguistic rhythm, often yielding disparate results. The reliability of several such metrics (percentage of vocalic intervals and consonantal standard deviation, pairwise variability indices, and variation coefficient) was tested using materials from stress-timed English and German, syllable-timed Spanish and Italian, Korean, an unclassified language, and Greek, which has resisted classification. The materials for each language were divided into three subsets: an uncontrolled subset of sentences excerpted from a representative author of each language, a subset exhibiting as much as possible "stress-timing" properties (complex syllable structure and extensive vocalic variability), and a subset exhibiting as much as possible "syllable-timing" properties (simple syllable structure and limited vocalic variability). The results suggest that rhythmic scores can be severely affected by the choice of materials, especially in languages such as Italian, in which it is easy to avoid or accentuate variability (e.g., by excluding or including geminates). Variation coefficient scores were the most resistant to the manipulation of materials but failed to show statistical differences across most of the languages examined. The overall results cast doubt on the reliability of metric scores as indicators of timing and linguistic rhythm.

2pSC4. Impact of segmentation rules on the rhythm metrics. Diana Stojanovic (Dept. of Linguist., Univ. of Hawaii, 1890 East-West Rd., Moore Hall 569, Honnolulu, HI 96822)

Rhythm of speech in the literature of the past decade is quantified using measures of durations: first order, such as mean duration and percentage of the total sample, or second order, such as variance and pairwise variability index. Results of different studies, however, vary even when the same materials and speech mode are used and imply uncertainty of cross-study comparisons. Allen (1978) discussed the issue of validity (the criteria used for segmenting) and reliability (how accurately these criteria can be applied) on duration studies. The current paper examines how different segmentation criteria affect the five most commonly used rhythm metrics. All metrics are computed on the segmented sample of the read story (The North Wind and the Sun) for three speakers each of American English, Indonesian, and Serbo-Croatian. Different segmentation criteria were applied to (1) borders between voiceless plosives and vowels and (2) borders between vowels and trills. As a result, durations of some segments were changed in the range of 10–50 ms. The results show how different metrics are affected by the choice of two segmentation rules at two different speech rates. Implications for cross-linguistic studies are discussed.

2pSC5. A rough task: Defining a measure for rough voice quality. Rahul Shrivastav, David A. Eddins, Sona Patel, and Stacie Cummings (Dept. of Commun. Sci. and Disord., Univ. of Florida, Gainesville, FL 32611)

Perceptual judgments of voice quality on a rating scale or a magnitude estimation task are highly context dependent, making it difficult to compare data from one experiment to another. In previous work, a matching task using a sawtooth wave mixed with speech-shaped noise was found useful for quantifying breathiness in vowels [Patel *et al.*, 2006, JASA, **119**, 3340]. In the present experiment, we attempted to adapt this matching task for the estimation of roughness in vowels. Ten listeners participated in a matching task where listeners compared the roughness of a voice standard and the reference signal. A low-pass filtered sawtooth wave mixed with speech-shaped noise was used as the reference signal. The roughness of this signal was manipulated by amplitude modulating the waveform with a 40-Hz square wave. The modulation depth at which the signal and standard were perceived to be equally rough was used as an index of vowel roughness. Preliminary results show that listeners are able to use the matching task to estimate roughness in vowels, particularly for moderately to severely roughness voices. However, voices with little or no roughness are difficult to match to the signal because of the inherent timbre of the sawtooth.

2pSC6. Phonetic realization of prominence among lexical tones in Mandarin Chinese. Mingzhen Bao and Ratree Wayland (Program in Linguist., Univ. of Florida, Gainesville, FL 32601, joanneb@ufl.edu)

Linguistic prominence is defined as words or syllables perceived auditorily as standing out from their environment Terken (1994). It is explored through changes in pitch, duration, and loudness Ladd (1996). In this study, phonetic realization of prominence was investigated among lexical tones in Mandarin Chinese. The primary concern was to compare how accent and focus are acoustically realized under four conditions: (a) unaccented and unfocused, (b) accented but unfocused, (c) unaccented but focused, and (d) accented and focused, among four tones. Ten native speakers of Chinese were recorded reading materials in a natural fashion. The recorded data were segmented and acoustically measured for acoustic parameters: vowel duration; mean and maximum of intensity; and mean, maximum, minimum, and slope of $F0$. The results showed that vowel duration lengthening was the main acoustic parameter associated with accent, while an increase in vowel duration, mean and maximum of intensity and $F0$, and slope of $F0$ was associated with focus realization. It was also found that acoustic parameters used to realize focus were differentially ranked and varied from tone to tone. These results suggested that phonetic realization of prominence in Mandarin Chinese was affected by category of prominence (i.e., focus or accent) and tonal contexts.

2pSC7. The relationship between open quotient and $H1^*$-$H2^*$. Jody Kreiman (Div. of Head/Neck Surgery, UCLA School of Medicine, 31-24 Rehab Ctr., Los Angeles, CA 90095-1794), Markus Iseli, Juergen Neubauer, Yen-Liang Shue, Bruce R. Gerratt (UCLA School of Medicine, Los Angeles, CA 90095-1794), and Abeer Alwan (UCLA, Los Angeles, CA 90095)

It is widely assumed that changes in open quotient (OQ) produce corresponding changes in $H1^*$-$H2^*$, but empirical data supporting this relationship are scant. To provide such data, high-speed video images and audio signals were simultaneously recorded from six speakers producing the vowel /i/ while varying $F0$ from high to low and voice quality from pressed to breathy. Across speakers, the observed relationship between OQ and $H1^*$-$H2^*$ was much weaker than generally assumed. Patterns of covariation also differed substantially from speaker to speaker. Estimation of harmonic amplitudes was complicated by difficulties in determining the frequency of $F1$ when $F0$ was high and by uncertainties regarding the $F1$ bandwidth in the presence of a persistent glottal chink. Use of analysis-by-synthesis allowed correction of formant values, but bandwidth estimation remains problematic and will be discussed further at the conference. [Work supported by NIH Grant DC01797 and NSF Grant BCS-0720304.]

2pSC8. Music melody perception in tone-language- and nontone-language speakers. Jennifer A. Alexander, Ann R. Bradlow (Dept. of Linguist., Northwestern Univ., 2016 Sheridan Rd., Evanston, IL 60208), Richard D. Ashley, and Patrick C. M. Wong (Northwestern Univ., Evanston, IL 60208)

Speech and music both utilize pitch variation to convey contrasts in meaning: music-pitch expresses composition (e.g., the key of a piece) and effect; speech-pitch conveys pragmatic meaning and, in tone languages, lexical information. This study investigated whether experience with processing lexical pitch affects music pitch processing. 28 nonmusicians (14 native English speakers and 14 native Mandarin speakers) discriminated (experiment 1) and identified (experiment 2) short melodies. The Mandarin listeners more accurately discriminated the melodies than the English listeners (Mann-Whitney $U=140.5$, $p<0.05$; two-tailed $t(21.86)=2.45$, $p<0.05$, $d=0.93$), but the English listeners more accurately matched the melodies with graphical representations of the pitch movements that the mandarin listeners (Mann-Whitney $U=26.5$, $p<0.005$; two-tailed $t(25.44)=-3.94$, $p<0$ .001, $d=1.15$). This indicates that experience with lexical-pitch processing

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may enhance attention to pitch and thereby facilitate pitch-pattern discrimination. However, learned linguistic pitch-pattern categories may interfere with novel, nonlinguistic, pitch-patterns, thereby impairing identification of musical pitch-patterns. Our study contributes to a literature suggesting that processing of music-pitch and speech-pitch utilizes shared cognitive mechanisms (e.g., Alexander *et al.*, Interspeech 2005). Results are discussed with regard to a cognitive-processing framework that involves the influence of experientially acquired top-down pitch category information upon novel bottom-up pitch input during certain tasks. [Work supported by NU Cognitive Science Graduate Fellowship to J.A. and NIH Grants DC005794 (to A.B.) and HD051827 DC007468 (to P.W.)]

2pSC9. Similarities in the acoustic expression of emotions in English, German, Hindi, and Arabic. Marc Pell, Silke Paulmann, Chinar Dara, Areej Alasseri (School of Commun. Sci. and Disord., McGill Univ., Montreal, QC H3G 1A8, Canada), and Sonja Kotz (Max Planck Inst. for Human Cognit. and Brain Sci., Leipzig, Germany)

Based on the hypothesis that emotion expression is in large part biologically determined ("universal"), this study examined whether spoken utterances conveying seven emotions (anger, disgust, fear, sadness, happiness, surprise, and neutral) demonstrate similar acoustic patterns in four distinct languages (English, German, Hindi, and Arabic). Emotional pseudoutterances (the dirms are in the cindabal) were recorded by four native speakers of each language using an elicitation paradigm. Across languages, approximately 2500 utterances, which were perceptually identified as communicating the intended target emotion, were analyzed for three acoustic parameters: $f0$Mean, $f0$Range, and speaking rate. Combined variance in the three acoustic measures contributed significantly to differences among the seven emotions in each language, although $f0$Mean played the largest role for each language. Disgust, sadness, and neutral were always produced with a low $f0$Mean, whereas surprise (and usually fear and anger) exhibited an elevated $f0$Mean. Surprise displayed an extremely wide $f0$Range and disgust exhibited a much slower speaking rate than the other emotions in each language. Overall, the acoustic measures demonstrated many similarities among languages consistent with the notion of universal patterns of vocal emotion expression, although certain emotions were poorly predicted by the three acoustic measures and probably rely on additional acoustic parameters for perceptual recognition.

2pSC10. Acoustic correlates of vocal effort. Stephen Tasko (Dept. of Speech Pathol. and Audiol., Western Michigan Univ., 1903 W. Michigan Ave., Kalamazoo, MI 49008-5355, stephen.tasko@wmich.edu), Madalyn Parker (Kalamazoo Area Math and Sci. Ctr., Kalamazoo, MI 49008), and James Hillenbrand (Western Michigan Univ., Kalamazoo, MI 49008-5355)

This study evaluated the influence of open quotient (OQ), fundamental frequency ($F0$), and intensity on listener ratings of vocal effort for isolated synthetic vowels. Glottal waveforms (duration: 1000 ms) were synthesized using a model developed by Rosenberg [J. Acoust. Soc. Am. **49**, 583–590 (1971)]. Signals were generated using six different OQs ranging from 0.2 to 0.7 at each of eight distinct $F0$ values. Four $F0$ values were in a typically male range (100–142 Hz) and four in a typically female range (178–253 Hz). The glottal waveforms were passed through formant resonators with formant frequency settings characteristic of the vowel /a/. Signals were randomly presented at 80, 83, and 86 dBA to 20 listeners who used visual analog scales to rate each sample on dimensions of vocal effort, loudness, pitch, breathiness, and naturalness. Mean vocal effort ratings showed a weak negative association with OQ ($r=-0.34$) and weak-moderate positive associations with F0 ($r=0.45$) and intensity ($r=0.55$). Multiple regression analysis revealed that a linear combination of the three variables accounted for 68% of the variance in vocal effort ratings. These results suggest that the perception of vocal effort relies on multiple acoustic cues.

2pSC11. Perceived age in normal and disordered voices. Dorothy Bourgeois, W.S. Brown, Jr., Rahul Shrivastav, Howard Rothman, and James D. Harnsberger (Dept. of Commun. Sci. and Disord., Univ. of Florida, Gainesville, FL 32611)

An experiment was conducted to identify the contribution of voice quality to perceived age. Voice quality effects were assessed in age estimation experiments using (1) natural pathological stimuli that incorporated voice qualities of interest and (2) young normal voices in which two voice cues, hoarseness and tremor, were modified through resynthesis. The disordered samples included single sentences from 227 talkers included in the Kay Elemetrics database of disordered samples. The resynthesized samples were sentences from ten young males in which (1) the $f0$ contour was multiplied by a random number within a fixed range (hoarseness) or (2) a 5–9-Hz wave with an amplitude of 7.5% of the original $f0$ value was incorporated (tremor). Sixty navive listeners estimated all speakers ages in years. The results demonstrated that disordered voices were overestimated in age by 11 years, with the greatest mismatch in chronologic and perceived age reaching 47 years older. The addition of synthetic tremor and hoarseness shifted perceived age older by 8 and 5 years, respectively. In summary, voice quality appears to play a significant role in the perception of perceived age, comparable to or exceeding other cues that have been identified to date.

2pSC12. Perception of falling and rising pitch contour contrast: Effects of first language background. Ratree Wayland, Mingzhen Bao, and Edith Kaan (Program in Linguist., Univ. of Florida, Gainesville, FL 32607, ratree@ufl.edu)

The main objective of this study is to examine the ability to discriminate falling and rising pitch contour contrasts among native speakers of a tonal language (Chinese) and native speakers of a nontonal language (English). Linearly falling and rising pitch contours on [ba:] syllables are presented to participants in (1) the "same different categorial" discrimination task and (2) an "oddball" detection task. Preliminary results obtained from 10 Chinese and 12 English speakers suggested that (a) native Chinese speakers found the falling contour to be easier to discriminate than the rising contour, while the ability to discriminate between the two pitch contours was comparable among the English speakers, (b) reaction time for the falling contour false alarms was longer than for the rising contour among the English speakers, (c) no difference in either "detection" rate or "reaction" time between the two contours was found among both groups of speakers. More data will be collected from both groups of speakers and from speakers of another tonal language. Language general as well as language specific factors will be considered to account for the patterns of results obtained.

2pSC13. The production and perception of prenuclear second occurrence focus. Melissa Wake, Jason Bishop, and Cathryn Panganiban (Dept. of Linguist., Univ. of California, Los Angeles, 3125 Campbell Hall, Los Angeles, CA 90095, mwake@ucla.edu)

Semantic focus in English is typically marked by intonational prominence, most canonically by a pitch accent. One case that has been presented as an exception to this generalization, however, is that of second occurrence (SO) focus. An SO focus is a repeated but focused item, usually associated with a focus sensitive operator such as "only" or "even." Previous studies have suggested that SO foci lack pitch accents for phonological rather than information structural reasons, in most cases examining such foci in the postnuclear domain. We present acoustic and phonological data that demonstrate that SO foci also lack considerably in intonational prominence (particularly in terms of F0) when prenuclear, although they show increased duration. Additionally, the perceptibility of prenuclear SO focus is tested. Perceptually weak prosodic marking would suggest an important role for pragmatics in a focal interpretation, as has been suggested previously for SO focus.

2pSC14. Perceived prosody: Phonetic bases of prominence and boundaries. Jennifer Cole (Dept. of Linguist., Univ. of Illinois, 707 South Mathews, Urbana, IL 61801, jscole@uiuc.edu), Louis Goldstein, Argyro Katsika (Haskins Labs., New Haven, CT 06511), Yoonsook Mo (Dept. of Linguist., Univ. of Illinois, Urbana, IL 61801), Emily Nava, and Mark Tiede (Haskins Labs., New Haven, CT 06511)

In comprehending speech, listeners are sensitive to the prosodic features that signal the phrasing and the discourse salience of words (prominence). Findings from two experiments on prosody perception show that acoustic and articulatory kinematic properties of speech correlate with native listeners' perception of phrasing and prominence. Subjects in this study were 114 university-age adults (74 UIUC + 40 Haskins), monolingual speakers of American English who were untrained in prosody transcription. Subjects listened to short recorded excerpts (about 20 s) from two corpora of spontaneous and read speech (Buckeye Corpus and Wisconsin Microbeam Database) and marked prominent words and the location of phrase boundaries on a transcript. Intertranscriber agreement rates across subsets of 17–40 subjects are significantly above chance based on Fleiss' statistic, indicating that listeners' perception of prosody is reliable, with higher agreement rates for boundary perception than for prominence. Prosody perception varies across listeners (both corpora) and across speakers (WMD, where perceived prosody varies for the same utterance produced by different speakers). Acoustic measures from stressed vowels (Buckeye: duration, intensity, $F1$, $F2$) and articulatory kinematic measures (WMD) are correlated with the perceived prosodic features of the word. [Work supported by NSF.]

2pSC15. Perception of contrastive meaning through the L+H*L−H% contour. Heeyeon Y. Dennison, Amy J. Schafer, and Victoria B. Anderson (Dept. of Linguist., Univ. of Hawaii, 1890 East-West Rd., Honolulu, HI 96822, linguist@hawaii.edu)

This study establishes empirical evidence regarding listeners's perceptions of the contrastive tune [L+H*L−H%; e.g., Lee *et al.* (2007)]. Eighteen native English speakers heard three types of test sentences: (1) contrastive, "The mailbox was(L+H*) full(L−H%)," (2) positive neutral, "The mailbox(H*) was full(H*L−L%);" and (3) negated neutral, "The mailbox(H*) was not(H*) full(H*L−L%)." The participants first scored them by naturalness, and then typed continuation sentences based on the perceived meaning. Three other native English speakers independently coded the continuations to evaluate participants' interpretations of the test sentences. The results clearly demonstrated that the L+H*L−H% tune generated contrastive meanings (e.g., "…but the mailman took the mail and now it is empty" significantly more often than both the positive and negative neutral counterparts. Moreover, sentences presented in the contrastive tune were perceived as natural utterances. High coder agreement indicated a reliable function of the contrastive tune, conforming to the existing literature based on intuitive examples [e.g., Lee (1999)]. Interestingly, however, the contrastive tune produced the expected contrastive meaning in only about 60% of trials (versus less than 10% contrastive continuations for the other contours). This finding shows that the interpretation of the L+H*L−H% contour is more complex than previously suggested.

2pSC16. Order of presentation asymmetry in intonational contour discrimination in English. Hyekyung Hwang (Dept. of Linguist., McGill Univ., 1085 Dr. Penfield Ave., Montreal PQ H3A 1A7, Canada, hye.hwang@mail.mcgill.ca), Amy J. Schafer, and Victoria B. Anderson (Univ. of Hawaii, Honolulu, HI 96822)

In the work of Hwang *et al.* (2007), native English speakers showed overall poor accuracy in distinguishing initially rising versus level (e.g., L*L*H- H*L-L% vs L*L*L- H*L-L%) or initially falling versus level (e.g., H*H*L- H*L-L% vs H*H*H- H*L-L%) contour contrasts on English phrases in an AX discrimination task. Results not reported in that paper found that it was easier to discriminate when a more complex $F0$ contour occurred second than when it occurred first. Several orders of presentation effects in the perception of intonation have been reported (e.g., L. Morton (1997); S. Lintfert (2003); Cummins *et al.* (2006)] but no satisfying account has been provided. This study investigated these asymmetries more systematically. The order effect was significant for falling-level contrast pairs: pairs with a more complex $F0$ contour last were discriminated more easily than the reverse order. Rising versus level contrasts showed a similar tendency. The results thus extend intonational discrimination asymmetries to these additional contours. They suggest that the cause of the asymmetries may depend more on $F0$ complexity than on $F0$ peak.

2pSC17. Alternatives to $f0$ turning points in American English intonation. Jonathan Barnes (Dept. of Romance Studies, Boston Univ., 621 Commonwealth Ave., Boston, MA 02215, jabarnes@bu.edu), Nanette Veilleux (Dept of Comput. Sci., Simmons College, Boston, MA 02115, veilleux@simmons.edu), Alejna Brugos (Boston Univ., Boston, MA 02215, abrugos@bu.edu), and Stefanie Shattuck-Hufnagel (Res. Lab of Electrons, MIT, Cambridge, MA 02139, stef@speech.mit.edu)

Since the inception of the autosegmental-metrical approach to intonation (Bruce 1977, Pierrehumbert 1980, Ladd 1996), the location and scaling of $f0$ turning points have been used to characterize phonologically distinct $f0$ contours in various languages, including American English. This approach is undermined, however, by the difficulty listeners experience in perceiving differences in turning point location. Numerous studies have demonstrated either listener insensitivity to changes in turning point location or the capacity for other aspects of contour "shape" to override turning-point alignment for contour identification (Chen 2003, D'Imperio 2000, Niebuhr 2008). Even labelers with access to visual representations of the $f0$ encounter similar challenges. By contrast, a family of related measurements using area under the $f0$ curve to quantify differences in contour shape appear more robust. For example, a measure of the synchronization of the center of gravity of the accentual rise with the boundaries of the accented vowel yields 93.9% correct classification in a logistic regression model on a data set of 115 labeled utterances differing in pitch accent type. (L*+H L+H* in ToBI terminology). This classification proceeds entirely without explicit reference to the turning points (i.e., beginning of rise, peak) traditionally used to characterize this distinction.

2pSC18. Comparison of a child's fundamental frequencies during structured and unstructured activities: A case study. Eric Hunter (Natl. Ctr. for Voice and Speech, 1101 13th St., Denver, CO 80126, eric.hunter @ncvs2.org)

This case study investigates the difference between children's fundamental frequency (F_0) during structured and unstructured activities, building on the concept that task type influences F_0 values. A healthy male child (67 months) was evaluated (31 h, 4 days). During all activities, a National Center for Voice and Speech voice dosimeter was worn to measure long-term unstructured vocal usage. Four structured tasks from previous F_0 studies were also completed: (1) sustaining the vowel /ɑ/, (2) sustaining the vowel /ɑ/ embedded in a phrase-end word, (3) repeating a sentence, and (4) counting from one to ten. Mean F_0 during vocal tasks (≈257 Hz), as measured by the dosimeter and acoustic analysis of microphone data, matched the literature's average results for the child's age. However, the child's mean F_0 during unstructured activities was significantly higher (376 Hz). The mode and median of the vocal tasks were respectively 260 and 259 Hz, while the dosimeter's mode and median were 290 and 355 Hz. Results suggest that children produce significantly different voice patterns during clinical observations than in routine activities. Further, long-term F_0 distribution is not normal, making statistical mean an invalid measure for such. F_0 mode and median are suggested as two replacement parameters to convey basic information about F_0 usage.

2pSC19. Effects of acoustic cue manipulations on emotional prosody recognition. Chinar Dara and Marc Pell (School of Commun. Sci. and Disord., McGill Univ., 1266 Pine West, Montreal, QC H3G 1A8, Canada, chinar.dara@mail.mcgill.ca)

Studies on emotion recognition from prosody have largely focused on the role and effectiveness of isolated acoustic parameters and less is known about how information from these cues is perceived and combined to infer emotional meaning. To better understand how acoustic cues influence recognition of discrete emotions from voice, this study investigated how listeners perceptually combine information from two critical acoustic cues, pitch and speech rate, to identify emotions. For all the utterances, pitch and speech rate measures of the whole utterance were independently manipulated by factors of 1.25 (+25%) and 0.75 (−25%). To examine the influence of one cue with reference to the other cue the three manipulations of pitch (+25%, 0%, and −25%) were crossed with the three manipulations of speech rate (+25%, 0%, and −25%). Pseudoutterances spoken in five emotional tones (happy, sad, angry, fear, and disgust) and neutral that have undergone acoustic cue manipulations were presented to 15 male and 15 female participants for an emotion identification task. Results indicated that both pitch and speech rate are important acoustic parameters to identify emotions and more critically, it is the relative weight of each cue which seems to contribute significantly for categorizing happy, sad, fear, and neutral.

2pSC20. Perception of emphasis in urban Jordanian Arabic. Allard Jongman, Sonja Combest, Wendy Herd, and Mohammad Al-Masri (Linguist. Dept., Univ. of Kansas, 1541 Lilac Ln., Lawrence, KS 66044, jongman@ku.edu)

Previous acoustic analyses of minimal pairs of emphatic versus plain CVC stimuli showed that (1) emphatic consonants have a lower spectral mean than their plain counterparts and (2) vowels surrounding emphatic consonants are characterized by a higher $F1$, lower $F2$, and higher $F3$ than

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vowels surrounding plain consonants [Jongman *et al.*, J. Acoust. Soc. Am. **121**, 3169 (2007)]. The present perception study explores whether Arabic listeners' recognition of emphasis is based on information in the consonant or vowel. Monosyllabic words were used with emphatic and plain consonants in either initial or final position. By means of cross-splicing, the emphatic consonant or its adjacent vowel replaced the plain consonant or its adjacent vowel. Thirty Jordanian listeners participated in the experiment. On each trial, they indicated which word (emphatic or plain) they heard. Results show that the contribution of consonantal and vocalic information to the perception of emphasis depends on vowel quality: In the context of |a|, listeners seem to make their decision primarily based on the vowel while in the context of |i| and |u|, properties of the consonant carry more weight in this decision. The perceptual data will be compared to the acoustic measurements. [Research supported by the NSF.]

2pSC21. The weighting of vowel quality in perception of English lexical stress. Yanhong Zhang (411 Windsor Court, Ewing, NJ 0868, zhang66@purdue.edu) and Alexander Francis (Purdue Univ., West Lafayette, IN 470907-2038)

Acoustically, English lexical stress is multidimensional, involving *F*0, duration, intensity, and vowel quality. Previous research found that Mandarin speakers had problems using vowel reduction in English lexical stress production. Assuming nativelike perception is a prerequisite to nativelike production for non-native speech, the weight of vowel quality with comparison to that of *F*0, duration, and intensity in Mandarin listeners' stress perception was examined. Mandarin and English listeners judged lexical stress placement in synthesized tokens of desert, in which the first syllable /de/ was varied along vowel quality and each of the other cues depending on the pair of cues in focus. Results showed that both Mandarin and English listeners consistently weighted vowel quality more than the other cues. Vowel quality and duration were treated as combinational cues by both groups. English listeners used both intensity and vowel quality (separately), while Mandarin listeners did not use intensity at all. Findings suggest that Mandarin listeners had a nativelike use of vowel quality for perceiving English stress. However, Mandarin listeners treated *F*0 in a different way from English listeners, possibly owing to the influence of their native tonal background. Implications for the interaction between production and perception in second-language learning will be also discussed. [Work supported by Purdue Linguistics.]

2pSC22. Duration of tone sandhi in Mandarin Chinese. Bei Yang (FLARE, the Univ. of Iowa, PH658, Univ. of Iowa, Iowa City, IA 52240, bei-yang@uiowa.edu)

Tone sandhi is the tonal alternation when they are connected in speech flow. Six types of tone sandhi of disyllabic words in Mandarin Chinese are investigated in this research, including the neutral tone, the dipping tone alternation, /bu/, /yi1/, /yi2/, and double change (both of tone changes in a disyllabic word). The paper explores the duration of these six types and whether the neutral tone can change into the citation tones (level, rising, dipping, and falling tones) if we change the duration of the neutral tone. Five Chinese native speakers participate in the study. Two tasks are used to elicit data. First, speakers are required to read 40 disyllabic words. The second one is an identification task. The duration of eight neutral tones are elongated by acoustic technology. The participants hear the eight processed neutral tones and eight unprocessed neutral tones, and are asked to judge the tone types. The results indicate that the neutral tone has the shortest duration, and the duration of other sandhi tone is shorter than that of the normal citation tones. The data also show that the neutral tone and citation tones can be altered based on duration and pitch.

2pSC23. The development of tonal duration in Mandarin-speaking children. Jie Yang (Dept. of Commun. Sci. and Disord., Univ. of Texas at Austin, 1 Univ. Station A1100, Austin, TX 78712, thyjessie@mail.utexas.edu), Randy Diehl and Barbara Davis (Univ. of Texas at Austin, Austin, TX 78712)

Previous research found that the duration of segments decreases as children grow older. The development of suprasegmental duration, however, has not been explored. The present study investigated developmental changes in duration of the four Mandarin tones. 5-, 8-, and 12-year-old monolingual Mandarin-speaking children and young adults participated in the study. Tone durations were measured in participants' production of monosyllabic target words elicited by picture identification tasks. The results were as follows (1) For each tone category, tone duration and variability decreased with age: 5- and 8-year-old children showed significantly longer durations than adults. Tone durations in 12-year-old children approximated adult values. (2) Despite longer durations, adultlike duration patterns across tone categories existed in all children: dipping tones were the longest, followed by rising and level tones, with falling tones being the shortest. (3) Duration differences between the rising and dipping tones became larger as children grew older. The results may be indicative of the general maturation of laryngeal control over age. Although 5- and 8-year-old children have already established lexical contrasts of tone, adultlike phonetic norms are still in the process of development. The developmental data also provide support for a hybrid account of speech production from a suprasegmental perspective.

2pSC24. Effects of language background on tonal perception for young school-aged children. Chang Liu and Jie Yang (Dept. of Commun. Sci. and Disord., Univ. of Texas at Austin, Austin, TX 78712)

Given that Chinese is a tonal language while English is not, the present study investigates how language background affects tonal perception for young children. Tonal identification and discrimination are measured on for three groups of school-aged children (6–7 years old): Chinese-monolingual (CM), English-monolingual (EM), and English-Chinese bilingual (ECB) children. Children's task is to identify and discriminate tone 1 (level), tone 2 (rising), and tone 4 (falling) of a Chinese syllable /ma/, for which the fundamental frequency contour is systematically manipulated from tone 1 to tone 2 and from tone 1 to tone 4. CM and ECB children show typical categorical perception in both identification and discrimination, while EM children cannot identify and discriminate the three tones. These results suggest that learning and exposure experience of the tonal language is critical for children to perceive tonal changes in speech sounds.

2pSC25. Hemispheric processing of pitch accent in Japanese by native and non-native listeners. Jung-Yueh Tu, Xianghua Wu, and Yue Wang (Dept. of Linguist., Simon Fraser Univ., 8888 Univ. Dr., Burnaby, BC V5A 1S6, Canada, jta31@sfu.ca)

It is well established that language processing is left hemisphere dominant. Previous findings, however, indicate that lateralization of different levels of linguistic prosody varies with their functional load as well as listeners' linguistic experience. This study explored the hemispheric processing of Japanese pitch accent by native and non-native listeners differing in experience with pitch, including 16 native Japanese participants, 16 Mandarin Chinese participants whose native language has linguistic tonal contrasts, and 16 English participants with no tone or pitch accent background. Pitch accent pairs were dichotically presented and the listeners were asked to identify which pitch accent pattern they heard in each ear. Preliminary results showed that for all the three groups, the percentage of errors for the left ear and that for the right ear were comparable, indicating no hemispheric dominance. The Japanese group did not reveal left hemisphere dominance, as previously found for linguistic tone processing by native listeners. The performance of Mandarin group infers that tone language background did not significantly affect the lateralization of pitch accent. These findings are discussed in terms of how linguistic function differentially influences the hemispheric specialization of different domains of prosodic processing by native and non-native listeners. [Work supported by the NSERC.]

2pSC26. The effect of weak tone on the *f*0 peak alignment. Seung-Eun Chang (Dept. of Linguist., Univ. of Texas at Austin, 1 University Station B5100, Austin, TX 78712, sechang71@gmail.com)

The *f*0 peak sometimes occurs after the syllable with which it is associated, and the peak alignment varies, depending on several factors such as lexical tone target, neighboring tones, focus, and so forth. This study investigates the effect of weak tones on the alignment of *f*0 peaks with three tone types (i.e., H, M, and R) of South Kyungsang Korean, spoken in the southeastern part of Korea. When three tone types are followed by one or two unstressed suffixes, R was found to have the maximum amount of peak delay and M was found to have the minimum amount, i.e., the peak came in the second syllable, following the R-toned syllable, but the peak came in the syllable following the H-toned syllable. This peak delay was not found for M. Thus, it is argued that the tone alternation patterns in suffixed words are not random; rather, they systematically reflect the phonetic implementation of each tonal target. For example, the peak is in the final portion of a syllable in R, and it takes more time for the peak to be fully realized. This effect can be clearly observed when the following tone is weak.

2pSC27. Investigating the influence of context frequency on lexical tone perception. Jingyuan Huang and Lori Holt (Dept. of Psych., Carnegie Mellon Univ., 5000 Forbes Ave., Pittsburgh, PA 15232, jingyuan@andrew.cmu.edu)

Tone languages such as Mandarin use pitch variations to contrast meaning. Within tone languages, large variability exists in the pitch of tones produced by different speakers. However, previous studies of speaker normalization for contour tones have produced inconsistent results; whether speakers rely on context information in tone perception is unclear. The present study intended to provide an unambiguous test of the effect of context on contour lexical tone perception and to explore its underlying mechanisms and sources of information. In four experiments, Mandarin listeners' perceptions of Mandarin first and second (level and rising) tones were investigated with preceding speech and nonspeech contexts. Results indicate that (1) the mean fundamental frequency ($f0$) of a preceding sentence affects the perception of contour lexical tones and the effect is spectrally contrastive: Following a sentence with a higher-frequency mean $f0$, a following word is more likely to be perceived as a low-frequency tone and vice versa; (2) nonspeech precursors also elicit this effect, suggesting general perceptual rather than articulatory-based mechanisms; (3) listeners can use information from both fundamental frequency and periodicity to normalize tone perception. [Work supported by NIH NIDCD 2 R01DC004674-04A2].

TUESDAY AFTERNOON, 11 NOVEMBER 2008 — LEGENDS 12, 1:55 TO 4:30 P.M.

Session 2pSP

Signal Processing in Acoustics and Underwater Acoustics: Signal Processing for High Clutter Environments

Ronald A. Wagstaff, Cochair
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Chair's Introduction—1:55

Invited Papers

2:00

2pSP1. Signal processor for detection of signals in cluttered environments. Ronald Wagstaff and Heath Rice (NCPA, Univ. of Mississippi, 1 Coliseum Dr., University, MS 38677, rwagstaf@olemiss.edu)

The well-publicized experience of the USS Cole in a foreign port demonstrates the potential danger our military ships are frequently exposed to. When docked, they are the most vulnerable to attack by small fast boats that can hit and run before their threat is recognized, and defenses can be activated. This is a challenge that has many facets. Attack by a small high-powered fast-boat is just one type of threat. However, it is important. One way of quickly identifying a fast-boat is with an underwater acoustic sensing system. Such a system is not without challenges. Harbors are typically busy and contaminated by many forms of acoustic clutter. Detecting and separating the signals of fast-boats from among the clutter are difficult tasks. Fortunately, a signal processor has been developed with highly coherent fast-boat signals, and harbor clutter in mind. This fast-boat processor invokes temporal coherence constraints, by degree, to strip away incoherent noise and less coherent shipping signals, and leaves the fast-boat signals exposed. This signal processor will be discussed, and the results will be presented to illustrate the substantial signal-to-noise ratio and automatic signal detection gains that can be achieved.

2:20

2pSP2. Human detection algorithm for seismic and ultrasonic detectors. Alexander E. Ekimov and James M. Sabatier (NCPA, Univ. of Mississippi, 1 Coliseum Dr., University, MS 38677)

Range detection methodologies for human acoustic signals are discussed. Seismic, passive, and active Doppler ultrasonic sensors are used in the presented methods. The algorithm developed for recognition of human's presence in the measured signals is based on detection of the specific acoustic signatures resulting from specific human motion characteristics. These signatures have two characteristic times, $T1$ (the footstep repetition time, which is equal to the time of the whole body periodic motion) and $T2$ (the footstep duration time). The footstep duration time is equal to the time interval for a single footstep from "heel strike" to "toe slap and weight transfer." Taking advantage of these times in signal processing for optimization of the signal-to-noise ratio and applying a procedure of cadence frequency detection allow recognition of human presence in the analyzed signals. This algorithm was tested and it was experimentally demonstrated that cadence frequencies for walking human and their harmonics for seismic and ultrasonic signatures (passive and active Doppler signatures) were the same. Cadence frequencies were stable and had no detectable variation in time for a regular walking style. This stability feature potentially may be used for human tracking. [Work supported by the Department of the Army, Army Research Office, Contract No. W911NF-04-1-0190.]

2:40

2pSP3. Hardware and software solutions to noise in resonance measurements. Joseph Gladden, III (Dept. of Phys., Univ. of Mississippi, University, MS 38677, jgladden@phy.olemiss.edu)

Resonance measurements have the benefit that the system itself is a natural signal amplifier with a gain proportional to the quality factor (Q) of the resonator. However, even this small respite from Murphy's law is not always enough to coax weak resonance signals out of a noisy background. In this talk I will discuss several aspects of improving signal to noise ratios in resonant ultrasound spectroscopy measurements using both hardware and software based solutions such as custom low-noise preamplifiers, lock-in amplifiers, dynamic averaging during data acquisition, and nonlinear fitting of noisy and overlapping (low-Q) peaks to an analytical model. Some particular issues of signal transmission across high-thermal gradients will also be discussed.

3:00—3:15 Break

3:15

2pSP4. Identifying individual clicking whales acoustically amid the clutter of other clicking whales. George E. Ioup, Juliette W. Ioup, Lisa A. Pflug, Arslan M. Tashmukhambetov (Dept. of Phys., Univ. of New Orleans, New Orleans, LA 70148, geioup@uno.edu), Natalia A. Sidorovskaia (Univ. of Louisiana at Lafayette, Lafayette, LA), Charles H. Thompson (NOAA/NMFS/SEFSC, Stennis Space Ctr., MS), and Christopher O. Tiemann (Univ. of Texas at Austin, Austin, TX)

Exceptionally clear recordings of sperm whale (*Physeter macrocephalus*) codas reveal time and frequency properties, which show that clicks within a coda are remarkably similar to each other but that they can differ from clicks in other codas. This is consistent with the hypothesis that individual whales can be identified by the characteristics within their codas. Research has centered on the cluster analysis of these codas to help establish whether acoustic identification of individuals is possible. Recently the cluster analysis has been made more robust. This increases the confidence in the applicability of the approach. Data are now available to couple visual sightings and acoustic tracking with recordings for acoustic identification to give an independent verification of the analysis. Acoustic identification of individuals has also been attempted using isolated echolocation clicks of sperm and beaked whales (*Mesoplodon densirostris* and *Ziphius cavirostris*). Although this is a more difficult and problematic undertaking, there has been some promising cluster analysis. Again data for acoustic tracking and visual observations are now available with the recordings for acoustic identification using echolocation clicks to test and perhaps validate the method. [Research supported in part by SPAWAR.]

3:35

2pSP5. Clutter prediction using artificial neural networks. Juliette Ioup (Naval Res. Lab. at Stennis Space Ctr., MS 39576 and Dept. of Phys., Univ. of New Orleans, New Orleans, LA 70148, jioup@uno.edu), Maura Lohrenz, and Marlin Gendron (Naval Res. Lab., Stennis Space Ctr., MS 39576)

Clutter is a known problem in electronic geospatial (map) displays, on which many different types of data can be combined and presented as a single image. In this context, clutter may be thought of as an overabundance of information, which reduces display usability by the viewer. To declutter a geospatial display, it is necessary to first predict the amount of clutter a human observer might perceive. Computer-aided classification of maps according to the amount of clutter likely to be perceived by a human viewer is the goal of the research described here. Artificial neural networks are among the possible choices of computing techniques that can be used for this task. The network is trained using prior clutter classifications of training maps made by humans, so that automated prediction of clutter for a new map is possible. The objective is to have the network classify maps according to clutter as perceived by human judges. Several neural network algorithms and trials with data consisting of cluttered map classifications and response times by human judges are described. The choice of input features, including the use of principal components, is discussed. Preliminary results with test maps show good prediction of clutter.

3:55

2pSP6. Mitigation of speckle noise due to laser Doppler vibrometer motion across a vibrating target. Richard Burgett (Planning Systems, Inc., Univ. of Mississippi, 1 Coliseum Dr., University, MS 38677, burgett@psislidell.com), Vyacheslav Aranchuk, and James Sabatier (Univ. of Mississippi, University, MS 38677)

Laser Doppler vibrometers (LDVs) are widely used for remote vibration measurements. In many applications, such as measurements with an LDV on a moving platform or measurements with a continuously scanning beam, an LDV beam moves across a target. The motion of a laser beam across the target generates noise at the LDV output due to dynamic speckles—speckle noise. Speckle noise is caused by phase fluctuations of laser speckles and by Doppler signal dropouts and increases with the speed of the beam. It has been observed that the noise floor increase due to speckles follows a logarithmic curve as the speed increases. Of course as the speed increases, the dwell time over the target area decreases, which limits the signal of interest. This smaller sampling of the vibration signal, combined with the increase in noise due to speckles, places a severe limit on the scanning speed. A way is being investigated to take advantage of this nonlinear increase in noise with respect to speed by using time multiplexing of multiple LDV beams to increase the length of time the LDV samples the vibrations of the target area, thereby allowing the target signal to be detected above the noise floor.

Contributed Paper

4:15

2pSP7. Acoustic Dopplergram for intruder defense. T. C. Yang (Naval Res. Lab., 4555 Overlook Ave., Washington, DC 20375)

This paper discusses the concept and presents preliminary experimental results using the Dopplergram to detect and localize an underwater vehicle, intended for harbor defense and/or protection of high value assets. The acoustic Dopplergram displays the Doppler frequency of the target echo from an active source as a function of time similar to the Lofargram (or spectral gram), which is widely used in passive sonar for detection of tonal and/or wideband transient signals. *m*-sequence signals, which are sensitive to Doppler shift, are transmitted with a rapid repetition rate from a source and received on a colocated receiver. Target detection is improved using the processing gain of the *m*-sequence and using Doppler discrimination (signal association) by eye-ball integration of the (Doppler) gram data. The target is localized using a two-way travel time and bearing estimation. [This work is supported by the US Office of Naval Research.]

TUESDAY AFTERNOON, 11 NOVEMBER 2008 LEGENDS 8, 1:30 TO 4:00 P.M.

Session 2pUW

Underwater Acoustics: Inversion and Uncertainty

Steven Finette, Chair
Naval Research Laboratory, Washington, DC 20375-5320

Contributed Papers

1:30

2pUW1. A sparse-grid, nonintrusive formulation of acoustic field uncertainty in ocean waveguides. Steven Finette (Naval Res. Lab., Washington, DC 20375-5320) and John Burkhardt (U.S. Naval Acad., Annapolis, MD 21402)

The inclusion of environmental uncertainty in simulations of acoustic wave propagation in ocean waveguides is important for the development of simulation-based prediction methods that quantify the influence of multiple sources of incomplete environmental knowledge on the simulation results. Polynomial chaos expansions have been suggested as a natural mathematical framework for describing both environmental and acoustic field uncertainties, their interaction, and propagation through the waveguide [S. Finette, J. Acoust. Soc. Am. **120** (2006)]. Previous research has described the inclusion of these expansions directly into the propagation equation (the intrusive approach), yielding coupled differential equations for the expansion coefficients. The solution for the coefficients contains the statistical properties of the uncertain field. Here we describe an alternative nonintrusive formulation, where existing acoustic propagation codes can be used to estimate the chaos coefficients rather than solve for them via a complex set of coupled differential equations. The nonintrusive formulation involves multiple solutions of an existing deterministic code, e.g., a wide-angle parabolic equation solver, in conjunction with the Smolyak sparse-grid interpolation and multidimensional quadrature to obtain uncertainty statistics on the acoustic field. [Research supported by the Office of Naval Research.]

1:45

2pUW2. A chaos-based wide-angle parabolic equation model for sound propagation in random ocean. Li Ma and Hao Xing (Inst. of Acoust., Chinese Acad. of Sci., Beijing 100190, China, mary1968@tom.com)

Recently, the chaos-based methods have attracted much attention in ocean acoustics committee. In this paper, a set of partial differential equations (PDEs), which are based on the wide-angle parabolic equation and Wiener-chaos decomposition, was established to investigate the propagation of sound propagation in the ocean environment with an arbitrary sound speed profile. Simultaneously, a numerical treatment of the PDEs, which employed an implicit difference scheme and a spectral method, was developed. By using this method, three cases with different kinds of randomness under a distinct ocean environment are studied numerically.

2:00

2pUW3. Uncertainty and resolution in continuum inversion of ocean bottom geoacoustic properties. Andrew A. Ganse and Robert I. Odom (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, aganse@apl.washington.edu)

Inversion of ocean bottom geoacoustic properties from acoustic receptions in the water column is a nonlinear inverse problem that is inherently unstable and nonunique. One common approach to stabilizing this problem is to assume that the ocean bottom is made up of a small number of layers. The solution from this approach does allow one to reproduce the scattered sound field if all the other experiment parameters such as frequency and geometry are also reproduced. However without extensive prior information about that ocean bottom, this approach yields only one of many equivalent nonunique solutions and may not accurately describe the actual ocean bottom itself. An alternate approach, which may allow one to reuse the results later with a different frequency or geometry, is to use the tools of geophysical continuum inversion to specify the degree of nonuniqueness by quantifying both the uncertainty and limited resolution of the continuum bottom solution. This work compares inversion uncertainty and resolution results for different formulations of the data (e.g., matched field versus matched modes versus waveform), different geometries, and different formulations of the uncertainty (e.g., normally distributed versus including some higher-order moments). [Work partially supported by ONR.]

2:15

2pUW4. Resolution matrix perturbation series applied to a nonlinear ocean acoustic inverse problem. Robert I. Odom and Andrew A. Ganse (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St, Seattle, WA 98105, odom@apl.washington.edu)

The resolution operator for a nonlinear inverse problem is the product of the estimated inverse of the forward model operator and the forward model operator itself. If the inverse of the forward model operator were exact, then the resolution operator would be the identity, but in general the resolution operator describes the (noninvertible) transfer function between the unknown true environmental model and the limited-resolution version, which can be estimated from measurements. The nonlinear model resolution operator can be computed iteratively from the Neumann series representation of an estimate of the inverse of the forward model operator with the assumption that both the data functional and the model perturbation functional possess regular perturbation expansions. An example of a problem that fits these

criteria is normal mode acoustic propagation with "slow enough" perturbations such that the modes adjust adiabatically to the perturbations, and the mode eigenvalues are "far" from cutoff. We examine the effects of the nonlinear components of the model and the higher order components of the model resolution on the reconstruction of the model estimate for a simple ocean acoustic propagation problem. [Work supported by ONR.]

2:30—2:45 Break

2:45

2pUW5. Open ocean seamount geoacoustic inversion. Kevin D. Heaney (OASIS Inc., Fairfax Station, VA 22039)

During the BASSEX-2004 (Basin Acoustic Seamount Scattering Experiment), sound from three sources [(250 Hz LFM, 100 Hz bandwidth) and (75 Hz, m-seq, 50 Hz bandwidth)] was transmitted in a range of 200–250 km to a moving towed array receiver (200 m aperture, cut for 250 Hz) in the vicinity of two seamounts in the central North Pacific. The seamounts rise from a depth of 6000 m to approximately 1000 m below the sea surface. A simple sediment model (soft surficial unconsolidated sediment overlying a hard basement) is used to try and model the energetics of the receptions. Two sets of receptions are used to perform this geoacoustic inversion. The first is a received energy versus range (equivalent to TL versus range) in the shadow behind the seamount. The depth of the received level minimum behind the seamount is sensitive to the sediment cover of the seamount. This approach is limited by the available signal-to-noise ratio at the receiver. The second approach is to use data taken when the receiver is just above the seamount. For this geometry, the beamformer is capable of resolving direct propagating paths and paths that bounce off the seamount up to the array from below.

3:00

2pUW6. Geoacoustic inversion over a basalt seafloor. Kevin D. Heaney and Michael Vera (Univ. of Southern Missisippi, 118 College Dr., Hattiesburg, MS 39406)

Geoacoustic inversion from data just off a volcanic island (Kauai) poses significant challenges. One of the significant challenges is to separate the effects of seafloor roughness and shear wave conversion in the basement. Lava flow fields are known to have significant roughness on several scales relevant to low-frequency acoustic propagation (75 Hz). Basalt is known to have a very high-compressional speed of 3000 m/s and a variable shear speed ranging from 500 to 2000 m/s. The combination of shear and roughness can lead to significant ambiguity while performing a geoacoustic inversion because both mechanisms can lead to attenuation of the coherent field. During the BASSEX 2004 experiment up to 20 transmissions from the fixed NPAL source (75 Hz, m-seq with a 50-Hz bandwidth) were received on a towed horizontal line array at many orientations and ranges from the source. In this paper we look at several short-range transmissions and attempt to sort out the scattering effects of roughness with the attenuation effects of shear wave conversion in the sediment. The two mechanisms will be sorted out using simultaneous narrowband (transmission loss) and broadband inversions.

3:15

2pUW7. Inverting for the properties of an elastic seafloor using complex-density equivalent fluids. Michael Vera (Univ. of Southern Mississippi, 118 College Dr., #5046, Hattiesburg, MS 39406) and Kevin Heaney (OASIS Inc., Fairfax Station, VA 22039)

Receptions were recorded from a bottom-mounted broadband source (with a center frequency of 75 Hz) located near Kauai as part of the basin acoustic seamount scattering experiment (BASSEX). Travel times for arrivals from this source have been modeled at basin-scale ranges using complex-density equivalent fluids for the elastic seafloor material. The collection of acoustic data at shorter ranges as part of BASSEX allows for a more detailed examination of the accuracy attainable by an equivalent-fluid representation of the seafloor. The use of equivalent fluids is intended to depict the conversion of acoustic energy into shear waves. The performance and stability of propagation models can be substantially improved if equivalent fluids can be used to accurately characterize bottom interaction. Simulations have been performed in order to search for the equivalent fluid that best explains received data at a range of a few kilometers. The reflection coefficient of this equivalent fluid can then be matched to parameters of an elastic solid (density, sound speed, and shear speed) for the relevant grazing angles. The ability of these models to reproduce features of the data will serve to indicate the importance of elastic effects in the experimental results. [Work supported by ONR.]

3:30

2pUW8. Time domain integration to improve geoacoustic parameter estimation. Donald DelBalzo (Planning Systems, 40201 Hwy 190 East, Slidell, LA 70461, ddelbalzo@plansys.com), James Leclere, and George Ioup (Univ. of New Orleans, New Orleans, LA 70148)

Shallow-water acoustic predictions are affected by uncertainty in sediment property characteristics. Inverse methods using controlled active source signals have been developed to estimate seabed properties. However, some applications prefer a more covert approach. Our work is focused exclusively on passive inversion techniques using signals from surface ships of opportunity. This study addresses the accuracy of low-frequency (100 Hz) matched-field correlations and subsequent geoacoustic estimates using signals from moving surface ships with unknown source levels at unknown ranges. A time-staggered technique is employed to reduce ambiguities in thick-sediment descriptions, with ever increasing confidence as time evolves. Matched-field techniques are applied in a simulated shallow-water environment with a single vertical line array and high signal-to-noise ratios. The simulations indicate significant potential for accurate estimates of thick-sediment characterizations of grain size and layer thickness out to ranges of tens of water depths in shallow water, despite moderate mismatch conditions in the environmental model. [Work sponsored by SPAWAR.]

3:45

2pUW9. Attenuation inversions using broadband acoustic sources. Gopu R. Potty (Dept. of Ocean Eng., Univ. of Rhode Island, Narragansett, RI 02882), Preston Wilson (The Univ. of Texas at Austin, Austin, TX 78712-0292), James F. Lynch, Arthur Newhall (Woods Hole Oceanograph. Inst., Woods Hole, MA 02543), and James H. Miller (Univ. of Rhode Island, Narragansett, RI 02882)

Low-frequency attenuation data are widely scattered. The spread most likely reflects differences in sediment type, degree of consolidation, layering, as well as other causes of effective attenuation encountered during low-frequency field measurements. Moreover, attenuation varies with depth, and longer wavelengths, i.e., low frequencies, encounter deeper sediments with different properties. The assumption that the attenuation in the sediment is constant with depth often results in interpretations which give sediment attenuations with a frequency dependence different when depth variations are included in the sediment description. Low-frequency attenuation inversions using broadband acoustic data from different field experiments will be presented and compared. Compressional wave attenuation will be estimated using modal amplitude ratios. Variation of modal attenuation coefficients with frequency and depth will be discussed. [Work supported by the Office of Naval Research.]