which it is nonphonemic in their native language (p's < 0.001). This result was equally robust even for stimuli in which the contrastive portion of the stimulus (*zo-uzo*, *do-ro*) was spliced from the nonphonemic context and pasted into the phonemic context. These data indicate that phonological context plays an integral role in the perception of phonetic signals.

1aSC22. Articulatory gestures influence the perception of speech. Henny Yeung (Dept. of Psych., Univ. of British Columbia, 2136 West Mall, Vancouver, BC V6T 1Z4, Canada, hhyeung@psych.ubc.ca), Mark Scott, Bryan Gick, and Janet Werker (Univ. of Birtish Columbia, Vancouver, BC V6T 1Z4, Canada)

A central claim of the motor theory of speech perception [Liberman *et al.* (1967)] is that speech perception involves motor representations. We report evidence that articulatory movements influence perception. Subjects made forced-choice identifications of naturally recorded /aba/ and /ava/ tokens, while silently making articulatory gestures in time with presentation of the auditory tokens. These articulatory gestures either agreed with the auditory token (e.g., articulating "aba" while hearing /aba/) or disagreed (e.g., articulating "ava" while hearing /aba/). Subjects more frequently misidentified auditory /aba/ as /ava/ when articulated gestures disagreed, as compared to a base line condition (i.e., simply listening). Two further conditions suggest that simple priming of /ava/-percepts is unlikely. First, subjects articulated "afa" instead of ava while hearing /aba/. If error rates are specifically related to articulatory gestures, then the similarity in gestural movements between /f/ and /v/ should result in similar error rates. A priming account would not make such a prediction. In line with the motor explanation, subjects did have an equivalent error rate in afa and ava conditions. Second, minimal interference was observed when subjects only imagined themselves saying /ava/. These results support the notion that activation of motor movements can influence the perception of speech.

1aSC23. Investigating the consonant-vowel boundary. II. Perceptual contributions of glimpse windows. Daniel Fogerty and Diane Kewley-Port (Dept. of Speech and Hearing Sci., Indiana Univ., 200 S. Jordan Ave., Bloomington, IN 47405, dfogerty@indiana.edu)

Discrete segmental units do not occur in fluent speech because of coarticulation. However, separation of the meaningful sounds in language into the categories of consonants and vowels is one of the most fundamental principles of how sound is structured. Our previous research has shown that for sentence presentations, vowels have a distinct perceptual advantage over consonants in determining sentence intelligibility. TIMIT sentences were used to investigate perceptual contributions of consonants and vowels across the consonant-vowel (CV) boundary, by shifting the CV boundary by specific proportions of the vowel, such that consonant duration increased while vowel duration decreased. Glimpse windows are defined as the speech signal preserved between noise replacements. The perceptual effect of windows either locked to specific segmental information or placed randomly was examined. Results from glimpse windows locked to segmental information confirmed a 2-to-1 vowel advantage for intelligibility at the traditional CV boundary and suggest that vowel contributions remain robust against deletions of the signal. When glimpses were presented randomly, summed duration of glimpses predicted performance. However, performance remained lower than when glimpse windows of equivalent duration were locked to vowels. Specific segmental information appears to differentiate performance between consonant and vowel conditions. [Work supported by the NIH.]

MONDAY AFTERNOON, 10 NOVEMBER 2008 LEGENDS 9, 1:00 TO 3:10 P.M.

Session 1pAAa

Architectural Acoustics: Acoustics of Single Family Residences

Richard D. Godfrey, Cochair
448 N. Pearl St., Granville, OH 43023

Nancy S. Timmerman, Cochair
25 Upton St., Boston, MA 02118-1609

Chair's Introduction—1:00

Invited Papers

1:05

1pAAa1. New home buyer noise reduction listening study. Harry A. Alter (19 Beechtree Ln., Granville, OH 43023)

In 2007 Owens Corning Science Technology in Granville, Ohio built a comparative wall assembly listening facility called the Acoustic Research Experience Laboratory (AREL) Annex. Within this listening facility, approximately 100 perspective new home buyers were selected to listen and evaluate various sound clips when played through wall assemblies at varying levels of noise reduction. Jurors responded to perceptual and lifestyle questioning. The building of the AREL Annex and listening study results will be discussed.

1:25

1pAAa2. Effects of residential audible distractions on the performance and perception of home office workers. Alicia J. Wagner and Lily M. Wang (Architectural Engr. Prog., Peter Kiewit Inst., Univ. of Nebraska-Lincoln, 1110 S. 67th St., Omaha, NE 68182-0681, aliciajean01@gmail.com)

This research examines the effects that typical residential audible distractions have on task performance and subjective perception of home office workers when transmitted across various residential wall constructions. Previous studies have investigated how speech, music, and office equipment noises may deteriorate one's performance, but few have presented these distractions in such a way as to imitate the home office environment or used audible distractions that are characteristic of a residential setting. In this study, subjects performed math, verbal, and typing tasks over 1 h while exposed to four types of audible distractions: (1) pop music, (2) television, (3)

conversation, and (4) a potpourri of kitchen, pet, and children's noises. A short questionnaire was also administered to determine subjective perception under the different noise signals. Statistical analyses of the results indicate that none of the audible distractions caused a significantly different effect on task performance. The loudest distraction (music), however, was subjectively rated to be most distracting, while the signal with the most variation in time (potpourri) was considered to be the most annoying.

1:45

1pAAa3. Auralization as a tool for acoustical design of single family residences. Carl Rosenberg, Jonah Sacks (Acentech Inc., 33 Moulton St., Cambridge, MA 02138, crosenberg@acentech.com), Erin Dugan, and David McDonald (USG Corp., Libertyville, IL 60048)

Acoustical requirements play an increasing role in residential building design. In an effort to understand these changes USG Corp., a manufacturer of building materials and developer of acoustical solutions for residential and commercial markets, engaged Acentech Inc. to develop aural demonstrations, or auralizations, of common acoustical issues in single family residences. Using computer modeling and audio processing techniques, a number of convincingly realistic simulations of familiar residential scenarios were developed. These auralizations allowed USG to better understand acoustical issues and recommend appropriate products and systems. This paper shares the experience of this project.

2:05

1pAAa4. Incorporating acoustically challenging functions into single family residences. Russ Berger (RBDG, 4006 Belt Line Rd., Ste. 160, Addison, TX 75001, russ@rbdg.com)

Recording and broadcast studios, screening theaters, music performance venues, and other acoustically challenging functions have been successfully incorporated into single family residences. Several examples will be discussed illuminating concerns beyond normal sound isolation and acoustical performance requirements.

Contributed Papers

2:25

1pAAa5. Sound in a single family house. Sergio Beristain (Lab Acoust., ESIME, IPN. IMA President, P.O. Box 121022, 03001 Mexico, D.F., Mexico. sberista@hotmail.com)

Apart from road and air traffic noise, and some noise from neighbors, the most important noise in single family residences usually is the noise generated within the residence itself, such as music, general activities, operating machinery for the house and people comfort, and the sound isolation characteristics of the different partitions within the house. Mexican single family residences are usually made out of brickwork, but some inner walls are nowadays changing from brickwork to rocksheet and similar materials, which reduces noise attenuation from kitchens, washing rooms, and living spaces. Home theaters are usually installed at will, and the resulting sound has limitations. A case is presented where family relations started to deteriorate due to the low isolation capabilities of the inner walls.

2:40

1pAAa6. A measurement survey of the acoustic conditions in home offices. Megan J. Christensen and Lily M. Wang (Architectural Engr. Prog., Peter Kiewit Inst., Univ. of Nebraska-Lincoln, 1110 S. 67th St., Omaha, NE 68182-0681, lwang4@unl.edu)

As home offices are becoming more prevalent in our society, it is critical to ensure that the acoustic conditions in those environments are suitable for optimal home office worker performance. In this paper, a measurement survey is presented of existing conditions in several home offices within the metropolitan Omaha, Nebraska area. The data that were gathered include background noise levels (both with and without ventilation noise), air-borne sound attenuation of the office envelope, and logged data of the office sound levels during a typical 48-h period over the work week. Summaries of the findings will be discussed and are useful in understanding how residential design and construction should be modified to improve the acoustic conditions of home offices.

2:55

1pAAa7. Acoustical considerations for luxury motor homes. Jennifer Shaw and Charles Moritz (Blachford Acoust. Lab., 1445 Powis Rd., West Chicago, IL 60185, jshaw@blachfordinc.com)

Although not considered a typical single-family home, a large number of luxury motor home owners spend at least 6 months per year living in their coach. A survey of class A motor home owners confirmed that for these owners, the coach becomes a home-away-from-home and requires many of the same amenities of a real home, including low interior noise levels. The survey noted that when parked at a campsite or motor home resort, residential noise issues were a significant concern, particularly noise from a neighbor's generator. Acoustical issues found in motor homes, along with the results of the survey, will be compared with updated results from previously reported motor home testing [J. T. Kunio and C. T. Moritz, SAE 2003 Trans., **112**, 1800–1810 (2003)] to understand how these concerns may be addressed.

Session 1pAAb

Architectural Acoustics: Multifamily Structure—Advances and Legal Issues

Angelo J. Campanella, Chair
Campanella Associates, 3201 Ridgewood Dr., Columbus, OH 43026-2453

Chair's Introduction—3:20

Contributed Papers

3:25

1pAAb1. Who pays? Defining construction defects and assigning blame in borderline acoustical assemblies. John LoVerde and Wayland Dong (Veneklasen Assoc., 1711 16th St., Santa Monica, CA 90404, jloverde@veneklasen.com)

It is often the case when performing field noise isolation testing that some assemblies will meet the relevant statutory requirements and other nominally identical assemblies will fail. Sometimes this can be attributed to errors in construction materials and/or methods. However, when an assembly is "borderline," that is, only just meets the requirements based on historical or laboratory data, it can be expected that a certain percentage will fail even when constructed without error. This is due to the variation in construction procedures and materials as well as the inherent uncertainties in acoustical measurements. As a result, it is often not clear whether any particular assembly failed because of construction error or inadequate design and, by extension, whether the contractor or the designer bears the responsibility and resultant financial liability. The definition of construction defects in acoustical assemblies and the resultant division of responsibility are discussed and illustrated with actual cases.

3:40

1pAAb2. Performance issues of resilient channels in wall systems. Stephen W. Payne, Jr. (USG Corporate Innovation Ctr., 700 N. Hwy., 45 Libertyville, IL 60048)

There has been much recent discussions on the issue of proper installation of resilient channels. Of particular concern is the issue of the "shorting out" of the resilient channels, wherein the gypsum board attachment screws passes through the resilient channel and into the structural member. This paper will present the test results obtained during a series of tests to evaluate the significance of this issue. In addition data will be presented from a further study on the significance of the geometry of the resilient channel design.

3:55

1pAAb3. Characterization of airborne sound transmission and impact isolation in floor/ceiling assemblies with structural cementitious floor sheathing and cold formed steel framing. Timothy D. Tonyan (Systems Development, Structural Technologies Group, U.S. Gypsum Co., 21925 W. Field Pkwy., Ste. 245, Deer Park, IL 60010), Stephen W. Payne, Jr. (USG Corp. Libertyville, IL 60048), and Robert Elfering (Shiner + Assoc., Chicago, IL)

There is growing demand for floor/ceiling assemblies that combine noncombustible structural cementitous floor sheathing panels with cold-formed steel (CFS) framing. The combination being light in weight and possessing high stiffness makes these floor systems particularly advantageous in seismic zones, where reduction in building mass can translate into structural efficiencies and cost savings. However, as a result of their relatively low mass appropriate design strategies must be used to control airborne sound transmission and impact noise. This paper presents research conducted to characterize the acoustical behavior of floor/ceiling assemblies using structural cementitious floor sheathing combined with CSF framing. A variety of isolation and damping strategies are described and tested. The influence of the use of different underlayments and acoustical mats on STC and IIC values in the floor/ceiling assemblies is documented and evaluated. Isolation approaches, including the use of drywall suspension systems for ceiling attachment, are also presented. The results demonstrate that, by using the appropriate damping and isolation strategies, floor/ceiling assemblies combining structural cementitious floor sheathing with CFS framing can provide excellent acoustical performance. Recommendations for further research are also provided.

4:10

1pAAb4. Acoustic and vibratory characterization of in-room footfall noise: Part 2. Robert M. Tanen, Jonathan C. Silver, Michelle C. Vigeant, and Robert D. Celmer (Acoust. Program + Lab., Mech. Eng. Dept., Univ. of Hartford, 200 Bloomfield Ave., W. Hartford, CT 06117, celmer@hartford.edu)

Prior to Phase I of this study [J. Acoust. Soc. Am. **22**], existing footfall literature was primarily focused on transmission between spaces, i.e., IIC. This study measured the sound power and vibratory spectra produced by footfalls using both human subjects and standard tapping machines. One tapper had rubber-tipped drop weights and the other had cored samples of shoe soles used by human participants. Within a reverberation room, 12 floor profiles were tested with the tapping machines. Human subjects were tested with three previously tested floors and a new floor profile. Fourteen male and female subjects walked on the floor surfaces while wearing three different types of footwear: leather-soled shoes (hard), rubber-soled shoes (medium), and sneakers (soft). Sound power spectra and vibratory signatures for each condition were measured using the procedures of ISO 3741. The current study verified repeatability from Phase I to Phase II, produced sound power footfall data for vinyl flooring applied directly to concrete, and developed improved correction curves to model tapping machines as human footfalls. Similarities between the rubber-tipped tapper and women's leather shoes and a comparison of sound power and corresponding vibratory signatures are discussed. [Worked supported by The Paul S. Veneklasen Research Foundation.]

4:25

1pAAb5. Sound quality of laminate flooring systems. James Wilson, III and Kenneth Cunefare (Woodruff School of Mech. Eng., Georgia Inst. of Technol., Atlanta, GA 30332-0405)

The laminate flooring industry identifies the ability of a laminate floor to recreate the sound of traditional hardwood floors as an important metric for their perceived quality and acceptance. The market describes the inability of laminate flooring to reproduce the natural sound of wood as a drawback to their market appeal. Therefore, if a laminate flooring composite can be offered to the market, which recreates the natural wood floor acoustic experience, the floor will offer additional value to the consumer. Products on the market today exist, which do improve the sound quality of laminate flooring composites. However, there is no unified standard to justify or prove marketing claims made by manufacturers. Prior work in the area focused on the spectral content of the signal. In the area of psychoacoustics, many additional metrics exist to describe the human perception of sound, which aids in describing sound quality. These metrics can be utilized to characterize the

sound of traditional hardwood floors and laminate systems, beyond what has been done in previous work. By employing these metrics and then correlating them to subjective sound jury perception, it is possible to better predict the sound quality of a laminate floor.

4:40

1pAAb6. Conceptual design considerations for the acoustical design of an animal research facility. Byron Davis and Kevin Richardson (VACC, 490 Post St., Ste. 1427, San Francisco, CA 94102, byron@vaconsult.com)

Conceptual design considerations from the acoustical design of an animal research facility are presented in this paper. Animals in laboratory settings are experimentally stressed to observe outcomes. Stressors external to the experiment confound data and reduce research productivity. Among these experimental contaminants is acoustical noise, which has well-documented extra-auditory effects in animals. Both steady-state and transient sounds can affect development, sleep, reproduction, behavior, and other parameters important to animal models. Reasonably comprehensive sensitivity data exist in the medical and laboratory literature; however, there are few treatments of the acoustical design of vivarium and animal laboratory facilities. Acoustical design of these facilities is challenging due to the broad array of sensitivities (including ultrasonic frequencies), scarcity of high-frequency data for noise sources, contamination control and cleanliness requirements, and surface durability concerns. Acoustical design criteria for animal holding areas and experimental test rooms are considered. The criteria are based not only on sound pressure levels but also on frequency content, isolation of incompatible species, expected vocalizations, and minimization of disruption. Noise sources include mechanical/electrical/plumbing equipment, maintenance activities, and other animals. The design efforts include categorization of animal groups, noise isolation concepts, acoustical design, and noise control measures.

4:55

1pAAb7. Room sound absorption provided by underbench sound absorbing material. Pamela Harght and Robert Coffeen (School of Architecture and Urban Planning, Univ. of Kansas, 1465 Jayhawk Blvd., Lawrence, KS 66045)

The concept that sound absorbing material added beneath "hard" benches and pews will be useful in the reduction in reverberation in religious worship facilities, and other places of assembly are occasionally advanced by facility owners and users who do not desire to employ bench and pew seat upholstery or seat cushions. This paper presents data on underbench sound absorption calculated from measurements made in a reverberant space with and without the addition of a sound absorbing material to the underside of wood benches.

MONDAY AFTERNOON, 10 NOVEMBER 2008 — LEGENDS 8, 1:25 TO 4:30 P.M.

Session 1pAO

Acoustical Oceanography and Underwater Acoustics: Acoustics and Inversions on the Continental Shelf and Canyons

James F. Lynch, Cochair
Woods Hole Oceanographic Inst., Woods Hole, MA 02543-1541

Jennifer L. Wylie, Cochair
Rosenstiel School of Marine and Atmospheric Sci., 4600 Rickenbacker Causeway, Miami, FL 33149

Chair's Introduction—1:25

Contributed Papers

1:30

1pAO1. Observed intensity and horizontal field coherence variability of low-frequency pulse transmissions on the continental shelf. Timothy F. Duda, Jon M. Collis, Ying-Tsong Lin, and James F. Lynch (Woods Hole Oceanograph. Inst., AOPE Dept., MS 11, Woods Hole, MA 02543)

Rapid coastal environmental evolution leads to highly variable acoustic fields. To quantify such variability, one component of the Shallow-Water 2006 (SW06) program on the shelf east of New Jersey was time series measurement of sound transmitted from fixed sources to joined horizontal and vertical line arrays. Transmission paths were both cross-shelf and along-shelf (across and along dominant internal-wave crests). Data were collected for over one month. Intensity time series of 100–400-Hz pulses was found to have strong variability at periods from hours to over a day, consistent with long-wavelength internal-tide effects. Such effects can arise from adiabatic mode and/or coupled mode propagation. Separation of fluctuations into slow and rapid contributions allows calculation of a time-varying horizontal coherence-length statistic. For along-wave crest transmission, this was highly variable, typified by values ranging from a few acoustic wavelengths to over 40 wavelengths, typically 10–25. The slow coherence-length fluctuations had signatures of periodic (tidal) mode-refraction episodes (with short scale) during active intervals, caused by internal-wave ducting. Conditions were more steady at other times. Across-crest transmissions showed shorter than expected scale lengths of tens of wavelengths with more subtle tidal dependence. [Work supported by the Office of Naval Research.]

1:45

1pAO2. Acoustic hindcasts of array performance using a combination of submesoscale hydrodyamic and acoustic models. Steven Finette, Roger Oba, Thomas Hayward, Colin Shen (Naval Res. Lab., Washington DC 20375-5320), Patrick Gallacher, Alex Warn-Varnas, and Steve Piacsek (Naval Res. Lab., Stennis Space Ctr., MS 39529-5004)

It is well known that submesoscale ocean processes such as nonlinear internal gravity waves can have a significant effect on both the amplitude and the phase of an acoustic field propagating through this type of ocean environment. We report here on hindcasts computed with a numerical model combination consisting of a submesoscale hydrodynamic solver to compute a set of three-dimensional (3-D) environmental (sound speed) volumes evolving in time and a 3-D wide-angle parabolic equation code for acoustic field computation within each environmental volume. The data set was chosen from the ASIAEX 2001 experiment in the South China Sea, during a period of strong internal wave activity. A nonlinear wave packet was simulated propagating up the shelf and passing through both the acoustic source and receiver positions. The hindcasts computed the time evolving beam response on a horizontal array, located approximately 19 km from a 300 Hz

low-frequency modulat source. Comparison of experimental and modeled beamformed results was made using three different submesoscale codes to compute the dynamic ocean environment. Each code was exercised using the judgment of a different oceanographer, given the same set of measured bathymetry and water column properties. [Work supported by the Office of Naval Research.]

2:00

1pAO3. Phase fluctuations and horizontal refraction of low-frequency sound signals in shallow water in presence of internal waves. Mohsen Badiey, Jing Luo (College of Marine and Earth Studies, Univ. of Delaware, Newark, DE 19716, badiey@UDel.Edu), Boris Katsnelson, Alexander Tskhoidze (Voronezh State Univ., Voronezh 294006, Russia), James Lynch (Woods Hole Oceanograph. Inst., Woods Hole, MA 02543), and James Moum (Oregon State Univ., Corvallis, OR 97331-5503)

Fluctuations of the sound pulse phase front propagating approximately perpendicular to the direction of motion of a train of intensive internal waves in the experiment Shallow Water 2006 (SW06) are studied. Acoustic data received by a horizontal/vertical line array (WHOI-Shark array) during 3 h are analyzed. Low-frequency modulated signals with a carrying frequency of 300 Hz were transmitted, propagating approximately along coastal line at the source-receiver distance of 20 km. During this time, a train of intense internal waves (Rosey) was moving toward the New Jersey coast. Internal waves were being monitored by the ship radars (aboard R/V Sharp and R/V Oceanus) and by the thermistor chains. It is shown that phase fluctuations of the acoustic pulses arise with the appearance of internal waves at the source-receiver acoustic track and they correspond to horizontal refraction of the sound pulses. For the low-frequency sound pulses, modal decomposition of received signals is used both for the processing of experimental data and for the theoretical analysis. Experimental data are compared with theoretical estimation of horizontal refraction. [Work was supported by ONR code 321OA and by CRDF, Grant BP3C10].

2:15

1pAO4. Anisotropic properties of long-time intensity fluctuations of midfrequency signals in presence of intense internal waves in shallow water. Boris Katsnelson, Valery Grigorev (Dept. of Phys., Voronezh State Univ., 1 Universitetskaya sq, Voronezh 394006, Russia, katz@phys.vsu.ru), and James Lynch (Woods Hole Oceanograph. Inst., Woods Hole, MA 02543)

Intensity fluctuations of sequence of midfrequency pulses radiated for 5 h in SW06 experiment were considered. Frequency band of pulses is 2–10 kHz; they were radiated by the source (R/V Knorr) with the interval about 15 s and received by four hydrophones (SHRUs) placed at the distances from 4 to 14 km and different directions (SW, SSW, NE, and NNE) from the source. During this time period, remarkable train of internal waves (amplitude of displacement up to 15 m) passed toward coastal line at the velocity of 0.6 m/s. Angles between wave front of train and acoustical tracks were about 65–80 deg. Mechanism providing sound fluctuations in this case is mode coupling (for the low-frequency signals) or additional scattering of rays by internal waves (in high- or midfrequency situation). It was shown that temporal fluctuations of the sound intensity of received pulses initiated by internal waves have spectra, depending on the direction of propagation of internal wave relative acoustic track. Results of comparison of experimental data with theoretical estimations demonstrate good consistency. [This work was supported by ONR and RFBR.]

2:30

1pAO5. Variability of the water column sound speed profile and its effect on acoustic propagation during the Shallow Water 2006 Experiment. Megan S. Ballard and Kyle M. Becker (Appl. Res. Lab. and Graduate Program in Acoust., Penn State Univ., P.O. Box 30, State College, PA 16804)

Water column sound speed variability can have a significant effect on acoustic propagation. However, measurements of water column properties, either by ship based systems or moored instruments, are often neither spatially nor temporally colocated with acoustic measurements. Variability of the water column over the acoustic propagation path must be approximated by extrapolation. During SW06, a low-frequency sound source, broadcasting pure tones, was towed along radials from a vertical array comprised of both hydrophones and oceanographic sensors. Concurrently, and from the same ship, a towed CTD chain was used to measure the water column as a function of both time and space. Changes in sound speed as high as 15 m/s were recorded over a 5-km aperture. The CTD chain, in combination with environmental sensors on the VLA, allowed for continuous monitoring of the water column sound speed profile at both the source and receiver locations. It is shown that horizontal wavenumber predictions for the acoustic field are biased by the sound speed measured at the source compared to at the receiver. In addition, predictions of the acoustic field are improved when the range dependence of the sound speed profile is considered. [Work supported by NDSEG and ONR.]

2:45

1pAO6. Temporal fluctuations and coherencies of broadband signals observed during Shallow Water 2006. Jennifer Wylie and Harry DeFerrari (Univ. of Miami, 4600 Rickenbacker Cswy, Miami, FL 33149, jwylie @rsmas.miami.edu)

A 50 h acoustic propagation experiment conducted during SW06 included receptions of broadband signals at three ranges with single hydrophone receiver units. 100–1600 Hz centered broadband signals were propagated through identical shallow channels during periods of low- and high-internal wave energies. Here, the transition from nearly perfect coherent and stable mode/ray arrivals to the formation of short live micromodes/ray arrivals is presented. The relative influence of volume scattering and bottom scattering is seen to shift with frequency; low frequencies see a smooth reflective bottom but have interaction with internal waves induced volume fluctuations, whereas higher frequencies are increasingly more sensitive to bottom scattering. Discrete mode arrivals give way to a continuum of numerous short lived arrivals with increasing frequency. For still higher frequencies, coherence times are shorter still and at some point the phase coherence is so reduced that signal processing gain is lost and the signals are no longer detectable. Ultimately, the performance of higher-frequency sonar and underwater communication systems will be limited by these effects. The limits of phase coherent gain for the 800 and 1600 Hz signals are estimated for this range site.

3:00—3:15 Break

3:15

1pAO7. Investigation of an unusual noise phenomenon with horizontal and vertical hydrophone array data and three-dimensional propagation modeling. Georges A. Dossot, James H. Miller, Gopu R. Potty (Dept. of Ocean Eng., Univ. of Rhode Island, Narragansett Bay Campus, Narragansett, RI 02882), James F. Lynch, Arthur E. Newhall (Woods Hole Oceanograph. Inst., Woods Hole, MA 02543), and Mohsen Badiey (Univ. of Delaware, Newark, DE 19716)

Acoustic data from an experiment conducted during Shallow Water 2006 (SW06) showed an unexplained broadband noise phenomenon. While the R/V Knorr towed a J-15 acoustic source, which emitted a 93 Hz continuous-wave signal, the R/V Endeavor performed Scanfish measurements in the vicinity to characterize internal wave phenomena. The WHOI Shark horizontal and vertical hydrophone array detected the 93 Hz signal, but eventually the signal was overpowered by broadband low-frequency noise. The broadband noise may be associated with either research vessel but is uncharacteristically intense given that both vessels were greater than 25 km away from the Shark array. The complex acoustic environment due to the shelfbreak front, an internal wave packet, and the bathymetry of the continental shelf may have caused ducting of the noise field. Both three-dimensional acoustic modeling and array processing techniques will be used to characterize the unexplained noise levels. *In situ* data from temperature sensors and Scanfish measurements provide environmental information needed to accurately model the sound speed field. These methods can also be applied to other SW06 experiments that were tailored to investigate the role of internal waves in acoustic propagation—such as those carried out on the R/V Sharp. [Work sponsored by the Office of Naval Research.]

3:30

1pAO8. Nonlinear internal wave parameter extraction from images synthetic aperture radar. C. Chaya Boughan (Rensselaer Polytechnic Inst., Troy, NY 12180, boughc@rpi.edu), Timothy F. Duda, James F. Lynch, Arthur E. Newhall (Woods Hole Oceanograph. Inst., Woods Hole, MA 02543), and William L. Siegmann (Rensselaer Polytechnic Inst., Troy, NY 12180)

Enhancement of satellite synthetic aperture radar (SAR) ocean images is needed for improved predictions of nonlinear internal wave effects on acoustic amplitude, phase, and coherence. Their effective use in acoustic applications can be hampered by the false negative problem of SAR images that fail to reveal internal wave presence. Our aim is a largely automatic extraction of nonlinear internal wave features such as wave amplitude, wave front width, wave front separation, and longwave correlation length. A complex discrete wavelet transform is used to denoise contrast-enhanced SAR images, and a fingerprint recognition algorithm is applied to extract edges. The image is segmented into connected components that are analyzed to obtain estimates for nonlinear internal wave parameters. To evaluate the process, estimates are compared with temperature and other data from moorings positioned approximately 100 miles east of the New Jersey coast. Extensions of the procedure may give an acoustically useful description of nonlinear internal waves in SAR images. Assimilation of parameter estimates with volume data offers a new approach to appraising internal wave effects on propagation. [Work supported by the ONR.]

3:45

1pAO9. Sensitivity issues in acoustic mode tomography. Tarun K. Chandrayadula and Kathleen E. Wage (Dept. of Elec. and Comput. Eng., Mailstop 1G5, George Mason Univ., Fairfax, VA 22030)

Ocean acoustic mode tomography relies on a high signal to noise ratio and an accurate measure of the time of arrival for the mode signals. The low-order modes measured during tomography experiments such as the 2004 long range ocean acoustic propagation experiment (LOAPEX) are sensitive to environmental and experimental conditions. For example, internal waves can cause mode coupling and time of arrival perturbations. The source and the VLA move due to currents and tides, which results in Doppler spreading and frequency selective fading. Uncertainties in the signal affects the accuracy of tomographic methods. Previous work [Chandrayadula et al., J. Acoust. Soc. Am. 121, 3053 (2007)] mainly focused on building a channel model for the internal wave effects. This paper extends the previous work by building a more complete channel model that also includes the effects due to residual source/receiver motion. The statistics from the channel model are used to study the effect of uncertainties on the mode signals measured during LOAPEX. [Work supported by ONR Ocean Acoustics Graduate Traineeship Award.]

4:00

1pAO10. Acoustic wave scattering from submerged turbulent wakes. Tokuo Yamamoto (Appl. Marine Phys. Div., RSMAS, Univ. of Miami, Key Biscane, FL 33149)

Acoustic wave scattering in submerged turbulent wakes has been solved on the assumption of small and smooth turbulent fluctuations. The turbulent fluctuations are modeled by the Kolmogorov spectrum. The turbulence in the submarine wakes is modeled by a MIT Ocean Model by pat.gallacher @nrlssc.navy.mil. The effect of turbulent velocity fluctuation and the effect of turbulent temperature fluctuation are roughly equal on the scattering of acoustic wave from turbulent wakes (1 Hz–1 MHz). Acoustic scattering by a turbulent wake is very strong in the forward direction while it is small in the backward direction. The scattering from internal waves is modeled using the Garret–Monk spectrum. Since the frequency spectrum (1 Hz–1 MHz) of wake turbulence and that of internal waves (0.0138 Hz–10 MHz) do not overlap the two scattering mechanisms do not interact. The two mechanisms of scattering can be treated separately and then added linearly. [Work supported by ONR Code 321 OA.]

4:15

1pAO11. An overview of the 2005 YFIAE: Yellow Sea Oceanic Front and Internal Waves Acoustic Experiment. Ning Wang (Dept. of Phys., Ocean Univ. of China, 238-Songling Rd., Qingdao 266003, China, wangyu@public.qd.sd.cn), Jin Zhong Liu (Weifang College, Shandong, China), Da Zhi Gao, Wei Gao, and Hao Zhong Wang (Ocean Univ. of China, Qingdao 266003, China)

An overview of the Yellow Sea Oceanic Front and Internal Waves Acoustic Experiment 2005 conducted in the South Yellow Sea is presented. The experiment was a multi-institutional effort which acquired high quality environmental and acoustic data. Two of its goals were to observe internal waves including linear random and nonlinear and cold water mass of South Yellow Sea to understand and describe their effects on acoustic signals. The talk contains three primarilary experimental results. First, a comparison of data from VLA and thermistor string record suggests that nonlinear internal waves cause coherent variations of arrival time structure in frequency bands 600–800 Hz. Second, normal mode coupling due to the South Yellow Sea Front (SYSF) is observed during the experiment and is used to monitor the boundary motions of the SYSF. Finally, to interpret the "frequency dispersion" in inverted effective sound speed of sea bottom, a new fast algorithm for the inversion of bottom acoustic parameters is proposed.

Session 1pBB

Biomedical Ultrasound/Bioresponse to Vibration: Ultrasound Interaction with Tissues

Michael L. Oelze, Chair
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Contributed Papers

1:30

1pBB1. Evaluation of a singular-spectrum analysis algorithm for detecting two types of brachytherapy seeds commonly used to treat prostate cancer. Jonathan Mamou, Sarayu Ramachandran, and Ernest J. Feleppa (Riverside Res. Inst., F. L. Lizzi Ctr. for Biomedical Eng., 156 William St., New York, NY 10038)

Brachytherapy for prostate-cancer treatment involves permanent implantation of radioactive seeds into the gland. Reliable imaging of implanted seeds is needed to correct dosimetry errors in the operating room. A singular-spectrum analysis algorithm previously was able to detect palladium seeds over various angles between the seed and ultrasound-beam axes. We subsequently evaluated algorithm performance with iodine seeds. The algorithm extracts pairs of eigenvalues from the autocorrelation matrix of seed echo signals. Selected eigenvalues are used to compute a *P*-value for seed presence. The algorithm was applied to echo signals obtained using a 5-MHz transducer to scan seeds implanted into an acoustically transparent gel pad and a piece of *ex vivo* beef. The angle of the seed axis with respect to the beam axis was varied from normal to the beam axis to 20 deg from normal. Algorithm performance was denoted by a score computed from *P*-values. Scores for both seed types in beef varied from 70 to 40 dB over the range of angles studied. Scores computed from *B*-mode images were approximately 30 dB lower indicating superior contrast with the SSA algorithm. The SSA algorithm was successful in detecting palladium and iodine seeds over a range of angles to the ultrasound beam.

1:45

1pBB2. Derating of nonlinear high intensity focused ultrasound fields to predict millisecond boiling in tissue. Michael Canney (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, mcanney@u.washington.edu), Olga Bessonova, Vera Khokhlova (Moscow State Univ., Moscow 119992, Russia), Michael Bailey, and Lawrence Crum (Univ. of Washington, Seattle, WA 98105)

The most commonly used method for derating high intensity focused ultrasound (HIFU) fields from water to tissue is based on multiplying the acoustic intensity measured in water by an exponential factor to compensate for attenuation in the tissue path assuming linear wave propagation. Yet, in nonlinear HIFU fields, the intensity provides little information about either heating or negative and positive pressure amplitudes, which are important in predicting bioeffects. In this work, a new derating method is presented and tested for a 2 MHz high gain focused ultrasound source. Focal waveforms are experimentally measured and modeled after propagation through both water and tissue paths at output intensities of up to 24 000 W/cm^2. The focal waveforms measured after propagation through tissues were made equivalent to those obtained in water by increasing the pressure amplitude at the source. From the change in source amplitude pressure, the attenuation of the tissue was determined. The focus was then shifted to within the tissue sample, and the measured attenuation was used to calculate the time to reach 100°C. Calculations were in excellent agreement with the time measured to attain boiling in the tissue, which was only several milliseconds. [Work supported by NIH DK43881 and NSBRI SMS00402.]

2:00

1pBB3. Viscoelastic response of cylindrical vessels surrounded by gelatin and excited using impulsive ultrasound radiation force. Matthew Urban (Dept. of Physiol. and Biomed. Eng., Mayo Clinic College of Medicine, 200 First St. SW, Rochester, MN 55905, urban.matthew@mayo.edu), Daniel Rosario (Cornell Univ., Ithaca, NY 14853), Miguel Bernal (Mayo Clinic College of Medicine, Rochester, MN 55905), Wilkins Aquino (Cornell Univ., Ithaca, NY 14853), and James Greenleaf (Mayo Clinic College of Medicine, Rochester, MN 55905)

The objective of this study was to assess the potential for noninvasively identifying viscoelastic material properties in arterial vessels using ultrasound radiation force and modern computational inverse problem techniques. A rubber tube embedded in gelatin was excited at different points along the axis using impulsive ultrasound radiation force. The resulting mechanical waves were measured using a laser vibrometer and pulse-echo ultrasound. The inverse problem was cast as an optimization problem, in which the discrepancy between the measured dynamic response and the computed finite element model response was minimized with respect to viscoelastic material parameters. The model was very sensitive to the tube's viscoelastic parameters and relatively insensitive to those of the gelatin. The viscoelastic properties of the tube obtained from the inverse problem were compared to the results obtained from a dynamic mechanical analyzer (DMA). For the overlapping range of frequencies (130–190 Hz), the results of the inverse problem gave an equivalent modulus that range from 7.3–8.5 MPa, while the DMA values range from 6.0–9.2 MPa. These results have important implications for using ultrasound radiation force in noninvasive characterization of the viscoelastic properties of arteries *in vivo*. [This work was supported in part by Grant No. EB002640 from NIH.]

2:15

1pBB4. Study of Lamb wave dispersion in porcine myocardium. Ivan Nenadic, Matthew Urban, and James Greenleaf (Ultrasound Res. Lab., Mayo Clinic College of Medicine, 200 First St. SW, Rochester, MN 55905)

Diastolic dysfunction is the impaired ability of the left ventricle (LV) to passively fill during diastole and maintain its stroke volume at physiologic filling pressures and is associated with impaired relaxation and reduced compliance of LV. Knowledge of the quantitative values of the myocardial viscoelastic properties would improve the clinician's ability to evaluate diastolic dysfunction in patients. To gain better understanding of the viscoelastic properties of the LV as a function of geometry, we studied Lamb wave propagation in different directions in excised porcine free-wall myocardium. The phantom consisting of a dissected porcine LV free-wall embedded in gelatin was fixed in a water tank. A mechanical shaker was used for harmonic excitation over a range of 40–500 Hz and a 7.5 MHz ultrasound transducer was used for motion measurements. Motion amplitude and phase at different depths of the sample were analyzed to obtain dispersion curves in four orthogonal directions. The data were fitted to the Lamb wave equations and shear elasticity and viscosity coefficients were estimated. The values of μ_1 and μ_2 were 13.75 $\pm$ 0.96 and 5.86 $\pm$ 0.75 Pa s. Our results suggest that the viscoelastic characteristics of the porcine LV free-wall in orthogonal directions were similar.

2:30

1pBB5. Development of a lung tissue fatigue and failure model. Mark S. Wochner, 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 time-dependent nature of mammalian lung damage due to acoustic excitation at the lung resonance has been documented in recent experiments [Dalecki *et al.*, J. Acoust. Soc. Am. **116**, 2560 (2004)]. A finite-element based model of human lung response to low-frequency underwater sound has been developed to calculate induced shear stresses and strains, but until now we have not attempted to simulate progressive damage. Unlike for many man-made materials, there are very little experimental data from which fatigue parameters can be calculated for soft tissue. This presentation discusses our investigation into methods of modeling the cyclic fatigue known to occur in mammalian lungs exposed to low-frequency underwater sound. Attempts will be made to correlate reported thresholds for lung damage to computational results obtained by our model. Discussions of the limitations of such an approach and future applications will be discussed. [Work supported by ONR]

2:45

1pBB6. A mathematical model of very soft tissue for acoustics applications. 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 theoretical model of very soft tissue is presented. The soft tissue is considered to be composed of two continuum homogeneous media: the fluid medium and the solid medium. The average pressure, the velocity in the fluid, and the average displacement in the tissue can be obtained by averaging the equations of continuum mechanics over a scale that is large compared with the mean spacing between the solid and liquid regions but small compared to the length scale of the sample considered and also smaller than the ultrasound wavelength. The biphasic model presented here is inspired by the one originally proposed by Pride *et al.* who used it to model fluid-saturated porous materials, but here it is altered to suit the conditions for ultrasound propagation in a very soft tissue. The phase-averaged equations of continuity and momentum, as well as the stress-strain relations in the quasistatic limit, are discussed. In linear approximation, the averaged momentum equation for the liquid phase gives rise to a generalized Darcy's law.

3:00

1pBB7. Computational models of aberration in ultrasound breast imaging. Yi-Ting Shen and James C. Lacefield (Dept. of Elec. and Comput. Eng. and Robarts Res. Inst., Univ. of Western Ontario, London, Ontario, Canada)

Two methods are proposed for simulation of distributed aberration. One method models aberration as a superposition of five parallel time-shift screens with 8–11 irregularly shaped strongly scattering inclusions. The second method employs an anatomically realistic three-dimensional model of breast anatomy that includes lobular ducts, periductal and intralobular loose fibrous tissue, interlobular dense fibrous tissue, Cooper's ligaments, fat, and skin. Simulations of two-dimensional linear ultrasound propagation in the two model media and digitized breast tissue specimens were performed using a first-order *k*-space ultrasound simulator [Tabei *et al.*, J. Acoust. Soc. Am. **111**, 53–63 (2002)]. The initial field was a planar pulse wavefront with a 7.5-MHz center frequency and a 5-MHz −6-dB bandwidth. Propagation was computed over 25-mm paths. Both of the proposed models reproduce two characteristics of aberration observed in simulations using digitized breast specimens that are not included in conventional aberration models: non-Gaussian first-order statistics of arrival-time fluctuations and sharp changes in root-mean-square arrival-time fluctuation as a function of propagation distance. The two models, respectively, represent a relatively simple and a detailed approach in simulating realistic challenging aberration for in-silico testing of adaptive focusing techniques. [Research supported by an NSERC Discovery Grant.]

3:15

1pBB8. Time-domain three-dimensional Green's functions for power law media. James Kelly (Dept. of Elec. and Comput. Eng., Michigan State Univ., 2120 Eng. Bldg., East Lansing, MI 48824), Mark Meerschaert, and Robert McGough (Michigan State Univ., East Lansing, MI 48824)

Frequency-dependent loss and dispersion are typically modeled with a power law attenuation coefficient, where the power law exponent ranges from 0 to 2. Typically, these effects are modeled in the frequency domain and then time-domain results are obtained via inverse fast Fourier transform because analytical solutions in the time domain were previously not available. To address this problem, analytical three-dimensional Green's functions are derived in power law media for exponents between 0 and 2 by utilizing stable law probability distributions. For exponents equal to 0, 1/3, 1/2, 2/3, 3/2, and 2, Green's function is expressed in terms of Dirac delta, exponential, Airy, and hypergeometric functions. For exponents strictly less than 1, Green's functions are causal and expressed in terms of the Fox function. For exponents strictly greater than 1, Green's functions are also expressed in terms of Fox and Wright functions and are noncausal. For exponents equal to 1, Green's function is expressed as a stable distribution and is noncausal. However, numerical computations demonstrate that for observation points only one wavelength from the radiating source, Green's function, is effectively causal for power law exponents greater than or equal to 1.

Session 1pMU

Musical Acoustics: Dynamical Approaches in the Study of Music Perception and Performance II

Edward W. Large, Chair
Florida Atlantic Univ., Ctr. for Complex Systems, 777 Glades Rd., Boca Raton, FL 33431

Invited Papers

1:00

1pMU1. Kinematics and kinetics of music-induced movement. Petri Toiviainen (Finnish Ctr. of Excellence in Interdisciplinary Music Res., Dept. of Music, PL 35(M), Univ. of Jyvaskyla, Finland, ptoiviai@campus.jyu.fi)

Music listening is often associated with spontaneous body movements, frequently synchronized with the musical beat. While there exists an extensive body of work on synchronization of tapping, spontaneous movements to music have been investigated to a much lesser extent. The present study investigated the kinematic and kinetic aspects of spontaneous movements using a high-resolution motion-capture system. Various kinematic variables were estimated from the data, while body-segment modeling was utilized to obtain estimates of kinetic variables, such as forms of mechanical energy as well as instantaneous power produced during the movements. Although the participants produced a wide variety of movement patterns, some commonalities between them were found. On the kinematic level, it was found that musical beat was most clearly represented by movements in the vertical direction. On the kinetic level, the instantaneous internal power of the body showed clear peaks at the instants of musical beat. The results indicate that, regardless of the wide variety of spontaneous movement patterns, musical beat tends to be associated with bursts of instantaneous muscular power. This could suggest that the perception of the temporal structure of music is associated with imitation-based corporeal representations.

1:30

1pMU2. Conductors' temporal gestures: Spatio temporal cues for visually mediated synchronization. Geoff Luck (Dept. of Music, Univ. of Jyvskyl, P.O. Box 35(M), 40014 Jyvskyl, Finland, luck@cc.jyu.fi)

A growing body of research utilizies computational feature-extraction methods to quantify both music and music-related movement. The use of signal processing and visualization techniques, for example, is widespread in the music information retrieval community. At the same time, similar techniques are being applied to the analysis of movement data acquired using motion capture systems and are being increasingly used to quantify both temporal and expressive movements of musicians and others engaged in music-related activity. One such activity is conducting, and this talk will focus on one specific aspect of this activity: how a conductor conveys the beat to musicians using gesture alone. A number of studies involving a variety of response paradigms, both movement and music feature extraction techniques, and both laboratory and live settings will be presented. Results of these studies suggest that perception of the beat is less related to spatial cues than it is to temporal cues, with periods of high acceleration along the trajectory of a gesture being the best predictor of perceived beat location. In other words, conductors communicate the beat primarily by varying the speed of movement along the trajectory, not by changing the direction of movement.

Contributed Paper

2:00

1pMU3. Analysis of movements for drumstick control using surface electromyograms. Takuya Fujisawa, Naoki Iwami (Graduate School of Sci. and Technol., Ryukoku Univ., Japan), Masafumi Kinou, and Masanobu Miura (Ryukoku Univ., Japan)

Skills in controlling drumsticks correctly and appropriately in drumming are required to play rhythms without mistakes at appropriate dynamic levels of sound. Results obtained by analyzing drummers' movements from visual information, such as motion captures or camera recordings, have been reported in past studies, but analysis using biological information has been neither reported nor suggested to be employed. The aim of this study was to investigate the movements of drummers' hands and fingers by recording their surface electromyograms (EMGs) when drumming. Three amateur drummers participated in an experiment to record drummers' surface EMGs in playing single-stroke under three different conditions, such as "no-drum single-stroke," "no-drumstick single-stroke," and "normal single-stroke" at three different tempi of 80, 100, and 120 bpm. They were asked to play 16 measures of 4 beats under the tempi denoted above, and employed score in this experiment is simple, where four quarter-notes are simply allocated in a measure. Results of the experiment show that the average of surface EMG for playing no-drum single-stroke was significantly higher than that for normal single-stroke, indicating that drummers played single-stroke under lowest load in terms of muscle control. [Work supported by the HRC, Ryukoku Univ.]

2:15—2:35 Break

Invited Papers

2:35

1pMU4. Dynamics of thalamocortical circuits for sound processing revealed by magnetoencephalography. Bernhard Ross and Takako Fujioka (Rotman Res. Inst., Baycrest Ctr., Univ. of Toronto, 3560 Bathurst St., Toronto, ON M6A 2E1, Canada)

Music perception and cognition involves multi-modal processing within a wide range of neural networks working in concert. Rhythmic brain activities, or neural oscillations, are thought to play an important role in such long-range communication. How are related networks established and dynamically reconfigured in order to adapt to the ever changing auditory environment? Oscillations in the 40-Hz range (gamma band) in thalamocortical connections are proposed as a key mechanism. A common problem to delineate the behavior of 40-Hz oscillatory activity, however, is the small effect size and unknown time, courses when using noninvasive megnetoencephalography (MEG) recording. To overcome this problem, auditory stimulation with sounds containing a strong 40-Hz rhythm can be used to drive neural networks into a state of high synchrony. These areas are successfully identified by beamforming techniques, which transform MEG signals to voxel-based source images. Phase lags between primary auditory cortices and thalamus and auditory association areas suggest the information flow across the regions. Moreover, changes in the sound stimulus were observed as temporal changes of synchrony reflecting dynamic reconfiguration of neural networks. The relevance of these observations for detecting changes in sound localization will be demonstrated.

3:05

1pMU5. A hybrid model of timbre perception. Hiroko Terasawa and Jonathan Berger (CCRMA, Dept. of Music, Stanford Univ., The Knoll, 660 Lomita, Stanford, CA 94305, hiroko@ccrma.stanford.edu)

Timbre is a fundamental attribute of sound. It is important in differentiating between musical sounds, speech utterances, and characterizing everyday sounds in our environment as well as novel synthetic sounds. A hybrid model of timbre perception, which integrates the concepts of color and texture of sound, is proposed. The color of sound is described in terms of an instantaneous (or ideally timeless) spectral envelope, while the texture of a sound describes the temporal structure of the sound, as the sequential changes of color with an arbitrary range of time-scale. The computational implementation of this model represents a sound's color as the spectral envelope of a specific window, and its texture as the granularity (or microtexture) of the corresponding window. The temporal structures across windows from both color and texture parts of the model serve as the texture of a sound in a larger time-scale. In support of the proposed theory a series of psychoacoutic experiments was performed. The quantitative relationship between the spectral envelope and subjective perception of complex tones used Mel-frequency cepstral coefficients as a representation. A perceptually tested quantitative representation of texture was established using normalized echo density.

Contributed Papers

3:35

1pMU6. Auditory roughness profiles and musical tension/release patterns in a Bosnian ganga song. Pantelis Vassilakis (DePaul Univ., School of Music/Libraries, 2350 N. Kenmore Ave., JTR 207, Chicago, IL 60614) and Roger A. Kendall (Univ. of California, Los Angeles, CA 90095-1657)

Within western musical tradition, auditory roughness constitutes one of the perceptual correlates of dissonance. In a previous study [P. N. Vassilakis (2006), "The worlds of music: Culture-dependent emotional reactions to an improvisation on the mijwiz," *Proceedings of the 51stSEM:*, University of Hawaii, Manoa, HI, pp. 197– (2006) an application that calculates roughness profiles of musical pieces [P. N. Vassilakis, "SRA: An online tool for spectral and roughness analysis of sound signals," J. Acoust. Soc. Am., **120**, 3677 (2006)], based on a previously published roughness calculation model [P. N. Vassilakis, "Auditory roughness as means of musical expression," *Selected Reports in Ethnomusicology* **12**, 119–144 (2005)], was used to examine if and how tension/release patterns within a stylized improvisation on the Middle Eastern mijwiz relate to auditory roughness changes. An extension of that study, the present experiment, examines the relationship between roughness and tension/release patterns in a Bosnian ganga song. Patterns were obtained by a Bosnian ganga singer/scholar and by American-raised musicians in a perceptual experiment. Similarly to the mijwiz study, cultural background differences were associated with tension/release judgment differences as well as with differences between tension/release patterns and auditory roughness time profiles. The results provide further evidence that the concepts of musical tension and release and their relationship to auditory roughness are culture-specific.

3:50

1pMU7. Optimization of piano tunings by minimizing perceived beat loudness. David J. Carpenter and Richard L. Tutwiler (Graduate Program in Acoust., The Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, djc315@psu.edu)

In order to create an ideal piano tuning that best matches the inharmonicities and spectra of each instrument, tuning frequencies are optimized using a new beat loudness perceptual model that considers the time dependence of the frequency and amplitude of individual partials. These optimized tunings are compared to prior inharmonicity-based methods of tuning calculation. First, several pianos are prepared to an initial tuning using traditional methods. The resulting tones are recorded and analyzed in MATLAB. Each partial's frequency and amplitude are estimated at discrete time intervals over the duration of the tones. Beat rates are derived for the consonant musical intervals by employing phase-based high resolution frequency estimation techniques on coincident partials. A beat loudness model is developed based on established perceptual theories of consonance and equal loudness contours. The total perceived beat loudness is then minimized by applying optimization and global search methods that alter the tuning of each note. The resulting tunings achieve more consonance than prior tuning methods. Comparisons show subtle frequency variations in the tuning curve that adapt to each piano's unique inharmonicity, amplitude, and decay rate of individual partials.

Session 1pNS

Noise and Animal Bioacoustics: Advances in Measurement and Noise and Noise Effects on Humans and Non-Human Animals in the Environment II

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Ann E. Bowles, Cochair
Hubbs Sea World Research Inst., 2595 Ingraham St., San Diego, CA 92109

Chair's Introduction—1:00

Invited Papers

1:05

1pNS1. Assessing noise impacts on wildlife under the National Environmental Policy Act. Sheyna Wisdom (URS Corp., 2700 Gambell St., Ste. 200, Anchorage, AK 99503, sheyna_wisdom@urscorp.com)

Under the National Environmental Policy Act, authors must address environmental impacts of various anthropogenic actions on wildlife. One such impact of increasing awareness and concern is effect noise on wildlife, both during construction and operation of the project. However, biologists often have difficulty in understanding the fundamentals of acoustics and noise analysts often have difficulty in understanding the biological implications of increased noise on wildlife. As a result, inappropriate weighting metrics (such as A-weighted decibel) or time descriptors (e.g., community noise equivalent level) are often used erroneously to assess noise impacts on wildlife. Noise exposure thresholds on wildlife exist for marine mammals and fish, as mandated by the National Marine Fisheries Service. However, no such thresholds exist for terrestrial wildlife. This talk provides specific examples of how noise impacts on wildlife have been assessed using GIS-based technology, industry-accepted noise propagation models, and peer-reviewed literature in the absence of management guidelines. Examples include assessing construction noise impacts on the California coastal gnatcatcher in southern California, aircraft noise impacts on sage grouse in central California, and helicopter disturbance on caribou in Alaska.

1:25

1pNS2. On the need for context-sensitive noise impact criteria in rural communities. Richard Horonjeff (81 Liberty Square Rd. 20-B, Boxborough, MA 01719, rhoronjeff@comcast.net) and Grant Anderson (76 Brook Trail, Concord, MA, 01742)

With the growing population, infrastructure elements previously the exclusive province of urban and suburban communities are pushing their way into rural settings. In their wake comes increased noise nuisance and the need for context-sensitive community noise standards and guidelines. Major differences between urban and rural soundscapes make difficult the stretching of urban guidelines into rurality. Urban noise impact considerations are typically those of minimally increasing the pre-existing anthropogenic din; rural considerations center on maintaining the absence of din altogether. Most quantitative community noise standards are loudness based, comparing source loudness with that of the ambient. However, the desires of rural communities suggest that an audibility-based metric would better suit their needs. Current noise standards explicitly considering rural areas do so by recognizing the lower ambient sound levels in rural areas but still relying on loudness-based metrics. Thus, the need exists to examine the situation from an audibility perspective if places of solitude are to be preserved. This paper identifies the soundscape values of rural populations and compares them with those of urban dwellers. It then identifies the essential elements of noise metrics needed for rural soundscape preservation and also identifies the legal challenges encountered in protecting rural soundscapes.

1:45

1pNS3. Applying the Occupational Safety and Health Administration (OSHA) ultrasonic noise ceiling values. Martin Lenhardt (Dept. of Biomedical Eng., P.O Box 980168, Virginia Commonwealth Univ., Richmond, VA 23298-0168)

The Occupational Safety and Health Administration (OSHA) voted in 2003 to accept the American Conference of Governmental Industrial Hygienists (ACGIH) increased threshold level values (TLVs) for airborne ultrasound from the more stringent levels set previously as a result of ultrasonic sickness studies reported in the 1960s. The impedance mismatch between the air and the body, which prevents most ultrasonic energy absorption, was the rationale. The TLVs were increased by 30 dB, unless solid or liquid coupling is possible, allowing the unintended transfer of acoustic energy into the worker. The TLVs were set based on two lines of evidence: (1) detectability of directly coupled ultrasound and damage reported from ultrasonic exposure. The risk of ultrasonic exposure underwater is also addressed in direct coupling to an ultrasonic source. The measurements needed are airborne sound pressure level up to 100 kHz, waterborne sound pressure up to 100 kHz, and high frequency vibration. Only a 2–5 dB threshold shift for 13–17 kHz for over three years has been reported after airborne exposure. The eye was found to be a window into the skull for ultrasonic energy, reducing the impedance mismatch to 33 dB, weakening the argument that high skin impedance is an acceptable barrier.

2:05

1pNS4. Effects of sounds from seismic exploration on the calling behavior of bowhead whales. Susanna B. Blackwell (Greeneridge Sci., Inc., 1411 Firestone Rd., Goleta, CA 93117 susanna@greeneridge.com), Christopher S. S. Nations, Trent L. McDonald (WEST, Inc., Cheyenne, WY 82001), Charles R. Greene, Jr. (Greeneridge Sci., Inc., Goleta, CA 93117), Aaron Thode (Scripps Inst. of Oceanogr., La Jolla, CA 92037), and A. Michael Macrander (Shell Exploration and Production Co., Anchorage, AK 99503)

The westward migration of bowhead whales (*Balaena mysticetus*) was studied during summer and autumn of 2007 to examine the effects of airgun sounds on the whale calling behavior. Whale calls were recorded by 35 directional autonomous seafloor acoustic recorders (DASARs), placed in groups of seven recorders at five locations in the Beaufort Sea covering an east-west span of 280 km. The directional capability of DASARs allowed triangulation of an estimated whale position for about 130 000 calls. Call detection rates and call locations were compared to the timing of seismic operations—which took place in the center of the study area—and the estimated received levels of airgun sounds at the whale call locations. The analyses showed that seismic operations led to a significant decrease in call detection rates used as a proxy for calling rates. Within about 30 km of the seismic activities, received sound pressure levels from airgun pulses at call locations were greater than 140 dB *re* 1 μPa for about 20%–40% of calls. Quantile regression analyses showed that seismic activities were correlated to statistically significant shifts in the whales' distance from the shore, either offshore or inshore. [Study funded by Shell Exploration and Production Company, Alaska.]

Contributed Papers

2:25

1pNS5. The presence of infrasound in our everyday life. Kimberly Lefkowitz, Arno S. Bommer, and Robert D. Bruce (CSTI Acoust., 15835 Park Ten Pl. Ste. 105, Houston, TX 77084, kim@cstiacoustics.com)

There has been a great deal of research in the past ten years pertaining to infrasound. The effect on humans and animals of high levels of infrasound in both water and air has been studied. The effect of infrasound on structures was examined. In some situations, there has been paranoia over the effect of infrasound and, in other cases, infrasound has been overlooked completely when examining an acoustical problem. This study addresses an important element that remains to be studied, the prevalence of infrasound in a variety of locations. Infrasound was measured in a house, at a bus stop, at a typical office environment, and a variety of other situations that a typical person would be exposed to during the course of a week. These levels are then compared to less typical sites such as refineries, dredging areas, manufacturing plants, and community areas where noise complaints have been lodged. These comparisons give a preliminary understanding of peoples' exposure to infrasound.

2:40

1pNS6. Subjective evaluation of community noise in Canada's National Capital Region and its relation to waking levels of salivary biomarkers. David Michaud, Stephen Keith, Jason Tsang (Health Canada, 775 Brookfield Rd., Ottawa, ON K1A 1C1, Canada), and Anne Konkle (Environ. Health Sci. Bureau, Health Canada, Ottawa, ON K1A OK9, Canada)

Some research has suggested an association between long-term exposure to traffic noise and relative risk of cardiovascular disease (Babisch *et al.*, 2005; Willich *et al.*, 2006). It has been assumed that noise may act as a nonspecific stressor. Acute exposure to noise can evoke physiological and behavioral changes reminiscent of a stress response in rodents (Michaud *et al.*, 2003), but it is equivocal that this occurs in humans chronically exposed to traffic noise. This pilot project examined annoyance to community noise and salivary biomarkers known to be influenced by stressor exposure. A face-to-face interview subjectively assessed community noise for 60 residents (30 males, 30 females; mean age 41.3, SD=14.98). Traffic sound levels will be determined and respondents categorized into high (>65 dBA, Leq24) and low (<50 dBA) noise areas. Participants also provided saliva samples upon awakening, 30 min after awakening, and prior to bedtime. Concentrations of salivary biomarkers of alpha-amylase and cortisol were spectrophotometrically determined using commercial enzyme-linked immunosorbant assays. Two-way (high-noise versus low-noise) mixed-model analyses of variance will examine differences in questionnaire and salivary responses. Sex differences will be evaluated with independent *t*-tests, and polynomial regression analyses will relate salivary biomarker levels to high- or low-noise areas.

2:55—3:20 Panel Discussion

Session 1pPA

Physical Acoustics: Henry Bass Session: Frontiers in Acoustics

Richard Raspet, Cochair
Univ. of Mississippi, Natl. Ctr. for Physical Acoustics, 1 Coliseum Dr., University, MS 38677

James M. Sabatier, Cochair
Univ. of Mississippi, Natl. Ctr. for Physical Acoustics, 1 Coliseum Dr., University, MS 38677

Chair's Introduction—1:25

Invited Papers

1:30

1pPA1. Early studies of vibrational relaxation phenomena. F. Douglas Shields (Natl. Ctr. for Physical Acoust., Univ. of Mississippi, University, MS 38677, dshields@olemiss.edu)

Bass received the Silver Medal in Physical Acoustics in 2006 in recognition for the prodigious amount of work he did, largely in the area of sound absorption in the atmosphere in 1971 and 1972. When he came to Ole Miss straight from graduate school in 1970, he was 27 years old with a wife and three children. He had worked his way through college while supporting his family and served a 2 year tour of active duty in the Signal Corp before entering graduate school. He continued his service in the Army Reserve and retired in 1993 with the rank of Lieutenant Colonel. During the first three years he was at Ole Miss, he published 17 refereed papers and gave 5 papers at ASA meetings. This talk traces the path his research followed during this period in moving from his dissertation study of rotational relaxation as a function of temperature in a series of polar polyatomic gases to the successful explanation of the variation of sound absorption in the atmosphere with humidity, temperature, and frequency. Also discussed are his experiments during this period with the spectraphone and his work with Hans Bauer in developing the theory for sound amplification from controlled excitation reactions (SACER).

1:45

1pPA2. Acoustical absorption in the atmosphere at high altitudes. Louis Sutherland (Consultant in Acoust., 27803 Longhill Dr., Rancho Palos Verdes, CA 90275)

Before 1978, acoustical absorption in the atmosphere was modeled semiempirically based on numerous experimental studies, especially those of Harris and Tempest. In 1969, this author presented a semiempirical model to explain the divergence of the absorption data from theoretical values predicted from classical absorption and molecular relaxation of just moist oxygen. Joe Piercy published a correction of my error in neglecting molecular relaxation of moist nitrogen. This activity stimulated formation of an ANSI working group on atmospheric absorption chaired by Joe Piercy and included the late Henry Bass and myself as members. The result was ANSI Std. S1.25-1978 (1995) which was stated as being valid only up to 20 km. Later, I had the distinct honor to collaborate with Bass in a paper [J. Acoust. Soc. Am. **115** (2004)] which provided a computational model for atmospheric acoustical absorption up to 160 km. Unique elements of this high altitude atmospheric absorption model are reviewed, including the molecular relaxation loss from carbon dioxide. This high altitude atmospheric absorption model was possible only with the key role played by Bass in its development. The Society has lost a true scientist, scholar, and gentleman of the highest caliber.

2:00

1pPA3. Tornadoes, thunder, and underwater sound from lightning. Anthony A. Atchley (Graduate Program in Acoust., 201 Appl. Sci. Bldg., University Park, PA 16802, atchley@psu.edu)

Outdoor sound propagation was a cornerstone of Bass' research throughout much of his career. During the late 1970s, propagation of sound from natural sources became of particular interest. This talk focuses on three aspects of this phase of Bass' research: (1) efforts to localize lightning from an analysis of recordings made with a distributed ground microphone array, (2) recording the underwater sound produced by lightning striking the surface of the ocean, and (3) development of a tornado alert detector based on the acoustic signature of tornadoes.

2:15

1pPA4. Vibrational relaxation effects on rise times of weak shocks. Richard Raspet (Dept. of Phys. and Astronomy and Natl. Ctr. for Physical Acoust., Univ. of Mississippi, University, MS 38677)

After I finished my Ph.D. on the "Application of Monte Carlo techniques in the analysis of laser induced microfission explosions," I was unemployed and trying to find a job. Hank had a contract from the Air Force to investigate the generation and propagation of thunder. As part of this work, Hank had assigned an undergraduate student the task of adapting a finite wave propagation program developed by Anderson at the University of Texas to incorporate Hank's newly developed sound attenuation algorithm for the ANSI Standard S1-26, "Sound absorption in the atmosphere." The student had graduated without making much progress and Hank offered me a summer job to work on the problem. I managed to find the single error in the program and as a result, Hank and I wrote a paper [J. Acoust. Soc. Am.]. This was the start of my career in acoustics and a lifelong collaboration with Hank. We and several co-authors published eight papers based on the study of the interaction of vibrational relaxation with finite wave distortion. This paper will present a summary of the findings of this line of research over a 24 year period.

2:30

1pPA5. The influence of Hank Bass on ground effect research between 1981 and 1995. Keith Attenborough (Dept. of Design, Development, Environment and Mater., The Open Univ., Milton Keynes MK7 6AA, UK, k.attenborough@open.ac.uk)

In 1981 Bolen and Bass published a paper on ground effect, which included a pioneering effort to deduce ground impedance from propagation data. This study was a forerunner to recent work on the ANSI standard for measuring ground impedance. My first interaction with Bass in the same year stemmed from a mututal interest in observations of buried geophone responses to airborne sound sources. Our initial experiements involved understanding the response of buried microphones to airborne sound and the relationship of ground properties to the phenomenon of ground effect. Subsequently, work with Bass and colleagues inspired treatments of the ground as a layered poroelastic medium. As a result of Bass' initiative, the first 12 Long Range Sound Propagation Symposia (LRSPS) was held in Mississippi. This helped to introduce researchers interested in atmospheric acoustics to the propagation codes used in underwater acoustics. The LRSPS initiative, together with Bass' involvement with the NATO RSG11 study group, resulted in benchmarks for testing the various atmospheric propagation codes that were available in 1995.

2:45

1pPA6. The interactions of Henry E. Bass with the National Research Council in Canada. Gilles Daigle, Tony Embleton, David Havelock, Joseph Piercy, and Michael Stinson (Inst. for Microstructural Sci., Natl. Res. Council, Ottawa, ON K1A 0R6, Canada)

Interactions between researchers at NRC and Bass span a period of more than 30 years during which many research areas of mutual interest were explored. This paper presents an overview of Bass' research in atmospheric sound propagation during this period and the synergistic collaborations that developed with his friends and colleagues at the NRC. Early work began with the study of relaxation in gases and absorption of sound in the atmosphere. This expanded to include sound propagation above a finite impedance ground, the effects of refraction and atmospheric turbulence, and the development of fast numerical codes to predict long-range sound propagation. The close collaboration and enduring friendships that we valued so much were forged over the years through many interactions, such as the numerous joint field experiments trudging through muddy fields, working in subzero temperatures, or suffering the blistering desert heat. Throughout, Bass was a driving force, and his contributions will fuel research for the community at large for some time to come.

3:00—3:15 Break

3:15

1pPA7. Work of Hank Bass in the mid-eighties to develop a comprehensive model of sound propagation through the atmosphere. Michael White (U.S. Army ERDC/CERL, P.O. Box 9005, Champaign, IL 61826-9005)

The success of stealth technologies in the mid-eighties simultaneously raised the importance of other possible modes of discovery, one of them being acoustic detection. Bass initiated work on a comprehensive model for predicting sound transmission through the atmosphere in order to better assess the propagation. The model first considered atmospheric absorption and ground reflection of sound and would render predictions for single frequencies and spectra. Variability in received signals seemed to be explained by the strong effects of atmospheric temperature refraction, wind advection, and turbulence, but few direct comparisons of measurement to theory existed. To address this, Hank formed several collaborations using ray-tracing, fast-field, and parabolic equation methods for modeling refraction and advection, leading to benchmarks for outdoor sound propagation in 1993. Measurements of the short-term distributions of phase and amplitude over a few hundred meter distance revealed separate scattering regimes, with each being accessible by theory and numerical computation.

3:30

1pPA8. Coupling of airborne sound into the earth. James Sabatier (NCPA, Univ. of Mississippi, University, MS 38677)

While a student at the University of Southwestern Louisiana in 1982, I was attracted to the Physical Acoustics Research Group at the University of Mississippi after reading a list of physical acoustics publications left by Professor Gordon Baird on a graduate student recruitment trip. I made a brief visit to UM and met Hank, Bolen, and Crum. Hank offered me an assistantship ten times greater than the one at USL. I recall him saying the U.S. Army had a seismic sensor to detect tanks, but when helicopters flew overhead, the sensors

alarmed. Aspects of this work were described in the paper, of Bass and Bolen ["Coupling of airborne sound into the earth: Frequency dependence," J. Acoust. Soc. Am. **67**, (1980)]. During graduate years at UM, I met Keith Attenborough and was introduced to the work of Keith, Hank, and Lee on microphones buried in soils and rigid frame porous media acoustics. Under their direction, my graduate acoustics education was spent applying Biot poroelasticity to first foot of the ground. In this talk, I will summarize the acoustic-to-seismic work that was done with Hank.

3:45

1pPA9. Hank Bass and the origins of the National Center for Physical Acoustics. Lawrence Crum (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105)

The National Center for Physical Acoustics was established by an Act of Congress due principally to the efforts of Henry Bass. This presentation will review the history behind the NCPA and the various roles that Hank played in making this center a reality.

4:00

1pPA10. Optoacoustics in liquids and gasses. Stanley Cheyne (Dept. of Phys. Astronomy, Hampden-Sydney College, Hampden-Sydney, VA 23943)

Optoacoustics, the production of sound from light, was studied experimentally and theoretically during the 1980s at the University of Mississippi. Bass's primary interest in optoacoustics was to understand the physical processes at the molecular level. In one paper [Ali *et al.*, "Spectrophone measurements in sulfur hexafluoride," IEEE **UFFC-33**, 615 (1986)], a detailed analysis of the energy transfer processes from IR laser radiation to the dissipation of the generated sound wave was investigated. In a second paper [Thompson *et al.*, "Optoacoustic observation of internal relaxation in liquid CS_2," J. Acoust. Soc. Am. **85**, 2405 (1989)], it was shown that a liquid with relatively slow relaxation times affected the temporal shape of the optoacoustic signal. Finally, in a third paper [S. A. Cheyne and H. E. Bass, "Observation of optoacoustic amplitude in CS_2 at high input energies," J. Acoust. Soc. Am. **88**, 1842 (1990)], it was shown that the amplitude of the optoacoustic signal changed nonlinearly with input energy due to molecular interactions.

4:15

1pPA11. The wave propagation in porous material: A continuum from Biot solids to thermoacoustic heat engines. William Arnott (Dept. of Phys., Univ. of Nevada Reno, MS 220, Reno, NV 89557, patarnott@gmail.com)

Wave propagation in poroelastic materials has applications ranging from ground motion induced by atmospheric sound to thermoacoustic heat engines. Bass was involved with research in this entire continuum, including both basic and applied problems. This paper will especially cover the wave propagation formulation of thermoacoustics as it was applied by Bass and others to understand and predict the behavior of thermoacoustic prime movers. Some recollections of Bass as a statesman, scientist, and enjoyable character to be with will be presented. My first meeting with Bass and some of his guidance will also be discussed.

4:30

1pPA12. Henry Bass' contributions to the infrasound renaissance: Notes from the field. Milton Garces (Infrasound Lab., Univ. of Hawaii at Manoa, 73-4460 Queen Kaahumanu Hwy., Kailua Kona, HI 96740-2638, milton@isla.hawaii.edu)

The deployment of the international monitoring system global infrasound network at the turn of the 21st century inspired a renaissance in innovation, development, and application of infrasound technology. In the United States, Hank Bass was responsible for defining and directing the scientific agenda, and he skillfully navigated the turbulent waters of national and international policy through a decade of administration changes. In addition, he created and nurtured a cohesive community of collaborating partners in academia, industry, and government. This presentation honors Hank's contributions to the field of infrasound within the context of past, present, and future research efforts in the U.S. and abroad.

Session 1pPP

Psychological and Physiological Acoustics: Potpourri (Poster Session)

Rahul Shrivastav, Chair
Univ of Florida, Communication Sciences and Disorders, 336 Dauer Hall, Gainesville, FL 32611-7420

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.

1:30

1pPP1. Nonzero threshold loudness: A circular argument in Zwislocki (1965) and Moore, Glasberg, and Baer (1997). Lance Nizami (1312 Grayson Pl., Decatur, GA 30030, nizamii2@aol.com)

ANSI S3.4-2005 updates the calculation of loudness to reflect empirical evidence that loudness is nonzero at detection threshold. Hellman [Acoustics Today (2007)], in reviewing the changes, cited the work of Zwislocki [*Handbook of Mathematical Psychology* (1965) Vol. III] and Moore *et al.* [J. Audio. Eng. Soc. **45** (1997)] as theoretical support. Zwislocki proposed that detection threshold reflects an internal noise that acts like an external masker, and that (using generalized notation here for clarity) the tone+noise loudness for an rms tone pressure x is a function f of the sum of noise contribution(s) c and a tone contribution $g(x)$. The Zwislocki $g(x)$ was zero at $x=0$ and increased monotonically thereafter. Altogether, tone+noise loudness is $f(g(x)+c)$. Zwislocki then assumed that listeners can perceptually separate tone from noise. He subtracted noise loudness from tone+noise loudness to get tone loudness, $[f(g(x)+c)-f(c)]$. That exceeds zero for whatever x is deemed the tone-detection threshold. Unfortunately, with $g(x)>0$ for $x>0$, a nonzero threshold-tone-loudness was predetermined. Moore *et al.*, using auditory filter output power as x, produced a congruent equation for tone loudness in quiet, repeating the circularity. Circularity was inevitable, because detection threshold is defined using a percentage-correct performance that indicates nonconstant loudness.

1pPP2. Tone-glide direction identification revisited. Lawrence Feth and Evelyn Hoglund (Speech and Hearing Sci., Ohio State Univ., Columbus, OH 43210)

Dawson and Feth (2004) reported that listeners showed an apparent difference in sensitivity for the direction of frequency change when listening to virtual frequency glides or frequency modulated (FM) tones. Their work was a partial replication and extension of the work by Gordon and Poeppel (2002) using a one-interval fixed-block paradigm. In each interval, the listener was asked to report whether the frequency was rising or falling using buttons labeled up and down. Psychometric functions were reported but the original data were not examined for interval response bias. This study addresses possible bias in two ways. The original data were reexamined for response bias, and direction identification was tested using FM glides in the single-interval adjustment-matrix (SIAM) procedure. The SIAM procedure [Kaernbach, (1990)] is designed to control response bias in a one-interval paradigm. Contrary to the previous work, data collected with the SIAM procedure do not show a consistent pattern of sensitivity toward change in either direction. Additionally, psychometric functions using this procedure are steeper than those of the prior studies. Implications of these results in comparison with previous work will be discussed. [Work supported by a grant from NIH/NIDCD R01-DC006879].

1pPP3. Informational masking of virtual frequency glides. Gayla Poling and Lawrence Feth (Dept. of Speech Hearing Sci., The Ohio State Univ., 110 Pressey Hall, 1070 Carmack Rd., Columbus, OH 43210, poling.75@osu.edu)

Informational masking (IM) experiments use multiple-burst maskers to change masker uncertainty. The effect of IM is believed to be mediated by interference in the central auditory processing of the signal. Virtual frequency (VF) glides are produced by amplitude modulating two sinusoids separated by one or more ERB at a given test frequency. The amplitude of the low-frequency tone is initially ΔI dB more intense than the higher-frequency tone, and then falls linearly as the amplitude of the higher frequency rises to be ΔI dB greater at the end. Listeners report hearing a weak rising pitch for such signals although no signal energy moves through the frequencies between the tones. Our hypothesis is that central auditory system processing of these signals is similar to that for frequency-modulated (FM) sinusoids, despite clear differences in their peripheral excitation patterns. This study was designed to investigate the effects of masker uncertainty on dynamic signals (FM and VF glides) for different ERB separations for the anchor frequencies. Preliminary results demonstrate a smaller release from IM for the VF glides, which may be explained by the salience of the VF signals compared to the FM tones. [Work supported by NIH/NIDCD RO1-DC006879.]

1pPP4. The effects of off-frequency listening on the pitch of narrowband complexes. Veronica Eckstein-Lilley and Bruce Berg (Dept. of Cognit. Sci. Univ. of California Irvine, 3151 Social Sci. Plaza, Irvine, CA 92697, veckstei@uci.edu)

A pitch-matching paradigm was used to investigate the filtering properties of the auditory system. Listeners adjusted the frequency of a pure tone to match the pitch of a five-tone complex. Each complex had five equal amplitude sinusoidal components with equal frequency spacing (20 Hz). Five of six total conditions constituted a signal added in phase to any one of the five components, while the sixth condition added no signal. Two different signal levels were tested. Pitch matches were best described by the envelope weighted average of the instantaneous frequency, but only if the stimulus was first passed through the skirt of an auditory filter [V. Eckstein and B. G. Berg, J. Acoust. Soc. Am. **120**, 3126 (2006)]. The results appear to be more consistent than those previously obtained with three-tone complexes. By using additional components and adding a signal at the edges of the signal complex, this study depicts a more complete inspection of the auditory filter shape, reduces pitch ambiguity, and eliminates nonlinearities. [Work supported by NSF.]

1pPP5. An acoustic comparison of nonlinguistic sounds to sentences spoken in American English. Corine Bickley (Gallaudet Univ., 800 Florida Ave. NE, Washington, DC 20002, corine.bickley@gallaudet.edu) and Yell Inverso (Pennsylvania College of Optometry, Elkins Park, PA 19027)

Acoustic characteristics of nonlinguistic (nonspeech) sounds (NLSs) were measured for duration and spectral variation, and compared to acoustic characteristics of spoken sentences (TIMIT database). The NLS, included

samples produced by animal, human, mechanical, and natural sources. The acoustic comparison examined stoplike onsets, fricativelike intervals, vowel-like intervals, and syllablelike variations in amplitude. The NLSs were identified by two groups of listeners: listeners with normal hearing and users of cochlear implants. Results of the listening tests have been reported previously by Inverso *et al.* (2007) and Inverso (2008). All of the NLSs were identified accurately by listeners with normal hearing, but not by the users of cochlear implants. The current analysis focuses on the ways in which the NLSs are similar to and other ways in which they are different from sentences spoken by a variety of talkers. It was found that speechlike variation in amplitude, both in terms of duration and event onset/offset, was a strong cue for listeners with cochlear implants; that is, NLSs that contained distinct events that were similar in duration and amplitude variation to syllables in speech were identified more accurately than ones that did not.

1pPP6. Statistical inference in the perceptual learning of non-native speech category. Yuan Zhao (Dept. of Linguist., Stanford Univ., Stanford, CA 94305, yuanzhao@stanford.edu)

The study investigated the influence of category bias and category variance in English speakers' learning of two lexical tone categories (high versus low). In Experiment 1, all subjects completed an *AX* discrimination pretest of adjacent tone pairs on a ten-step pitch continuum. They were then trained to associate a "low tone" with the meaning "pink" and a "high tone" with the meaning "blue" via a picture-sound association task. Subjects received either a biased training with a 5:1 ratio between high and low tones or an unbiased training with a 1:1 ratio between the two categories. It was then followed by an identification post-test. The results showed that poor pitch discriminators categorized significantly more sounds into the biased category. Good discriminators categorized significantly more sounds into the other category. Therefore, good discriminators have a novelty preference, while poor discriminators have a familiarity preference. In Experiment 2, subjects received either a small or a big variance training. The results showed that poor discriminators performed significantly better if trained with a big-variance input. However, good discriminators' categorization performance was significantly better if trained with an input of a small variance. The study suggests that statistical cues are utilized differently among subjects of different auditory sensitivities.

1pPP7. Acoustic discrimination and categorization of formant dynamics for novel and familiar stimuli. Daniel Fogerty (Dept. of Speech and Hearing Sci., Indiana Univ., 200 S. Jordan Ave., Bloomington, IN 47405, dfogerty@indiana.edu)

Perception of a formant transition continuum from /b/ to /d/ was tested for a high-fidelity speech continuum and an analogous sinewave speech continuum. Endpoint stimuli were recorded by a male talker, matched for fundamental frequency and duration, and interpolated using a STRAIGHT morphing algorithm to produce a continuum of 11 steps. The first three formants were replaced with sinewaves to create a sinewave speech continuum. This study used a between-subject paradigm, which enabled comparison of highly familiar natural stimulus categories to novel sinewave stimulus categories while preserving fundamental acoustic cues for perception (the first three formants). Categorization training and discrimination blocks were interleaved during testing to track the effect of category learning on discrimination. Results for sinewave speech failed to demonstrate significant short-term category learning or a change in discrimination performance across three blocks. However, discrimination and categorization performance were fundamentally different between continua. While discrimination for natural speech was best near the category boundary, participants who heard sinewave speech appeared to use detailed differences in pattern dynamics, resulting in best discrimination near specific stimuli characteristic of the dynamic pattern. Results will be discussed as they relate to categorical perception and category learning. [Work supported by the NIH.]

1pPP8. Measuring listening effort independent of task difficulty. Mini Shrivastav, Rahul Shrivastav, and James Harnsberger (336 Dauer Hall, Dept. of Commun. Sci. and Disord., Univ. of Florida, Gainesville, FL 32611, mnarendr@csd.ufl.edu)

The effects of task difficulty on many listening tasks are often confounded by those of the effort expended during the task. It is important to have valid measures of effort independent of task difficulty. The present experiment was aimed at doing so by measuring various markers of effort for listening conditions of varying difficulty but with equal performance. Task difficulty in a tone-detection experiment will be manipulated by changing the level of a broadband noise added to the background so that the signal-to-noise ratio changes from favorable to unfavorable. The participants will be a group of young normal-hearing individuals. Effort will be measured using several tools such as self-reported questionnaires and various physiological markers such as galvanic skin response, heart rate, and blood oxygen level. For each background noise condition, the percentage of correct detection performance will be calculated. Conditions that result in greater than 90% accuracy will be considered as having equal performance. Any difference in physiological measures between these conditions will then likely reflect differences in listening effort. The study will provide preliminary candidates for developing an objective and real-time measure of listening effort.

1pPP9. A functional model of the slope sensitivity of the primary auditory cortex neurons in awake cats. Kenji Ozawa, Yoshikazu Koike, Hiromi Wakagi, Yu Sato, and Sohei Chimoto (Univ. of Yamanashi, 4-3-11 Takeda, Kofu 400-8511, Japan, ozawa@yamanashi.ac.jp)

Our measurement of single-spike responses to tone-burst stimuli showed that there are heterogenic response types of the primary auditory cortex (A1) neurons in awake cats: phasic neurons sensitive to stimulus amplitude-rise-slope at the onset of stimuli, sustained cells sensitive to the stimulus slope, and the steady state level at later time-phase of stimuli. In this study, in order to explain the behavior of the slope-sensitive phasic cells, an auditory model has been developed. The model consists of six stages: the inner hair cell (IHC), the primary auditory nerve (AN), the cochlear nucleus (CN), the inferior colliculus (IC), the medial geniculate body (MGB), and the A1 neuron. The model was implemented as a MATLAB program on a personal computer using Meddis's model for IHC [R. Meddis, J. Acoust. Soc. Am. **79**, 702-711 (1986)] and Maki's model for AN, CN, and IC [Maki *et al.*, J. Acoust. Soc. Jpn. **60**, 304-313 (2000)], while the models of MGB and A1 neuron were newly developed. The model output successfully replicated the physiological data in terms of response rise-time, maximum driven rate, and onset-latency corresponding to the slopes of input stimuli. [Work supported by KAKENHI 20300076.]

1pPP10. The sound field reproduction method based on the spacial covariances and its reproduction efficiency. Hiroki Hagiwara, Yoshinori Takahashi (Kogakuin Univ., 1-24-2 Nishi-shinjuku, Shinjuku-ku, Tokyo 163-8677, Japan, hagiwara@b09.itscom.net), Mikio Tohyama (Waseda Univ., Tokyo 169-8050, Japan), and Kazunori Miyoshi (Kogakuin Univ., Tokyo 163-8677, Japan)

This paper describes a sound field reproduction method based on the spatial covariances. Wave surface control and convoluting the head related transfer function are major techniques for sound field reproduction. On the contrary, Takahashi and Tohyama, coauthors of this report, considered that the spatial impressions of a sound field, which we perceive through our ears, are reproduced by preserving the relative relationship between the observation points even if the wave surface is not completely controlled. Moreover, they proposed a new sound field reproduction method that can control the point-to-point covariance in a sound field [19th ICA, RBA15-012]. In this work, we demonstrated about the differences between the covariance based method and the conventional methods, such as stereophonic playback and boundary surface control based on the Kirchhoff–Helmholtz equation. Then, we discussed the performances of several methods on the point of view of reproductivity and device cost.

1pPP11. Improvement of scale factor for time-domain audio watermarking based on low-frequency amplitude modification. Harumi Murata, Akio Ogihara, Motoi Iwata, and Akira Shiozaki (Dept. of Comput. Sci. and Intelligent Systems, Osaka Prefecture Univ., 1-1 Gakuen-cho, Naka-ku, Sakai, Osaka 599-8531, Japan, murata_h@ch.cs.osakafu-u.ac.jp)

The objective of this work is to improve the sound quality for the audio watermark method based on amplitude modification. We improve the sound quality by modifying the scaling curve smoothly. The audio watermarking method based on amplitude modification has been proposed by Lie as a prevention technique against copyright infringement. The watermark information is embedded into audio signals in the time domain. One-bit watermark information is embedded by modifying the differences of average-of-absolute-amplitude from three sections in a group of samples. At the section boundary points, the scaling factors are forced to 1.0 to ensure continuity. The factors are then progressively increased or decreased to the stable value near the center of section. However, when the adjacent sections were both increased or both decreased, signal discontinuities happen in this boundary point and cause the "click" sounds that are perceivable to human ears. In this paper, we aim to make the audio waveform continuous and smooth with the scaling factors of the section boundary points that are not set to be 1.0 but are determined according to the stable values of the adjacent sections. By the proposed method, sound quality can be improved in objective and subjective evaluations.

1pPP12. Applying acoustic ray tracing from the full wave equation solution to determine the nonair conduction pathways into the human head. Jared McNew, Alessandro Bellina (Dept. of Elec. and Comput. Eng., Univ. of Illinois, B428C Beckman Inst., 405 N. Mathews, Urbana, IL 61801), and William D. O'Brien, Jr. (Univ. of Illinois, Urbana, IL 61801)

Flight deck crews are subject to high intensity sounds for prolonged periods of time. Even while using current hearing protection devices such as ear plugs, ear muffs, and helmets, hearing loss is still possible. It is likely that this hearing loss is due to the cochlear stimulation through pathways other than the usual air conduction pathways. In order to develop new hearing protection devices to prevent hearing loss due to these nonair conduction pathways, ray tracing from the full wave equation solution is being used to determine and visualize these dominant pathways. Numerical techniques for performing ray tracing are presented along with the validation of the method by comparing with Snell's law. For small changes in sound speed, angle of transmission errors are less than 20%. As the frequency of the incident pulse is increased, these errors are reduced as would be expected. Results of ray tracing performed on simulation data from two concentric fluid spheres as an approximation to the human head are also presented. Using this model, an insertion loss of 26 dB is observed. [Work supported by AFOSR FA9550-06-0128.]

1pPP13. Simulation of bone-conducted sound pathways to the inner ear. William O'Brien (Bioacoustics Res. Lab., Dept. of ECE, Univ. of Illinois, Urbana, IL 61801)

A project to determine bone-conducted sound pathways to the inner ear is sponsored by the Air Force in an effort to reduce hearing loss in personnel working in high-noise environments. The study includes a computer simulation, in three dimensional (3D), of an acoustic pulse wave propagating through and around a human skull. Numerical 3D simulations are benchmarked with known analytic results in order to verify accuracy. The pulse center frequency can be varied in order to identify how frequencies affect the vibrations of the skull. The program is used to validate an experimental mapping of the skull in which the response in the inner ear, with and without hearing protection, is measured as function of transducer input location at different external head locations and center frequency. An FETD algorithm written in MPI and executed on highly parallelized clusters is used to achieve these results efficiently. The mesh, on which the code operates, is a uniform grid of eight node brick elements. Thus the program can directly operate on any digitized image (in two dimensional) or a list of digitized images (in 3D) in which each image is a single slice of a 3D volume. [Work supported by AFOSR FA9550-06-0128.]

1pPP14. Numerical simulation of mechanisms of sound conduction through the human head with a fast elasto-acoustics integral equation solver. Elizabeth Bleszynski, Marek Bleszynski, and Thomas Jaroszewicz (Monopole Res.,739 Calle Sequoia, Thousand Oaks, CA 9160)

The purpose of this paper is to describe our progress in development and applications of numerical simulation tools designed to (a) investigate mechanisms of energy transfer to cochlea through air- and bone-conduction pathways and to (b) assess effectiveness of suitable noise protection devices. Our numerical simulations employ a recently implemented acoustoelastic integral equation solver, capable (through the use of nonlossy fast Fourier transform based matrix compression algorithm and parallelization on distributed-memory systems) of accurate large-scale numerical simulations with anatomically realistic models of the human head, discretized with several millions of tetrahedral elements. Recently added new solver features (including treatment of shear waves, described in terms of node-based linear elements) enable us to analyze detailed aspects of acoustic/elastic wave interaction with the human head with significant accuracy and efficiency. In particular, they facilitate examination of the relative amounts of energy transferred to the middle and inner ears directly through the outer ear and indirectly via excitation of elastic waves in the skull and soft tissues. [This work is supported by AFOSR.]

1pPP15. Influence of static force on bone conduction threshold measurement. Lynn Brault, Woojae Han, Charissa Lansing, Ron Chambers (Dept. of Speech Hearing Sci., Univ. of Illinois at Urbana-Champaign, 901 S. Sixth St., Champaign, IL 61820), Jared McNew, Alessandro Bellina, and William D. O'Brien (Univ. of Illinois at Urbana-Champaign, Urbana, IL 61801)

The influence of static force on bone conduction threshold measurement behavioral bone conduction thresholds are evaluated in clinical settings using a standard headband. The standard headband is designed to apply 5.4 N of static force to the oscillator (ANSI S3.6-1996 R2004), but in practice the static force varies with head size over a range of several Newtons. The influence of the varied force on behavioral thresholds is uncertain. These thresholds have clinical utility in determining the type and severity of hearing loss and research utility in modeling bone-conducted sound pathways. Behavioral bone conduction thresholds for a group of normal-hearing young adult listeners (ages 18–30) were obtained in 1-dB steps as a function of several force levels (2 N, 5 N, variable force) using custom-calibrated and standard headbands for mastoid and forehead oscillator placements across a frequency range from 250 to 8 kHz in sixth-octave bands. No significant differences in behavioral thresholds were obtained as a function of the static force levels tested, although the results supported previous findings that bone conduction thresholds are more sensitive for mastoid than forehead placement. The impact of static force levels on performance for speech understanding with bone conduction listening systems requires further study. [Work supported by AFOSR FA9550-06-0128.]

Session 1pSC

Speech Communication: Perception of Words, Sentences, and Indexical Information (Poster Session)

Sarah H. Ferguson, Chair
Univ. of Kansas, Dept. Speech-Language Hearing Sciences and Disorders, 1000 Sunnyside Ave., Lawrence, KS 66045

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.

1pSC1. Subjective ratings of sentences in clear versus conversational speech. Sarah Hargus Ferguson and Emily E. Kerr (Dole Ctr., Dept. of Speech-Lang.-Hearing: Sci. and Disord., Univ. of Kansas, 1000 Sunnyside Ave., Rm. 3001, Lawrence, KS 66045)

Investigations of the relationship between speech acoustics and intelligibility face a significant methodological challenge, especially when using materials produced by multiple talkers and/or in multiple speaking styles. Ideally, acoustic analyses comparing talkers or styles will use identical materials for each. In contrast, intelligibility studies using meaningful stimuli must use either different materials or different listener groups to avoid learning effects. The latter solution becomes prohibitive when the number of talkers is large, such as in the Ferguson clear speech database (2004). While talkers recorded different materials in each speaking style, materials were the same for all talkers. For word intelligibility, familiarization prior to testing can prevent learning effects. For sentences, however, different listeners would be needed for each talker, resulting in 41 listener groups. An alternative solution is to use subjective rather than objective measures. This project explored the feasibility of this approach. Young listeners with normal hearing rated the clarity of sentences produced by eight talkers from the Ferguson database. Rated clarity was significantly higher for clear versus conversational sentences and varied significantly among talkers. Further analyses will compare clarity to perceptual and acoustic measures such as vowel intelligibility, speaking rate, and vowel space characteristics.

1pSC2. Acoustic correlates of speaker discrimination in English. Chandan Narayan (Inst. for Res. in Cognit. Sci. Univ. of Penn., 3401 Walnut St. Ste. 400A, Philadelphia, PA 19104) and Jiahong Yuan (Univ. of Pennsylvania, Philadelphia, PA 19104)

This study investigated speaker discrimination in utterances varying in syllable length and speaker gender taken from the TIMIT corpus of American English. Twenty native English speakers presented one-, two-, and three-syllable utterances (within speaker gender) in a two-alternative forced-choice task. Perception results were analyzed in light of both source level ($F0$ and long-term average spectrum of LPC residuals) and formant level measurements ($F1$–$F4$). Results showed that male speakers were discriminated better than female speakers. Source features ($F0$ and LTAS of LPC residuals) significantly predicted listener response, while higher spectral information ($F1$–$F4$) had little effect. The varying importance of vocal source and vocal tract characteristics in speaker discrimination is discussed.

1pSC3. Effects of talker and token variability on perceptual learning of dialect categories. John K. Pate and Cynthia G. Clopper (Dept. of Linguist., Ohio State Univ., 1712 Neil Ave., Columbus, OH 43210, jpate@ling.osu.edu)

Dialect classification is difficult for naive listeners, but perceptual learning tasks using sentence-length utterances have been shown to produce modest improvements in performance. The goal of the current study was to explore perceptual learning by naive listeners in a speeded dialect classification task with shorter word-length utterances. In a series of experiments, participants were trained in a two-alternative forced-choice dialect classification task (Cleveland versus Cincinnati) with feedback and were then tested in the same task with novel talkers and novel words without feedback to assess learning. Variability in the stimulus materials in the training phase, including the number of talkers from each dialect and the number of different tokens produced by each talker, was manipulated across experiments to determine how variation in the input affected perceptual learning of dialect categories. The results revealed that training materials consisting of utterances produced by multiple different talkers from each dialect with multiple different tokens produced by each talker led to a significant improvement in dialect classification performance compared to a baseline condition without training. These findings suggest that dialect classification performance can improve with training on short word-length utterances, but that robust dialect category learning requires high variability stimulus materials.

1pSC4. Channel segregation improves perception of speech with temporally desynchronized bands. Michael Kiefte (School of Human Commun. Disord., Dalhousie Univ., Halifax, NS B3H 1R2, Canada, mkiefte@dal.ca), Christian E. Stilp, and Keith R. Kluender (Univ. of Wisconsin, Madison, WI 53706)

Stilp *et al.* [J. Acoust. Soc. Am. **122**, 2971 (2007)] investigated the intelligibility of temporally desynchronized bands of speech and concluded that listeners's resilience to temporal asynchrony may possibly be due to differences in fundamental frequency (f_0) across syllables which may help in segregating bands. This hypothesis is tested in two experiments in temporally desynchronized bands but with spectral manipulations designed to either aid or inhibit stream segregation. Seven-syllable sentences were synthesized at three different speaking rates and processed by four nonoverlapping 1/3-octave filters. Onsets of the lowest- and highest-frequency bands were parametrically delayed. In Experiment 1, f_0 in delayed bands was uniformly elevated using pitch-synchronous overlap add synthesis. In the control condition (no f_0 manipulation), intelligibility was nonmonotonic with delay across speaking rates with local minima corresponding to the duration of one syllable. The two-speaker manipulation decreased spectral similarity across bands making intelligibility more resilient to temporal distortion. In Experiment 2, f_0 contours were flattened. Intelligibility was uniformly poorer than in the control condition; the increased spectral similarity compromised listeners' ability to segregate information from band pairs at various delays. Acoustic measures of potential information, absent explicit linguistic information, reinforce the strong relationship between spectral predictability and intelligibility. Overall, results provide support that channel segregation plays an important role in the perception of temporally desynchronized bands. [Supported by NIDCD]

1pSC5. Interaction between speech coding and semantic predictability in speech perception. Nirmal Srinivasan and Thomas D. Carrell (Dept. of Special Education and Commun. Disord., Barkley Memorial Ctr., Univ. of Nebraska-Lincoln, Lincoln, NE 68583, nirmal@bigred.unl.edu)

The present experiments investigated how humans process speech presented in the form low-passed natural sentences (5000 Hz), cell phone coded sentences, and computer synthesized sentences. Spin sentences [Kalikow *et al.*, (1977) and Bilger *et al.*, (1984)] were presented in a background of multispeaker babble using these three coding representations. Word accuracy and visual-motor task performance were measured. Intelligibility was based

on the participants' accuracy in repeating the final word of the sentence. Attention was based on the participants' performance on the simultaneous pursuit rotor task [Srinivasan and Carrell (2007)]. In the pursuit rotor task, a stylus was used to maneuver a cursor on a computer display with the goal of keeping the cursor aligned as closely as possible to a moving target. Word accuracy was higher for predictable sentences than for unpredictable sentences for all three encodings. However, pursuit rotor performance showed a complex set of interactions. For example, participants performed better on the pursuit rotor task for unpredictable sentences than predictable sentences, but only for the cell phone speech not for natural or synthetic speech. The findings suggest that the perception of cell phone speech does not degrade the same way that natural or synthetic speech does.

1pSC6. Effect of level of presentation on scaled speech intelligibility of speakers with dysarthriafurther data. Yunjung Kim (Dept. of Commun. Sci. and Disord., Louisiana State Univ., Baton Rouge, LA 70803), Gary Weismer, Raymond Kent (Dept. of Communicative Disord., Univ. of Wisconsin-Madison, Madison, WI 53705), and Joseph Duffy (Mayo Clinic, Rochester, MN)

This study seeks to examine the effect of level of presentation on scaling of speech intelligibility of speakers with dysarthria. Previous data [Kim *et al.* (2007)] reported the somewhat surprising result that different presentation levels do not cause a significant change in speech intelligibility scores of speakers with dysarthria, but do for healthy speakers. The current study reports further data using the same speech stimuli, however, with a main focus on the across-conditions effect of presentation level, since the previous data were limited to within-conditions effects. Four conditions of level of presentation were generated by varying absolute level of presentation (high and low) and normalizing each utterance for peak vowel intensity (adjusted and nonadjusted). In this presentation, we will report how scaled speech intelligibility is affected by the level of presentation of speech samples produced by speakers with various types and severity of dysarthria and by healthy speakers. The results of this study will be helpful to understand the rationale of speech therapy for dysarthria, especially considering the frequent use of speech intelligibility and speech therapy techniques designed based on intensity variation of speech. [Work supported by NIH DC00319 and internal fund from Louisiana State University.]

1pSC7. Evaluation of two voice stress analyzers. Harry Hollien and James Harnsberger (IASCP, 46 Dauer Hall, Univ. of Florida, Gainesville, FL 32611)

The purpose of this study was to evaluate two commonly used voice stress analyzers: NITV's computer voice stress analyzer (CVSA) and Nemesysco's layered voice analysis (LVA) system. In both cases, a speech database was used, which contained materials recorded (1) in the laboratory, while highly controlled deceptive and shock induced stress levels were systematically varied and (2) during a field procedure. Subjects were 24 males and females (age range 18–63 years) drawn from a population representative of the United States. All held strong views on an issue and were required to make sharply derogatory statements about it. The systems were then evaluated in a double blind study using two sets of examiners: (1) two UF scientists trained/certified by the manufacturers and (2) either three experienced CVSA operators or two LVA instructors provided by the manufacturer(s). The results for both devices showed that the "true positive" (or hit) rates ranged from chance to somewhat higher levels—50% to 65 %—for all conditions/types of materials (stressed-unstressed, truth, or deception). However, the false positive rate was just as high—often higher. Sensitivity statistics demonstrated that these systems operated at about chance levels. [Work supported by Counterintelligence Field Agency, DoD.]

1pSC8. Getting phonetic structure from an aphonetic signal. Joanna H. Lowenstein and Susan Nittrouer (Dept. of Otolaryngol.–Head and Neck Surgery, Ohio State Univ., 4331 Cramblett, 456 W. 10th Ave., Columbus, OH 43210, lowenstein.6@osu.edu)

Several studies have shown that adults can use a global structure such as that found in sine wave and amplitude envelope signals for speech perception. Recent work has demonstrated that children can perceive speech processed as sine wave signals as well as the adults, but have more difficulty perceiving vocoded signals. If speech perception can occur even when signals are abstracted down to the level of sine wave speech, why then does the human vocal tract produce a signal with such complexity? Perhaps the redundancy encoded in the complexity of the speech signal is useful for other tasks, such as phonemic awareness and short-term recall. This study presented three lists of eight near-rhyming words in sine wave speech, speech in 0-dB noise, and unprocessed speech to adults and 8-year olds as a short-term serial recall task. Adults performed the same on the unprocessed speech and speech in noise, and performed more poorly on the sine wave speech. 8-year olds, on the other hand, performed identically in all conditions, but made more errors overall than the adults. Adults who were not able to understand sine wave speech had poorer overall recall scores on the other two tasks. [Work supported by NIDCD Grant No. DC-00633.]

1pSC9. Effects of acoustic transformation on cross-modal speech information and audiovisual gain II. James W. Dias and Lorin Lachs (Dept. of Psych., California State Univ. Fresno, 2576 E. San Ramon ST11, Fresno, CA 93740)

Auditory and visual perceptual processes interact during the identification of speech sounds. Some evaluations of this interaction have utilized a comparison of performance on audio and audiovisual word recognition tasks. A measure derived from these data, *R*, can be used as an index of the perceptual gain due to multisensory stimulation relative to unimodal stimulation. Recent evidence has indicated that cross-modal relationships between the acoustic and optical forms of speech stimuli exist. Furthermore, this cross-modal information may be used by the perceptual mechanisms responsible for integrating disparate sensory signals. However, little is known about the ways in which acoustic and optic signals carry cross-modal information. The present experiment manipulated the acoustic form of speech in systematic ways that selectively disrupted candidate sources of cross-modal information in the acoustic signal. Participants were then asked to perform a simple word recognition task with the transformed words in either auditory-alone or audiovisual presentation conditions. It was predicted that audiovisual gain would be relatively high for those transformations in which the relative spacing of formants was preserved but would be nonexistent for those transformations that destroy the relative spacing of formants. The results are discussed in terms of existing theories of audiovisual speech perception.

1pSC10. Channel and language effects on reaction time in a voice identification. Kyna Fasnacht and Ruth Huntley Bahr (Dept. of Commun. Sci. Disord., Univ. of South Florida, 4202 E. Fowler Ave., PCD 1017, Tampa, FL 33620)

Previous research has indicated that listeners have difficulty with bilingual speaker identification when voice samples are compared across languages [Bahr and Frisch (2002)]. Communication channel can negatively impact voice identification, but its effect on bilingual speaker identification has not been well defined. Reaction time has also been found to increase for speech processing when speaker accent is strong [Schmid and Yeni-Komshian (1999)]. Therefore, it is the purpose of this investigation to assess the effect of channel differences and language to determine how these factors influence recognition accuracy and processing time. A paired comparison listening task was used to evaluate monolingual listeners' performance when channel (laboratory quality, landline phone, and cell phone transmissions) and language (English versus various dialects of Spanish) were varied. Reaction time was recorded to investigate changes in processing demands. Results indicated that listeners were less accurate when channel conditions varied. Difficulties with language/dialect were more noticeable in conversation. These findings were evident in both the accuracy and reaction time data. The role of dialectal variation in the forensic speaker identification process will be described and related to findings from the stereotype literature, which describes "an other race effect," i.e., all dialects of Spanish sound the same.

1pSC11. Using eye-tracking and long-term repetition-priming to examine talker-effects in spoken word recognition. Conor T. McLennan (Dept. of Psych.-CB175, Cleveland State Univ., 2121 Euclid Ave., Cleveland, OH 44115, c.mclennan@csuohio.edu) and Paul A. Luce (Univ. at Buffalo, SUNY, Buffalo, NY 14260)

Our research examines the circumstances in which talker variability affects listeners' perception of spoken words. In our previous work, we used the long-term repetition-priming paradigm in which listeners were presented

with two blocks of spoken words. Some words were repeated from one block to the next and some words heard during the second block had not been presented during the first block. We found that repeated words were responded to more quickly than new words (a repetition-priming effect). Moreover, repeated words spoken by the same talker in both blocks were responded to more quickly than repeated words spoken by different talkers (a talker-effect). Crucially, talker-effects emerged relatively late during perceptual processing. In the current study, we extended our previous work by using the eye-tracking paradigm and by investigating whether an attentional manipulation would modulate the role that talker variability plays during the perception of spoken words. Because eye-tracking has been shown to tap into processing quite early, we hypothesized that talker-effects would only be obtained if listeners' attention were drawn to the talkers during the first block. Results will add to our knowledge of the circumstances in which talker variability affects the perception of spoken words. This research was supported in part by research Grant No. 5 R03 DC 7316-4 from the National Institute on Deafness and Other Communication Disorders, National Institutes of Health.

1pSC12. Alternative data analysis techniques when using the eye-tracking paradigm to investigate spoken word recognition. Conor T. McLennan (Dept. of Psych., Cleveland State Univ., CB175 Cleveland, OH 44115, c.mclennan@csuohio.edu)

Researchers have been using the eye-tracking paradigm as a tool for investigating the representations and processes involved in listeners' perception of spoken words for over a decade now (thanks in large part to the work by Tanenhaus and colleagues). Moreover, this paradigm is becoming increasingly widespread, with the number eye-tracking researchers, laboratories and publications increasing rather substantially over the past few years. Although the eye-tracking paradigm has been used to investigate a number of different issues related to spoken word recognition, including lexical competition, parallel activation in bilinguals, and ambiguity resolution, most (if not all) of these studies use participants' eye fixations as the unit of measurement when performing statistical analyses (e.g., fixations on a picture that corresponds to the spoken word heard during a trial). Fixations are a perfectly fine source of data; indeed we have learned a lot about spoken word recognition from studies performing analyses of fixation data. However, there are different ways to use fixations, as well as various alternative data analysis techniques, that researchers using the eye-tracking paradigm to investigate spoken word recognition may wish to consider. [Work was supported in part by the National Institute on Deafness and Other Communication Disorders, National Institutes of Health (Grant No. 5 R03 DC 7316-4).]

1pSC13. Investigating lexical influences on the accuracy of speechreading words presented in isolation and in sentence context. Edward Auer, Jr. and Rebecca Reed (Dept. of Speech-Lang.-Hearing, Univ. of Kansas, 1000 Sunnyside Ave., Rm. 3001, Lawrence, KS 66045, auer@ku.edu)

Isolated spoken words that are perceptually similar to many other words in the mental lexicon are typically more difficult to recognize than words that are perceptually unique. This effect has been demonstrated for spoken words presented in both auditory and visual presentation conditions. Here, this effect of perceptual similarity on visual recognition (speechreading) of spoken words was compared for words presented in isolation and in sentential contexts. Sentences were developed containing key words that varied in predicted perceptual uniqueness. 123 (41-easy, 41-moderate, 41-difficult) spoken sentences were presented visual-alone to 40 participants for identification. 301 keywords spoken in isolation were presented visual-alone to 23 participants for identification. In both presentation conditions, identification accuracy varied as a function of the perceptual similarity of the target word to other words in English. Specifically, identification accuracy decreased as the number of perceptually similar words increased. Furthermore, the effect of perceptual similarity interacted with presentation condition (isolated words versus sentence). The results suggest that the typical advantage afforded by perceptual uniqueness is reduced when words are presented in a sentence context. Implications for future modeling studies and for predicting the intelligibility of sentence length materials will be discussed. [Work supported by an NIH/NIDCD R01DC04856.]

1pSC14. Examining the relationship between lexical access and the perceived strength of foreign-accents. Amee P. Shah (Dept. of Health Sci., Cleveland State Univ., 2121 Euclid Ave., MC 423, Cleveland, OH 44115, a.shah101@csuohio.edu) and Conor T. McLennan (Cleveland State Univ., Cleveland, OH 44115)

Foreign-accented speech is one source of variability that listeners may face during the perception of spoken language. Although a number of previous studies have examined the role that foreign-accented speech plays in listeners' ability to process spoken language, we are interested in examining the converse of this relationship. That is, previous work has demonstrated that accent manipulations affect listeners' ability to process spoken language. For example, the speed and accuracy with which listeners access spoken words produced with a foreign-accent is compromised (at least initially) relative to the same words produced with a native-accent. However, the present study, which builds on our previous work [Shah and McLennan (2007)], examines whether manipulations known to affect listeners' ability to access spoken words will affect listeners' subjective judgment of foreign-accented speech. More specifically, because spoken words are typically easier to process when they have been heard recently (repetition priming), we investigated whether listeners would respond more quickly in an accent-rating task to primed words relative to unprimed words and/or whether primed words would be perceived as having been produced with a weaker foreign-accent. Results provide new information regarding the relationship between lexical access and the perception of foreign-accented speech.

1pSC15. Effects of noise, cognitive demand and lexical characteristics on word recognition by normal hearing native listeners. Astrid Zerla Doty, Catherine L. Rogers (Dept. of Commun. Sci. and Disord., Univ. of South Florida, 4202 E. Fowler Ave., PCD1017, Tampa, FL 33620), and Judith Becker Bryant (Univ. of South Florida, Tampa, FL 33620)

Although listeners can adapt to many challenging listening conditions, the combined effects of multiple co-occurring challenges on speech understanding are less well known. The present study examines the combined effects of background noise, cognitive demand, and lexical characteristics on isolated word recognition. Young normal-hearing native English speakers heard six lists of 24 words, each composed of 12 lexically "easy" and 12 lexically "hard" target words in an open-set word identification task. Easy words had high target word frequency, few phonological neighbors, and low-neighborhood frequency; conversely, hard words had low frequency, many neighbors, and high-neighborhood frequency. Word lists were presented in quiet, in a moderate degree of background noise, and with or without a competing digit recall task. In the digit recall task, listeners saw three or six digits on the monitor prior to presentation of the word list and were asked to recall the digits at the end of the word recognition task. Word recognition scores and the size of the "easy" word benefit will be compared in quiet, in noise, and with and without the digit recall task. The accuracy of digit recall will also be compared across conditions of background noise and number of digits presented.

1pSC16. Time course of talker-specific learning in spoken word recognition. Lynne C. Nygaard, Sabrina K. Sidaras, and Jessica E. D. Alexander (Dept. of Psych., Emory Univ., Atlanta, GA 30322, lnygaar@emory.edu)

The current study examined the effects of talker-specific perceptual learning on the time course of spoken word processing to assess the point at which effects of talker familiarity emerge during spoken word recognition. Listeners learned to identify six talkers' voices (three males, three females) over three days of training. At test, listeners either completed an immediate or a delayed word-shadowing task. Items at test were novel words produced by the six familiar talkers heard during training and by a set of six unfamiliar talkers. A separate group of controls completed the test phase only. The results indicated that effects of talker familiarity were found in both the immediate and delayed shadowing tasks. Listeners were faster to shadow words produced by familiar than unfamiliar talkers both when responding immediately to the word and when responding after a cued delay. When tested with unfamiliar talkers, the trained listeners did not differ from untrained controls. These findings suggest that effects of talker familiarity emerge relatively early in spoken word processing and persist as word recognition unfolds. Perceptual learning of talker-specific characteristics appears to influence the time course of lexical processing by facilitating the rapid recovery of the linguistic structure of spoken words.

1pSC17. Representation of multiple variant forms in spoken word recognition: Friends, not enemies. Larissa Ranbom, Eleni Pinnow, and Cynthia Connine (Dept. of Psych., Binghamton Univ., P.O. Box 6000, Binghamton, NY 13902, lranbom1@binghamton.edu)

Prior research [Ranbom and Connine (2007)] showed that alternative variant forms of a word are represented in the lexicon as a function of variant frequency. Two experiments investigated whether these multiple representations conspire or compete during spoken word recognition of nasal flap variants. In experiment 1, nasal flap productions of high (i.e., "counter") and low (i.e., "enter") variant frequency words, as well as a set of nonword controls (i.e., "penter"), were presented in a phoneme monitoring task (detection of a /t/ sound). The results showed "higher-t" detection rates for words than for nonwords, and higher-t detection for high than for low-variant frequency words. Experiment 2 used truncated versions of the stimuli from experiment 1 and showed comparable t-detection rates across all conditions. The results suggest that coactivated multiple representations of phonological variants conspire during spoken word recognition.

1pSC18. Effects of lexical status within phonetic categories. Sam Soleimany and James Sawusch (Dept. of Psych., Univ. at Buffalo, Park Hall, Buffalo, NY 14260-4110, sss26@buffalo.edu)

Previous work has shown an interaction between lexical knowledge and phonetic categorization. For example, Ganong (1980) showed that ambiguous initial phonemes were identified so that the resulting syllable was a word. While previous studies have emphasized lexical influences at the phonetic category boundary and their time-course, the present work examines lexical influence effects within categories. Changes in within category discrimination for word and nonword tokens were examined with an ABX discrimination paradigms. Signal detection analysis was employed to determine whether changes in rating responses within a phonetic category were attributable to changes in sensitivity, bias, or both. While changes in identification and discrimination between phonetic categories may be attributable to postperceptual biases, changes within categories may not be. Results will be discussed in regard to their impact on theories of word recognition and whether lexical influences are prelexical, postlexical, or both.

1pSC19. Speech perception in a language-trained chimpanzee (Pan troglodytes). Lisa Heimbauer, Michael Beran, and Michael Owren (Lang. Res. Ctr., Georgia State Univ., 3401 Panthersville Rd., Decatur, GA 30034, lheimbauer1@student.gsu.edu)

After decades of research, the question of whether humans perceive spoken language using a specialized "speech mode" remains unresolved. Studies in nonhumans suggest that animals perceive phonemic contrasts much as humans do, but involve subjects trained for thousands of trials on single discriminations. This work reports initial speech perception results from Panzee, a chimpanzee (*Pan troglodytes*) reared by humans speaking to her as they would to a child and also training her to use graphical wordlike "lexigrams." Panzee comprehends approximately 126 spoken words, documented through a procedure in which a digitally presented spoken word is matched to one of four lexigrams presented on a monitor. First experiments have compared performance with natural digitized versions of 24 spoken words to synthetic LPC-based replicas and to whispered versions. Using a different subset of eight test words within each of three 96-trial sessions showed comparable mean performance for natural (83.3%), synthesized (82.5%), and whispered (78.5%) versions. Percent-correct performance on 24 trials representing the first time a given test word was heard was also comparable for synthesized (79.2%) and whispered (78.5%) sounds. The possibility that Panzee is showing speech-mode perception will be tested in experiments with noise-vocoded and sine-wave speech. [Work supported by NICHD.]

Note: Payment of additional registration fee required to attend tutorial.

MONDAY EVENING, 10 NOVEMBER 2008 LEGENDS 8, 7:00 TO 9:00 P.M.

Session 1eID

Interdisciplinary: Tutorial Lecture on Aircraft Noise Prediction

Lily M. Wang, Chair
Univ. of Nebraska, Lincoln, Architectural Engineering, 1110 S. 67th St., Omaha, NE 68182-0681

Chair's Introduction—7:00

Invited Paper

7:05

1eID1. Aircraft noise prediction. Joe Posey (NASA Langley Res. Ctr, M.S. 461, Hampton, VA 23681, joe.w.posey@nasa.gov)

Noise has been an issue for airport communities and passengers since the advent of commercial air transport almost a century ago. Impressive gains have been made in aircraft noise control, but expectations are rising and air traffic will at least double in the next 20 years. Also, lighter composite structures pose a challenge for interior noise control. Furthermore, societal expectations for mobility, new technology, demands for carbon footprint minimization, and other environmental imperatives will lead to revolutionary aircraft designs in the future. This tutorial lecture will consider noise from present and future subsonic jet aircraft, rotorcraft, propeller aircraft, and supersonic transports. Noise prediction capabilities will be identified, along with an overview of noise control technology. Acousticians are preparing to predict and control community and interior noise for arbitrary configurations using models based more on first principles, whereas the state-of-the-art is largely semiempirical. It is imperative that acousticians be included in the early stages of the design process and for all design team members to have some exposure to noise control principles to avoid wasting resources on designs that should be nonstarters from the noise perspective.