Session 5aPA

Physical Acoustics: General Topics in Physical Acoustics II

Sam Wallen, Chair
The University of Texas at Austin

Contributed Papers

8:00

5aPA1. Generation of ultrasonic waves in vegetable tissue via non-ablative laser pulses. David A. Collazos-Burbano (Ctr. for Bioinformatics and Photonics (CIBioFi), Universidad del Valle, calle 13 No. 100-00, Univalle, 351, of 2009, Cali, Valle del Cauca 760032, Colombia, david.collazos@correounivalle.edu.co), Carlos A. Melo-Luna (Quantum Technologies, Information and Complexity Group (QuanTIC), Universidad del Valle, Cali, Valle del cauca, Colombia), Joao L. Ealo (Vibrations and Acoust. Lab. (LaVA), Mech. Eng. School, Universidad del Valle, Cali, Colombia), and John H. Reina (Quantum Technologies, Information and Complexity Group (QuanTIC), Universidad del Valle, Cali, Valle del Cauca, Colombia)

The study of vegetable tissue is essential in plant growth analysis. The elasticity is an important characteristic when firmness is studied in fruits or when turgidity is tested in plant leaves. This property gives information about the ripeness of fruits, the water content in leaves, and, in general, the morphological and physiological state of plants. Hence, the propagation of elastic waves in these media is a useful tool that gives information about the developmental stage of the plant. In this sense, laser ultrasonics is a promising method in the characterization of composite materials. In a certain way, vegetable tissues can be thought of as a kind of biological composite abstracted as a stratified plate of several layers. In this work, as a first approximation, we propose a methodology to generate guided waves in vegetable tissues by using laser ultrasonics techniques. We implement a pulsed laser system to induce ultrasonic waves without damage on the samples. Besides, we develop a methodology for processing the propagated acoustic waves. This analysis allows us the assessment of dispersive properties, a useful tool when determining important descriptors such as firmness index, water content, and provide useful information to feed biophysical models, among others.

8:15

5aPA2. Probing dynamic response to temperature changes in Berea Sandstone using acoustic time-of-flight and resonant ultrasound spectroscopy. Eric S. Davis, Cristian Pantea, Blake Sturtevant, and Dipen N. Sinha (Los Alamos National Lab., P.O. Box 1663, Los Alamos, NM 87545, esdavis@lanl.gov)

In this study, through-transmission acoustic time-of-flight experiments and bulk resonance tracking were used to study the anomalous elastic behavior in Berea Sandstone with temperature cycling. Previously published results (Davis *et al.*, 2016) using Resonant Ultrasound Spectroscopy (RUS) indicated that Berea Sandstone softens by as much as 10% when cooled from 205°C to 110°C. This result is in contrast to the behavior of most materials, which stiffen when cooled. The time-of-flight experiments confirmed the RUS results and revealed the need for additional analysis of this phenomenon. By tracking the location of resonant frequencies at each temperature step, the qualitative behavior of the sample was obtained without the need for a rigorous RUS analysis. The results from this analysis showed that Berea Sandstone has significant hysteretic behavior under temperature cycling. It was also revealed that the qualitative elastic behavior of Berea Sandstone is unchanged with increasing relaxation time from the thermal shock induced by rapid cooling of the sample.

8:30

5aPA3. Elastic constants of self-healing polyethylene co-methacrylic acid determined via resonant ultrasound spectroscopy. Kenneth A. Pestka II, Jacob W. Hull, Jonathan D. Buckley (Chemistry and Phys., Longwood Univ., 201 High St., Farmville, VA 23909, pestkaka@longwood.edu), and Stephen J. Kalista (The Dept. of Biomedical Eng., Rensselaer Polytechnic Inst., Troy, NY)

Self-healing polyethylene co-methacrylic acid (EMAA) is a thermoplastic material that can be shaped into structures that are capable of autonomously healing after being cut, punctured, or shot. In this work, we present results from several samples of self-healing EMAA-0.6 Na, with 60% of the methacrylic acid groups neutralized by sodium, known as DuPont Surlyn 8920. This ionomer, unlike typical crystals with extremely high quality factors and easily identifiable ultrasonic resonances, is a relatively soft polymeric material with significant damping, resulting in sample resonances that can overlap and are often difficult to identify and isolate. In this work, we show the resonant spectrum of several different EMAA-0.6 Na samples as well as the methods used to isolate individual resonances and determine the elastic constants.

8:45

5aPA4. An ultrafast broadband digital signal processing approach to pulse echo measurements. Blake Sturtevant, Dipen N. Sinha, and Cristian Pantea (Mater. Phys. and Applications, Los Alamos National Lab., MPA-11, MS D429, Los Alamos, NM 87545, bsturtev@lanl.gov)

Pulse-Echo (PE) measurements are commonly used to determine the sound velocity in a sample of known length. In PE measurements, the interface between the transducer and buffer rod, and that between the buffer rod and sample, introduces a total phase shift, Φ, to the acoustic wave that must be accounted for if high accuracy in the velocity is desired. The appropriate time correction, $\tau=\Phi/\omega$, is traditionally determined by measuring the time-of-flight of multiple acoustic waves with different angular frequencies, ω. A single PE measurement can take several minutes, depending on experimental details such as the number of frequencies measured and the amount of signal averaging performed. Several minutes of data acquisition is tolerable for static experiments, but it is too long for many dynamic processes of interest in biology, pulsed magnetic fields, or any other rapidly changing system. This work describes an approach to PE in which a single broadband signal is collected and later processed offline using standard digital signal processing techniques. We demonstrate that a single captured waveform can be processed to yield the same information as 80 separate waveforms collected over an 80 MHz span, enabling accurate PE measurements on sub-second time scales.

9:00

5aPA5. Two-point statistics of synthetic three-dimensional polycrystalline microstructures. Musa Norouzian and Joseph A. Turner (Dept. of Mech. and Mater. Eng., Univ. of Nebraska-Lincoln, W342 Nebraska Hall, Lincoln, NE 68588-0526, jaturner@unl.edu)

The two-point spatial statistics of a heterogeneous material are crucial in ultrasonic attenuation and backscatter calculations. In this presentation, several common assumptions used for modeling polycrystals are examined. In most studies, the spatial and tensorial elements of the covariance function are assumed to be decoupled. Also, two-point correlation models often assume a single effective grain size for polycrystalline materials and a spatial correlation function that has an exponential decay at a rate directly proportional to the average grain diameter. This investigation uses the Dream.3D software platform to generate three-dimensional (3D) realizations of polycrystalline materials in order to study their two-point statistics. Using lognormal distributions with fixed means but different widths, a variety of different polycrystalline microstructures are simulated. The results show that the spatial correlation function is directly influenced by the grain size distribution. For a fixed mean grain size, wider distributions exhibit longer range correlations than narrower ones. The results also show that decoupling of the tensorial and spatial elements of the covariance function may not be valid for all lognormal grain size distributions. These results are anticipated to have an important impact on ultrasonic attenuation and backscatter models. [Research supported by AFRL under prime contract FA8650-15-D-5231.]

9:15–9:30 Break

9:30

5aPA6. Uncertainty bounds on ultrasonic phase velocities for polycrystalline media. Musa Norouzian and Joseph A. Turner (Dept. of Mech. & Mater. Eng., Univ. of Nebraska-Lincoln, W342 Nebraska Hall, Lincoln, NE 68588-0526, mnorouzian@huskers.unl.edu)

Most theoretical work related to the study of the effective properties of polycrystals assume infinite media with randomly oriented grains. Therefore, the bulk material has absolute isotropy. However, real samples always include a finite number of grains such that the inspection volume will have some associated anisotropy. Hence, bounds on the bulk properties can be expected for a given measurement. In this work, the effect of the number of grains on this anisotropy variation is studied using Dream.3D software. The effective elastic modulus tensor is derived using Voigt, Reuss, and self-consistent techniques with 1700 microstructures comprised of equiaxed cubic grains in 17 different volumes. The bond transformation is utilized to quantify the standard deviation of the average elastic modulus. The standard deviations of several materials are shown to be inversely proportional to the square root of the number of grains. Based on the single-crystal anisotropy, a master curve is derived which relates modulus anisotropy to the number of grains. In addition, Christoffel equation is used to study the relevant phase velocities. With appropriate normalization, a similar master curve is derived. Such results will have important impacts on ultrasonic models associated with metals. [Research supported by AFRL under prime contract FA8650-15-D-5231.]

9:45

5aPA7. A study on warning sound for drowsiness driving prevention system. Ik-Soo Ann (Cultural Contents, Soongsil Univ., 369 Sangdo-ro. Dongjak-gu, Seoul 156-743, South Korea, aisbestman@naver.com)

The Korea Expressway Corporation has studied various preventive measures to prevent drowsiness driving accidents, such as providing various drowsiness driven prevention measures, establishing drowsiness rest areas, and operating alarms on drowsiness driving in highway tunnel have been implemented. Recently, the Korea Transportation Safety has also used the monitoring of the driven state (DSM—Driven State Monitoring) and the driving information of the vehicle (VDI—Vehicle Driving Information) to confirm the state of the driver in the running car and perform a drowsiness driving and we are showing a good reaction by introducing a drowsiness driven prevention system which sounds vibration and warning sound. As described earlier, when a driver wakes up during a drowsiness operation, a warning sound plays an important role by installing it in a tunnel or in a car. In this paper, we further developed the warning sound for battle against drowsiness used in the tunnel of the existing highway and studied focusing on using inside the vehicle. This research aims to verify and further improve the warning sound part of the in-vehicle drowsiness driven prevention system which is just introduced and enforced, effectively and contribute effectively to prevent drowsiness driven accidents.

10:00

5aPA8. A study on the comparison of characteristics between blower and sound fire extinguisher. Seonggeon Bae (Div. of Comput. Media Information Eng., Kangnam Univ., 40, Kangnam-ro, Giheong-gu, Youngin-si, Gyeonggi-do, Youngin 446-702, South Korea, sgbae@kangnam.ac.kr) and Myungjin Bae (Information and Commun., Soongsil Univ., Seoul, South Korea)

In general, the characteristics of the blower used for cleaning and the characteristics of the sound fire extinguisher have the same point that they use wind. However, unlike a blower that allows the wind to clean by the wind, a sound extinguisher is a principle that turns off the fire by blocking the air of oxygen necessary for ignition in the air. This study was carried out by separating the two features.

10:15

5aPA9. A robust smartphone based multi-channel dynamic-range audio compressor for hearing aids. Yiya Hao (Engineer and Comput. Sci., Univ. of Texas at Dallas, 380 Vistacourt Dr., APT2317, Plano, TX 75074, yxh133130@utdallas.edu), Ziyan Zou (Engineer and Comput. Sci., Univ. of Texas at Dallas, Garland, TX), and Issa M. Panahi (Engineer and Comput. Sci., Univ. of Texas at Dallas, Richardson, TX)

Hearing impairment degrades perception of speech and audio signals due to low frequency-dependent audible threshold levels. Hearing aid devices (HADs) apply prescription gains and dynamic-range compressor for improving users' audibility without increasing the sound loudness to uncomfortable levels. Multi-channel dynamic-range compressor enhances quality and intelligibility of audio output by targeting each frequency band with different compression parameters such as compression ratio (CR), attack time (AT), and release time (RT). Increasing the number of compressor channels can result in more comfortable audio output when appropriate parameters are defined for each channel. However, the use of more channels increases computational complexity of the multi-channel compressor algorithm limiting its application to some HADs. In this paper, we propose a nine-channel dynamic-range compressor with an optimized structure capable of running on smartphones and other portable digital platforms in real time. Test results showing the performance of proposed method are presented too.

Session 5aPP

Psychological and Physiological Acoustics: Examining the Auditory Brain

Ross K. Maddox, Chair
Biomedical Engineering and Neuroscience, University of Rochester, 601 Elmwood Avenue, Box 603, Room 5.7425, Rochester, NY 14642

Contributed Papers

8:00

5aPP1. Human middle-ear muscles rarely contract in anticipation of acoustic impulses. Heath Jones (APPD, USAARL, Bldg. 6901, Fort Rucker, AL 36362, heath.g.jones2.ctr@mail.mil), Nate Greene (APPD, USAARL, Aurora, CO), and William A. Ahroon (APPD, USAARL, Fort Rucker, AL)

The Auditory Hazard Assessment Algorithm for Humans (AHAAH) is an electrical equivalence model of the human ear designed to predict the risk for auditory injury from a given impulse noise exposure. One concern with the model is that the middle-ear muscle contraction (MEMC) associated with the acoustic reflex is implemented as a protective mechanism for certain instances in which a person is "warned" prior to the impulse. The current study tested the assumption that the MEMC can be elicited by a conditioning stimulus prior to sound exposure (i.e., a "warned" response). In order to quantify the MEMC, we used laser-Doppler vibrometry to measure tympanic membrane motion in response to a reflex-eliciting acoustic impulse. After verifying the MEMC, we attempted to classically condition the response by pairing the reflex-eliciting acoustic impulse (unconditioned stimulus, UCS) with various preceding stimuli (conditioned stimulus, CS). Changes in the magnitude and/or time-course of the MEMC following repeated UCS-CS pairings would be evidence of MEMC conditioning. Out of the 55 subjects tested so far, both non-conditioned ($n = 53$) and conditioned ($n = 2$) responses have been observed. These findings suggest that a "warned" MEMC may not be present in enough people to justify inclusion for being considered protective.

8:15

5aPP2. Neural mechanisms of selective listening to pitch and timbre in musical sounds. Yang Zhang, Zach Grice-Patil, Phillip Burton, and Cheryl Olman (Univ. of Minnesota, 164 Pillsbury Dr. SE, Minneapolis, MN 55455, zhang470@umn.edu)

Previous research showed that different dimensions of sound such as pitch and timbre elicit distinct patterns of neural activities. Here, we report event related potential (ERP) along with functional magnetic resonance imaging (fMRI) data during a selective listening task of musical sounds. The participants were 12 normal adults. The stimuli were 4 musical tones with timbre (piano vs. violin) and pitch (high vs. low) contrasts. The stimuli were normalized for duration and intensity. In the timbre condition, the listener needed to indicate the instrument category. In the pitch condition, the listener needed to indicate the pitch level. While both conditions reached the ceiling-level accuracy, the pitch condition required significantly longer reaction time. Significant fMRI activations were found in the superior temporal gyri with right dominance for both conditions. The right superior temporal gyrus and left inferior frontal regions were important for coding pitch whereas the anterior cingulate cortex was exclusively active in selectively attending to timbre. These data demonstrate how different cortical networks contribute to selective listening. The ERP data further showed larger N1 and N2 peaks for the timbre condition than the pitch condition, suggesting that attentional modulation operates at both early and later stages of processing for selective listening.

8:30

5aPP3. Neurofeedback training of auditory selective attention enhances speech-in-noise understanding. Inyong Choi, Caroline Emory, Subong Kim (Commun. Sci. and Disord., Univ. of Iowa, 250 Hawkins Dr., Iowa City, IA 52242, inyong-choi@uiowa.edu), and Adam Schwalje (Otolaryngol. - Head & Neck Surgery, Univ. of Iowa Hospitals and Clinics, Iowa City, IA)

Speech-in-noise (SiN) understanding involves multiple cortical processes including feature extraction, grouping, and selective attention. Among those processes, we aimed to investigate the causal relationship between auditory selective attention and SiN performance. Selective attention enhances the strength of cortical neural responses to attended sounds while suppresses the neural responses to ignored sounds, which forms an evidence of sensory gain control theory. We hypothesized that the cortical response-guided neurofeedback could strengthen the sensory gain control, which in turn will improve selective attention performance and may result in better SiN understanding. With a single-blinded, between-subjects design including a placebo group, subjects were asked to attend to one of two simultaneous but asynchronous streams. For the participants assigned to the experimental group, a visual feedback was provided after each trial to demonstrate whether their attention was correctly decoded using their single-trial EEG response. The experimental group participants with four weeks of this neurofeedback training exhibited amplified cortical evoked responses to target speech as well as improved SiN understanding, while the placebo group participants did not show consistent improvement. To our best knowledge, this is the first report of selective-attention training enhancing SiN performance.

8:45

5aPP4. Effect of musical training on EEG measures of spectro-temporal processing. Coral Dirks (SLHS, Univ. of Minnesota, Minneapolis, MN), Anthony Shahin (Ctr. for Mind and Brain, Univ. of California, Davis, CA), Timothy Kwan (Biomedical Eng., Univ. of Malaya, Kuala Lumpur, Malaysia), and Evelyn E. Davies-Venn (SLHS, Univ. of Minnesota, 164 Pillsbury Dr SE, Minneapolis, MN 55445, venn@umn.edu)

Recent studies show that musical training enhances auditory processing abilities such as sensitivity to temporal fine structure and narrowband frequency resolution. Little is known about the effect of musical training on broadband spectro-temporal processing using rippled noise. This study evaluated whether musician enhancement in frequency resolution can be generalized to broadband spectro-temporal resolution using electrophysiological measures. We tested the hypothesis that musicians have enhanced broadband spectral resolution abilities compared to non-musicians. Spectral processing was measured using P1 N1 P2 responses to spectro-temporally modulated rippled noise. The test stimulus was spectro-temporally modulation wideband (4 octaves) rippled noise from 350 to 5600 Hz. EEG recordings were made using a 32-channel standard 10/20 configuration BioSemi system with a sampling rate of 4 kHz. For each condition, 1000 repetitions of the test stimuli were presented at 100% modulation with 2 Hz temporal modulation via electrically shielded ER-3A earphones, using a passive listening paradigm. The amplitude and latency of the P1-N1-P2 components

were measured to determine whether musicians had better sensitivity to spectro-temporal modulation compared to age-matched non-musicians. Results to date show slight musician enhancement in spectral processing. The clinical implications of these findings for auditory-rehabilitation will be discussed.

9:00

5aPP5. Timing effects in auditory-related forward predictions. Leonard Varghese and Barbara Shinn-Cunningham (Dept. of Biomedical Eng., Boston Univ., 610 Commonwealth Ave., Boston, MA 02215, lennyv@bu.edu)

Violation of learned sensory consequences may generate neural "error" signals that inform subsequent behavior. We investigated how this feedback loop might function in human audition, focusing specifically on timing mismatch signals in cortical structures. Participants pressed a button to trigger a sound at an expected, delayed onset. This sound was usually a broadband click ("standard"), but sometimes a low-pass filtered click ("target"). During 100% of trials during the first two experimental blocks, and on >80% of subsequent trials, these sounds occurred ~1 s after a button press. On the remaining <20% of trials, standards were presented slightly early ("early deviants") or slightly late ("late deviants") by 150–200 ms. Listeners were told to respond with a button press to targets; they were not informed about the timing manipulation. We used electroencephalography (EEG) to compare responses evoked by the physically identical expected standards, early deviants, and late deviants. EEG responses evoked by standards and temporal deviants appeared to reflect inter-subject differences in baseline auditory encoding strength, differences in mismatch encoding strength, and/or motor-induced suppression. Subsequent analyses will attempt to disambiguate the relative strength of these contributions, with the goal of determining the extent to which forward sensory predictions in audition reflect expected timing.

9:15

5aPP6. Midbrain sensitivity to frequency "chirps:" Implications for coding voiced speech sounds. Laurel H. Carney, Langchen Fan, and Kenneth S. Henry (Univ. of Rochester, 601 Elmwood Ave., Box 603, Rochester, NY 14642, Laurel.Carney@Rochester.edu)

Many neurons in the mammalian midbrain (inferior colliculus, IC) have strong selectivity for direction of frequency transitions ("chirps"), which often co-exist with the better-known sensitivity for amplitude-modulation (AM) frequency. Chirps occur when harmonics with a phase gradient are summed, as in Schroeder-phase harmonic stimuli. Chirps also occur in voiced speech due to the phase properties of vocal tract resonances. IC neurons are direction-selective for chirps in stimuli with fundamental frequencies in the voice-pitch range, and chirp selectivity can be achieved in an AM-tuned IC model by adding off-CF inhibition. Here, we will show physiological and model responses illustrating how chirps influence IC responses to speech in awake rabbit. In healthy auditory-nerve (AN) responses, the frequency extent, or size, of chirps vary with proximity of AN tuning to formant peaks due to inner ear nonlinearities. For example, synchrony capture causes one harmonic to dominate responses of AN fibers near spectral peaks, and responses that are dominated by one harmonic do not exhibit chirps. The systematic variation in chirp size across frequency channels and strong selectivity of IC neurons for chirps suggest a role for this feature in coding voiced speech. Importantly, this coding mechanism is vulnerable to sensorineural hearing loss.

9:30–9:45 Break

9:45

5aPP7. Improving acquisition time of the frequency-specific auditory brainstem response through simultaneous independently random toneburst sequences. Mark S. Orlando (Dept. of Otolaryngol., Univ. of Rochester, Rochester, NY) and Ross K. Maddox (Departments of Biomedical Eng. and Neurosci., Univ. of Rochester, 601 Elmwood Ave., Box 603, Rm. 5.7425, Rochester, NY 14642, ross.maddox@rochester.edu)

The frequency-specific auditory brainstem response (ABR) is an objective electroencephalographic (EEG) method for estimating auditory thresholds when behavioral thresholds are not attainable, primarily in infants and toddlers. It is recorded by repeatedly presenting brief, band-limited tonebursts and averaging the scalp potential response. In a typical ABR evaluation, measurements are made at multiple intensities, at multiple frequencies, and in both ears, leading to a large combinatorial space and long test times. Here, we present a method of recording the responses at five frequencies in both ears simultaneously, along with preliminary data demonstrating its effectiveness. We created ten toneburst trains—five octave frequencies (500–8000 Hz) by two ears—for simultaneous presentation. Importantly, the toneburst timing for each train was dictated by an independent random process, rather than being periodic. Independent sequences allow separate computation of the ABR to each of the toneburst trains as the cross-correlation of its timing sequence and the EEG recording. Early results suggest substantial improvements in recording time, particularly at lower stimulus intensities. It also appears that cochlear spread of excitation—a problem at high intensities—is mitigated by our method. This improvement in place specificity is predicted by modeling of auditory nerve activity.

10:00

5aPP8. On the relationships between auditory evoked potentials and psychoacoustical performance. Dennis McFadden, Edward G. Pasanen, Mindy M. Maloney (Psych., Univ. of Texas, Austin, 108 E. Dean Keeton A8000, Austin, TX 78712-1043, mcfadden@psy.utexas.edu), Erin M. Leshikar, Michelle H. Pho, and Craig A. Champlin (Dept of Commun. Sci. & Disord., Univ. of Texas, Austin, TX)

Performance was measured on several common psychoacoustical tasks for about 70 subjects. The measures included simultaneous and temporal masking, masking by tones and by complex sounds, critical bandwidth, release from masking, and detection in the quiet. Also measured were auditory evoked potentials (AEPs, short and middle latency). Of interest were the correlations between psychoacoustical performance and the AEP measures, as well as any mean differences in the AEPs by sex and by menstrual cycle. Subjects were tested behaviorally in same-sex crews of 4—8 members; behavioral testing required from 8–10 weeks for each crew. Correlation and effect size were the primary measures of interest. Resampling was used to determine implied significance for the various comparisons studied. (This lab previously reported race differences in psychoacoustical performance and in otoacoustic emissions for these same subjects.) Several AEP measures exhibited large sex differences that were similar across race. Some correlations between psychoacoustical tasks and the various AEP measures were moderately high, but, across sex and race, the individual differences observed in psychoacoustical performance generally were not compellingly related to the individual differences in AEP latency or amplitude. N.F. Viemeister never worked on any of these topics. [Work supported by NIH/NIDCD.]

10:15

5aPP9. Behavioral and electrophysiological evidence of incidental learning, generalization, and retention of speech categories from continuous speech. Yunan C. Wu (Psych., Carnegie Mellon Univ., Psychology-Baker Hall, 5000 Forbes Ave., Pittsburgh, PA 15213, charleswu.01@gmail.com), Ran Liu (MARi, Pittsburgh, PA), Sung-Joo Lim (Dept. of Speech and Hearing Sci., Biomedical Eng., Boston Univ., Boston, MA), and Lori L. Holt (Psych., Carnegie Mellon Univ., Pittsburgh, PA)

Speech learning involves discovering appropriate functional speech units (e.g., speech categories) embedded in a continuous stream of speech. However, speech category learning has been mostly investigated with isolated sound tokens. Here, we used a videogame to encourage incidental learning of speech categories from continuous speech input (Lim *et al.*, 2015). Native English participants (N = 17) played the videogame while listening to acoustically-variable continuous Mandarin sentences. Unbeknownst to participants, four acoustically-variable Mandarin keywords were embedded in the continuous sentences. During training, participants were not informed about the keywords, made no overt categorization decisions, and received no feedback. Participants' post-training categorization test demonstrated robust incidental learning of keyword that persisted even 10 days after training, and generalized to novel utterances and talkers. Further, the N100 response in the frontal EEG site evoked by keyword onsets within

continuous Mandarin speech during passive listening was greater post-training compared to pre-training. This neural enhancement was specific to the Mandarin keywords functionally useful in videogame training. Our results demonstrate that although participants were not informed about the keywords, they did not make overt categorization decisions during videogame play, they incidentally learn functionally-relevant non-native speech categories from continuous speech input across considerable acoustic variability.

10:30

5aPP10. Sequential stream segregation based on spatial cues: Behavioral and neural measurements. Marion David and Andrew J. Oxenham (Psych., Univ. of Minnesota, 75 East River Parkway, Minneapolis, MN 55455, david602@umn.edu)

Differences in simulated spatial cues in the horizontal plane have been shown to enhance both voluntary and obligatory stream segregation of sounds with realistic spectro-temporal variations, such as sequences of syllables. In this experiment, listeners were presented with sequences of speech tokens, each consisting of a fricative consonant and a voiced vowel. The CV tokens were concatenated into interleaved sequences that alternated in simulated spatial positions. The interleaved sequences lasted 1 min. The listeners had to press a button each time they heard a repeated token. In the selective attention task, the listeners were asked to attend only one of the two interleaved sequences; in the global attention task, the listeners had to perceive the interleaved sequences as single stream to detect a repetition between the sequences. Simultaneous EEG measurements were made. The behavioral results confirmed that listeners were able to either attend selectively or globally, depending on the task requirements. The EEG waveforms differed between the two tasks, despite identical physical stimuli, reflecting the difference between global and selective attention. Both behavioral and EEG results reflected the effects of increasing spatial separation in enhancing selective attention and making global attention to the sequences more difficult.

10:45

5aPP11. Behavioral measures of cochlear gain reduction and gain reduction in with normal hearing or minimal cochlear hearing loss. Elizabeth A. Strickland, Hayley Morris, Miranda Skaggs, William Salloom, and Alexis Holt (Speech, Lang., and Hearing Sci., Purdue Univ., 500 Oval Dr., West Lafayette, IN 47907, estrick@purdue.edu)

This is a continuation of a study examining the relationship between cochlear hearing loss and various psychoacoustic measures thought to be related to cochlear function. In the listeners tested, thresholds for long tones ranged from well within the clinically normal range to just outside this range. Where thresholds were elevated, other clinical tests were consistent with a cochlear origin. Because the medial olivocochlear reflex (MOCR) decreases the gain of the cochlear active process in response to sound, when possible, measures were made with short stimuli. Signal frequencies were 2, 4, and 8 kHz. Maximum gain was estimated by measuring the threshold masker level for a masker at the signal frequency and a masker well below the signal frequency. The effect of signal duration on threshold was measured using 10 and 200-signals. One point on the lower leg of the input/output function was measured by finding the threshold masker level for a masker slightly less than one octave below the signal frequency needed to mask a signal at 5 dB SL. Gain reduction was estimated by presenting a pink noise precursor before the signal and masker, and measuring the change in signal threshold as a function of precursor level. The relationship between these measures will be discussed. [Work supported by NIH(-NIDCD)R01 DC008327 (EAS), and grants from the Purdue Office of the Executive Vice President for Research and the Purdue Graduate School (WS).]

Session 5aSC

Speech Communication: Speech Production (Poster Session)

Marc Garellek, Chair
UCSD, 9500 Gilman Drive #0108, La Jolla, CA 92093-0108

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

Contributed Papers

5aSC1. Speech formant changes due to repeated measurements, instruction, and simulated environments using both automated and manual feature extraction. Mark L. Berardi (Dept. of Communicative Sci. and Disord., Michigan State Univ., 1026 Red Cedar Rd., Rm. 211D, Oyer Speech & Hearing Bldg., East Lansing, MI 48824, berardi1@msu.edu), Jessica J. Staples, Sarah H. Ferguson (Commun. Sci. and Disord., Univ. of Utah, Salt Lake City, UT), and Eric J. Hunter (Dept. of Communicative Sci. and Disord., Michigan State Univ., East Lansing, MI)

Speech production can differ depending on how speech is elicited (e.g., spontaneous speech, read text, speaking style instructions, the speaking environment). Previous studies have shown different speaking styles being elicited via instruction (e.g., clear speech) or via the speaking environment (e.g., Lombard speech). There is evidence that the acoustic features of clear speech elicited by reading are similar to those observed in semi-spontaneous interaction between two interlocutors, but that clear speech changes are of a greater magnitude in the read speech than the semi-spontaneous speech. Ten talkers (five male, five female) performed read sentences (BVD) and picture descriptions in several conditions. The present study compares vowel formants from BVD sentences and spontaneously produced picture descriptions in a variety of conditions. Conditions were as follows: (1) repeated measures given the same instructions, (2) the effects of two speaking style instructions (conversational and clear), and (3) four simulated listening environments (quiet, 55 dB SPL of white noise, 63 dB SPL of white noise, and a reverberant environment) presented via earphones. Formants were extracted both manually (hand marked) and using automated techniques. Acoustic features relevant to the speaking styles and simulated conditions will be discussed in terms of the two extraction techniques.

5aSC2. Source-filter interaction: A study using vocal-tract data of a soprano singer. Tokihiko Kaburagi, Momoyo Ando, and Yasufumi Uezu (Kyushu Univ., Shiobaru 4-9-1, Minami-ku, Fukuoka 815-8540, Japan, kabu@design.kyushu-u.ac.jp)

When the fundamental frequency of voice is close to a formant frequency, the voice-source system in the larynx and the acoustic filter of the vocal tract can be coupled acoustically. This source-filter interaction (SFI) can cause unsteadiness in vocal fold oscillations, sudden jump of the fundamental frequency, and transitions in voice register. In this study, a magnetic resonance imaging device was used to scan the vocal tract of a soprano singer in three dimensions when she produced sustained vowels with different notes ranging from A3 (220 Hz) to A5 (880 Hz). The cross-sectional area and the input impedance of the vocal tract were then determined from image data. As a result, the frequency and the magnitude of the first impedance peak for the /i/ vowel increased when the note became higher, suggesting that a strong source-filter interaction may take place for high notes when the fundamental frequency is close to the peak frequency. We therefore conducted a computer simulation of voice generation using models of the vocal folds and the vocal tract. Our preliminary results showed different SFI effects depending on the magnitude of the impedance peak. [This work was supported by JSPS KAKENHI Grant No. JP16K00242.]

5aSC3. Impact of phonatory frequency and intensity on glottal area waveform measurements derived from high-speed videoendoscopy. Rita R. Patel (Dept. of Speech and Hearing Sci., Indiana Univ., 200 South Jordan Ave., Bloomington, IN 47405-7002, patelrir@indiana.edu), Michael Döllinger, and Stefan Kniesburges (University Hospital Erlangen, Med. School, Div. of Phoniatrics and Pediatric Audiol., Univ. Erlangen-Nürnberg, Erlangen, Bavaraia, Germany)

Measurements of glottal area waveform from high-speed videoendoscopy were made on vocally healthy females (n = 41) and males (n = 25) during sustained /i/ production at typical pitch and loudness, high pitch, and soft phonation. Three trials of each condition were performed yielding 594 samples. Statistical analysis of glottal cycle quotients (open quotient (OQ), speed quotient (SQ), rate quotient (RQ), glottal gap index (GGI)), glottal cycle periodicity (amplitude, time (TP)), glottal cycle symmetry (phase asymmetry index, spatial symmetry index, and amplitude symmetry index), glottal area derivative (maximum area declination rate (MADR)), and mechanical stress measures (stiffness index (SI), amplitude-to-length ratio (ALR)) revealed that only SI varied systematically across pitch and loudness conditions for males and females. Variations in pitch and loudness results in changes in SI, ALR, RQ, MADR, and SQ for females, whereas variations in target pitch and loudness results in changes in SI, ALR, RQ, MADR, GGI, OQ, and TP for male speakers. Carefully selecting the laryngeal parameters derived from high-speed videoendoscopy has the potential to provide insights clinically into the known laryngeal biomechanics expected for the increase in pitch and reduction in vocal loudness.

5aSC4. Voice instabilities in a three-dimensional continuum model of phonation. Zhaoyan Zhang (UCLA School of Medicine, 1000 Veteran Ave, 31-24 Rehab Ctr., Los Angeles, CA 90095, zyzhang@ucla.edu)

The goal of this study is to identify vocal fold conditions that produce irregular vocal fold vibration and the underlying physical mechanisms. Using a three-dimensional computational model of phonation, parametric simulations are performed with co-variations in vocal fold geometry, stiffness, and vocal tract shape. For each simulation, the cycle-to-cycle variations in the amplitude and period of the glottal area function are calculated, based on which the voice are classified into three types corresponding to regular, quasi-steady or subharmonic, and chaotic phonation. The results show that the presence of a vocal tract significantly increases the occurrence of irregular vocal fold vibration, which naturally occurs even under symmetric vocal fold conditions, in particular for vocal folds with very low transverse stiffness in the coronal plane or a soft vocal fold body layer. The occurrence of voice instability is suppressed by increasing the transverse stiffness, increasing the body-cover ratio in the longitudinal stiffness, or decreasing the subglottal pressure. The implications of these observations on the production of certain voice qualities are discussed. [Work supported by NIH.]

5aSC5. Direct measurement of glottal flow waveform in an excised K9 larynx with a vocal tract. Alexandra Maddox, Liran Oren, Ephraim Gutmark, Charles P. Farbos de Luzan, and Sid M. Khosla (Univ. of Cincinnati, 3317 Bishop St., Apt. 312, Cincinnati, OH 45219, maddoxat@mail.uc.edu)

False vocal folds (FVF) or ventricular folds have been shown to impact the vibration of the true folds. Using computational and experimental models to examine the aerodynamic and acoustic effects the FVF, previous studies have shown that the presence of FVF lowers the phonation threshold pressure, the overall intraglottal pressure distribution, and enhances intraglottal vortical structures. However, the full effect of the FVF on the glottal flow waveform has never been measured in a tissue model of the larynx. Therefore, the objective of this study was to evaluate the impact of FVF on the glottal flow waveform in an excised canine model. A vocal tract model was placed over the larynx, and direct velocity measurements were taken at the glottal exit using tomographic particle image velocimetry. Measurements were taken with a systematic change of the FVF constriction and integrated to calculate the air flow waveform. The results show that a restriction above the folds increased the overall flow rate (dQ/dt) and change the maximum flow declination rate (MFDR). The 3-D measurements at the glottal exit can also be used to validate previous computational studies on how the vocal tract affects the phonation mechanism. The clinical significance of these findings will be further discussed.

5aSC6. Tongue muscle shortening patterns during speech. Maureen Stone, Natalie Leem, Katie N. Garret (Univ. of Maryland Dental School, 650 W. Baltimore St. Rm. 8207, Baltimore, MD 21201, mstone@umaryland.edu), Jonghye Woo (Massachusetts General Hospital, Boston, MA), and Jerry L. Prince (Johns Hopkins Univ., Baltimore, MD)

This study will used tagged-MRI to examine the 3D shortening patterns of the genioglossus, verticalis, and transversus muscles during speech tasks that move the tongue backwards, forwards, and in complex deformations. Tissue points at each muscle's origin, insertion, and points in between are used to calculate length changes over time. To enhance the range of subject differences, 10 controls and 5 glossectomy subjects will be examined. The motivation is to understand how muscle behavior is linked to internal tongue motion patterns. Activated muscles typically shorten along their line of action. In the tongue, however, muscles interdigitate, so that when one muscle shortens, others are passively pulled out of linear alignment. Thus, muscle fibers may not form a straight line when activated or at rest. In addition, the mechanical consequences of activating a specific muscle can be complex since the tongue moves continuously during speech and different regions of the tongue experience different demands. Moreover, since the tongue stretches in three-dimensions, some muscles may activate to augment a primary muscle, that is, to prevent expansion in the wrong direction. Muscle shortening from tagged MRI does not capture actual muscle activity, but can provide insight into interactive muscle behaviors.

5aSC7. Automatic tongue contour extraction in ultrasound images with convolutional neural networks. Jian Zhu, Will Styler (Dept. of Linguist, Univ. of Michigan, 1052 Island Dr., Apt 104, Ann Arbor, MI 48105, zhujianbw@gmail.com), and Ian C. Calloway (Dept. of Linguist, Univ. of Michigan, Ypsilanti, MI)

Ultrasound imaging of the tongue can provide detailed articulatory information addressing a variety of phonetic questions. However, using this method often requires the time-consuming process of manually labeling tongue contours in noisy images. This study presents a method for the automatic identification and extraction of tongue contours using convolutional neural networks, a machine learning algorithm that has been shown to be highly successful in many biomedical image segmentation tasks. We have adopted the U-net architecture (Ronneberger, Fischer, & Brox 2015, U-Net: Convolutional Networks for Biomedical Image Segmentation, DOI:10.1007/978-3-319-24574-4_{28}), which learns from human-annotated splines using multiple, repeated convolution and max-pooling layers in the network for feature extraction, as well as deconvolution layers for generating spatially precise predictions of the tongue contours. Trained using a preliminary dataset of 8881 human-labeled tongue images from three speakers, our model generates discrete tongue splines comparable to those identified by human annotators (Dice Similarity Coefficient = 0.71). Although work is ongoing, this neural network based method shows considerable promise for the post-processing of ultrasound images in phonetic research. [Work supported by NSF grant BCS-1348150 to P.S. Beddor and A.W. Coetzee.]

5aSC8. A method for distinguishing tongue surface topology for different categories of speech sound. Jonathan N. Washington (Linguist. Dept., Swarthmore College, 500 College Ave., Swarthmore, PA 19081, jwashin1@swarthmore.edu) and Paul A. Washington (Dept. of Petroleum Eng. and Geology, Marietta College, Marietta, OH)

An analytic method has been developed to assist in interpreting data from two-dimensional ultrasound tongue imaging by determining whether the position of the tongue surface significantly differs for two categories of speech sound and, if so, which region is most associated with the difference. Individual tongue traces are modeled as curves in a polar coordinate system around a reference point. Two categories of speech sound are compared as a function of the distance between the traces of the two categories at each arc angle. The regions of the arc with the most significant deviation (i.e., where the categories are best contrasted) are shown by the largest ratio of mean to standard deviation (the greatest number of standard deviations separating the mean tongue surfaces). Categories are most distinguished in regions with more than one standard deviation separating them. A ratio of the angular distance between a point of complete overlap (no contrast) and the maximum positive and negative differentiation points has potential as a speaker-agnostic measure of which regions of tongue position are important at the level of the linguistic variety. To demonstrate this method, its application to the vowel anteriority contrast in several Turkic languages is presented.

5aSC9. Palate shape and articulatory range. Keith Johnson (Linguist, UC Berkeley, 1203 Dwinelle Hall, Berkeley, CA 94720, keithjohnson@berkeley.edu)

Using the Wisconsin X-ray microbeam corpus, this paper explores the relationship between palate shape in the saggital plane and articulatory variation. The corpus has been updated with tags (Praat Textgrids) marking the acoustic onsets and offsets of words and phones. In addition, the sound files have been translated to WAV format, and the articulatory data saved as plain text files. The shape of the palate for each of 57 speakers of American English was measured in terms of the total 2D area of the palate vault above the occlusal plane. Articulatory variability was measured in terms of the interquartile ranges of the x and y locations of 7 pellets attached to the tongue (4 pellets), lips (upper and lower), and jaw. Although there is a significant correlation between palate area and the range of motion of some articulators (particularly on the body of the tongue, but surprisingly to this author, not the motion of the jaw), the study also finds that there is a great deal of variation among speakers. Articulatory motions involved in particular speech sounds will also be examined to determine if there are specific patterns of variation associated with the palate size.

5aSC10. Acoustic correlates of prominence in Yawarana. Natalia Cáceres Arandia, Alyssa Moore, Zac Post, Spike Gildea, and Melissa M. Baese-Berk (Dept. of Linguist, Univ. of Oregon, 1290 University of Oregon, Eugene, OR 97403, mbaesebe@uoregon.edu)

Yawarana (Cariban) is a critically endangered language of Venezuela, with 20–30 speakers and little published research beyond wordlists. In these wordlists, orthographic indications of stress and/or vowel length are inconsistent. As part of our language documentation project, we examine acoustic correlates of prominence in Yawarana. Many other languages in the Cariban family have a clear rhythmic stress system, with vowel lengthening and pitch excursion marking prominence; Yawarana does not mark prominence in the same way. Even so, in some situations, native speakers do appear to attend to stress (i.e., when correcting the pronunciation of language learners). Here, we ask what the acoustic correlates of prominence are, paying particular attention to intensity, pitch, and duration information. We are examining over 200 lexical items produced by four native speakers. These items are repeated in isolation, produced in carrier phrases, and produced in narratives and conversations. We compare measures of prominence in each of these situations. This investigation will pave the way for a more in-depth study of acoustic features of Yawarana and will inform future studies of prominence in other Cariban languages.

5aSC11. Acoustic measures of delabialized velars in Hong Kong Cantonese. John Culnan (Dept. of Linguist, Univ. of Arizona, Tucson, AZ 85721, jmculnan@email.arizona.edu) and Suki Yiu (Linguist, Univ. of Hong Kong, Hong Kong, Hong Kong)

Hong Kong Cantonese is undergoing a neutralization of the labialized velar /k^w/ with the plain velar /k/ before the round back vowel /ɔ/ that is well-documented auditorily but not acoustically (Bauer, 1982; Newman, 1987; To *et al*., 2015). The present data from 14 native speakers of Hong Kong Cantonese seeks to determine what acoustic characteristics differentiate non-neutralized /k/ and /k^w/, and whether neutralization of these segments in neutralization environment is complete or incomplete. Initial data analysis suggests that locus equations on the second and third vowel formants may not be reliable cues in distinguishing plain and clearly labialized (non-merged) velars in this dialect of Cantonese, although formant lowering and increased vowel length for non-merged, labialized productions appear consistent, and intensity measures at 50 ms post-vowel onset may prove to be an important cue as well. The most consistent cues for labialization in clearly labialized articulations of target words will then be measured in neutralized (phonetically plain) productions in order to determine whether any acoustic traces of labialization remain for these realizations.

5aSC12. The position of the tongue root in the articulation of posterior sibilants in Polish. Malgorzata E. Cavar (Dept. of Linguist, Indiana Univ., Ballantine Hall 844, 1020 E. Kirkwood Ave., Bloomington, IN 47405-7005, mcavar@indiana.edu) and Steven M. Lulich (Speech and Hearing, Indiana Univ., Bloomington, IN)

Standard Polish has a typologically relatively rare contrast between "hard" post-alveolar affricates and fricatives, transcribed by Ladefoged and Disner (2012) as (/tṣ, dẓ, ṣ, ẓ/ and "soft" alveolo-palatal affricates and fricatives (IPA transcription /tɕ, dʑ, ɕ, ʑ /). The hard post-alveolars are notoriously ambiguous—phonetically they are neither sense stricto retroflexes (though some authors adopt such an analysis based on phonological arguments, e.g., Hamann 2003) nor typical palatoalveolars. Additionally, the "hard" post-alveoalars can be allophonically palatalized in the context of [i]. Multiple approaches have been proposed to differentiate the hard and soft series, all of them primarily focusing on the shape of the body of the tongue, including the level of raising of the body of the tongue, the place of articulation, and the length of the constriction. In this study, we present 3-D tongue shapes of the consonants. The 3D ultrasound images—besides additional details of the shape of the tongue body—reveal a consistent difference in terms of the tongue root position, with a fronting of the tongue root and a pronounced groove along the center of the tongue in the root part of the tongue for prepalatals and no such fronting for the "hard" post-alveolars.

5aSC13. Creaky phonation and Mandarin 214-Sandhi. Noah Elkins (Linguist, Macalester College, 1731 Dayton Ave., Apartment 4, St. Paul, MN 55105, nelkins@macalester.edu)

Mandarin has four contrastive tones (55, 35, 214, and 51). The 214 tone is consistently produced at the lowest part of a speaker's pitch range, and is therefore frequently accompanied by allophonic creaky phonation. Additionally, Mandarin has a complex system of tone sandhi; the type of sandhi being investigated in this paper is whereby, when two adjacent 214 tones are uttered, the first 214 changes to a 35, and the second 214 changes to a 21. Twenty speakers of Mandarin were recorded at the campus of Peking University using a sentence list constructed to elicit minimal pairs between sandhi-unaffected and sandhi-affected 214 tones. Age and gender were not considered a contributing factor to the presence of 214-linked creak, and so such factors were not controlled for. Preliminary results show that there is no significant difference between the amount of creak on affected and unaffected vowels. Therefore, the duration spent uttering the vowel at a lower pitch did not increase the intensity of creak. However, the continued presence of creak on a sandhi-affected 214 vowel may serve as a perceptual cue to that tone, whereby a Mandarin speaker can understand the meaning of the syllable even without a robust pitch contour.

5aSC14. The effects of prosody on pitch and voice quality of White Hmong tones. Marc Garellek (Linguist, UCSD, 9500 Gilman Dr. #0108, La Jolla, CA 92093-0108, mgarellek@ucsd.edu) and Christina M. Esposito (Linguist, Macalester College, St. Paul, MN)

White Hmong contrasts two high-falling tones (one breathy, the other modal) and two low tones (one modal level-tone, the other creaky low-falling). Perceptual studies [Garellek *et al*. (2013)] have shown that listeners rely on breathy voice to distinguish between the high-falling tones, but ignore creaky voice when distinguishing between the low tones. We test whether such differences stem from prosodic variation, by examining tokens from stories read by native speakers. Vowels were annotated for phrasal position and neighbouring tones. We obtained f0 and voice quality measures. Results support and help elucidate previous perceptual research: (1) the breathy high-falling tone is breathy in all prosodic positions, (2) the breathy high-falling tone has a prosodically-variable f0, (3) the creaky low-falling tone has a prosodically-stable f0. The creaky low-falling tone is creakier than the modal low tone in all positions. Therefore, listeners may ignore f0 in the identification of the breathy high-falling tone because its pitch is prosodically variable. However, the creaky low-falling tone is consistently low-falling, which could explain why listeners rely on f0 as the dominant cue. Discussion will include why listeners ignore creaky voice if it is robust in production.

5aSC15. Acoustic phonetic measurements of grammatical tones in Anyi. Ettien N. Koffi (English, Saint Cloud State Univ., 720 Fourth Ave. South, Saint Cloud, MN 56301, enkoffi@stcloudstate.edu)

Talkers and hearers of Anyi, a West African language of the Akan family, are very adept at discriminating between sentences containing verbs conjugated in the indicative, the intentional, and the subjunctive moods just by relying on subtle variations in pitch. A phonetic investigation is undertaken to determine as precisely as possible the acoustic cues that contribute the most to intelligibility. Ten speakers produced five sentences each in all three moods for a total of 150 utterances. The tone bearing units (TBU) on the subject pronoun and the disyllabic verbs are analyzed acoustically. The acoustic correlates investigated are F0, intensity, and duration. All in all, 810 TBUS are measured. The main findings are as follows. The F0 of the subject pronoun and of the verb are the most robust cues for discriminating between the declarative and the intentional moods. The differentiation between the intentional and the subjunctive moods rests principally on the F0 of the subject pronoun. The distinction between the indicative and the subjunctive moods depends mainly on the F0 of the verb and to a lesser extent on duration. Intensity does not seem to be a robust cue for discriminating between these three tone-induced grammatical constructions.

5aSC16. Timing patterns of voiceless and voiced singleton and geminate plosives of Yanagawa Japanese. Shigeko Shinohara (Lab. of Phonet. and Phonology, CNRS/Univ. of Sorbonne-Nouvelle, Paris, France), Qandeel Hussain (Dept. of English (Linguist Program), North Carolina State Univ., 221 Tompkins Hall, Campus Box 8105, Raleigh, NC 27695-0001, qhussai@ncsu.edu), and Masako Fujimoto (Adv. Res. Ctr. for Human Sci., Waseda Univ., Tokorozawa, Saitama, Japan)

The paper examines acoustic timing patterns of word-medial voiceless and voiced singleton and geminate plosives of Yanagawa Japanese, one of the Chikugo varieties of Japanese spoken in the center of Kyushu (Japan). Unlike standard Tokyo Japanese, Yanagawa Japanese is characterized by frequent gemination of any types of consonants including voiced obstruents (e.g., /kuzzoko/ [kuddzoko] "sole, the fish," /mirogge/ [miɾogge] "a shell fish"). Five Yanagawa Japanese speakers were recorded. Another five speakers of standard Tokyo Japanese were also recorded as a control group. The stimuli consisted of nonsense words with $C_1V_1C(C)_2V_2$ structure (e.g., /kak(k)a/, /kag(g)a/). The findings suggest that the whole word duration containing geminate consonants was longer in proportion to mora count difference those containing singleton consonants, confirming moraic timing at word level in Yanagawa Japanese. When C_2 was a geminate (voiceless or voiced), it lengthened the preceding vowel (V_1) and shortened the following one (V_2) in both dialects. However, the magnitude of influence of geminates to adjacent vowels was dialect specific: V_1 duration in Yanagawa was consistently longer than Tokyo when the following consonant was a geminate.

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The results of the current study point towards the timing differences in singletons and geminates across Japanese dialects.

5aSC17. Phonetic realization of vowel reduction in Brazilian Portuguese. Sejin Oh (Linguist, The Graduate Ctr., CUNY, 365 5th Ave., New York, NY 10016, soh@gradcenter.cuny.edu)

This study explored the phonetic realization of vowel reduction in Brazilian Portuguese, which reportedly exhibits raising of non-high vowels in unstressed syllables. Specifically, we tested the influence of speech rate on the realization of /a/ in five prosodic positions: word-initial pretonic, medial pretonic, tonic, medial posttonic, and final posttonic. The results showed that, while both speech rate and prosodic position had a clear effect on the phonetic duration of vowels, F1 values were far better predicted by the vowel's prosodic position (non-posttonic vs. posttonic), although some effects of speech on F1 were observed in vowels at word edges. Correlations between phonetic duration and F1 were statistically significant but generally weak in all positions. We argue these findings suggest that vowel reduction in Brazilian Portuguese primarily reflects phonological patterning rather than phonetic undershoot, although phonetic reduction is also apparent. We discuss the results in the context of cross-linguistic studies of vowel reduction, and the relation between phonetics and phonology.

5aSC18. Phonetic variability of nasals and voiced stops in Japanese. Yoichi Mukai and Benjamin V. Tucker (Dept. of Linguist, Univ. of AB, 3-26 Assiniboia Hall, Edmonton, AB T6G2E7, Canada, mukai@ualberta.ca)

We investigated the phonetic variability of nasals and voiced stops in a large-scale Japanese speech corpus. We then analyzed the types of variation across speech styles. In particular, we examined the instances where target segments are deleted or realized as different phonemes. We identified 285 lexical entries displaying variability of target segments in the Corpus of Spontaneous Japanese (Maekawa, 2003). In line with the findings of Arai (1999), we observed a few variants of voiced stops, such as /d/ becoming [n] in /doɴna/ "what" and /g/ being deleted in /daigaku/ "university." We also found /b/ turning into [m], such as /boku wa/ "I am" becoming [mo kaː]. For nasals, we observed /zeɴiɴ/ "all the people" and /geɴiɴ/ "reason" becoming [zẽːɪɴ] and [gẽːɪɴ], in which the first /ɴ/ was deleted and the preceding /e/ was nasalized and lengthened. Our findings suggest that the extent to which speakers produce phonetic variants of target segments is likely specified lexically more than stylistically because we find instances of [zẽːɪɴ] and [gẽːɪɴ] produced across multiple speech styles. Further, we find more than 90% of the instances of /geɴiɴ/ were pronounced as [gẽːɪɴ] and 80% of /zeɴiɴ/ were pronounced as [zẽːɪɴ].

5aSC19. Stress-sensitive consonant gemination through plural noun reduplication in Tohono O'odham. Daejin Kim and Robert Cruz (Linguist, Univ. of New Mexico, 1 University of New Mexico, MSC03 2130, Albuquerque, NM 87131-0001, daejinkim@unm.edu)

This study examines how plural noun reduplication in Tohono O'odham (TO) language is phonetically realized. To make a noun plural, the first CV sequence (i.e., base; e.g., *goks* "a dog" → *go**go**ks* "dogs") or the first consonant of the base (e.g., *pado* "a pig" → *pa**p**do* "pigs") are reduplicated after the base (Hill & Zepeda, 1998). Base and reduplicant have been regarded phonologically equivalent (i.e., consonant gemination; Fitzgerald, 2003). In a sense that TO has the strong-weak stress pattern across syllables (e.g., *tówa* "turkey"), stress may influence reduplicants' duration. The CV reduplicant at the second syllable with the weak stress may not be phonologically same as the base and the C reduplicant at the coda of the base with the strong stress. If CV and C reduplicants are equivalent even with stress, an additional prosodic unit (i.e., mora) may exist. The analysis of TO speech supports that the base (CV_1) is longer than the reduplicant (CV_2 or C_2) even with other contextual variables. Reduplication process may be stress-timed, but it is unknown whether speakers equally treat base and reduplicants. This paper will also discuss a need to examine how TO speakers control the timing with rhythm between base and reduplicant.

5aSC20. Place of articulation effects on voice onset time and phonation bleed persist in languages with no voicing distinction. Stephanie Kakadelis (Linguist, The Graduate Ctr., CUNY, 365 Fifth Ave., New York, NY 10016, skakadelis@gradcenter.cuny.edu) and D. H. Whalen (Linguist, The Graduate Ctr., CUNY, New Haven, CT)

Place of articulation effects on voice onset time (VOT) and phonation bleed in the closure of oral stops has been observed in languages which utilize these as perceptual correlates to voicing distinctions (Lisker & Abramson, 1967; Ohala & Riordan, 1979). In this study, measurements from intervocalic oral stops were collected from three No Voicing Distinction (NVD) languages, Bardi (ISO-639 bcj), Arapaho (ISO-639 arp), and North Puebla Nahuatl (ISO-639 ncj). Tokens surfacing with positive VOT increased in duration as place of articulation went from more anterior to dorsal. VOT for Arapaho coronal oral stops was longer on average than velar orals stops. Nahuatl showed a similar pattern, with average VOT increasing from labial to coronal to velar. Average VOT for Arapaho labial tokens and all Bardi tokens was negative, so they were not included in VOT measurements. Conversely, average duration of phonation bleed in all three languages was greatest for labial oral stops, decreasing in average duration for coronals and velars. This suggests a physiological effect of vocal tract volume on both VOT and maintenance of phonation during the closure of an oral stop even in the absence of a phonological distinction.

5aSC21. Inter- and intra-speaker variability in the production of voiceless nasal consonants. Maureen Hoffmann (Dept of Linguist, Univ. of Arizona, P.O. Box 210025, Tucson, AZ 85721, mhoffm@email.arizona.edu)

This study uses both acoustic and aerodynamic data to investigate variation in the realization of phonemically voiceless nasal consonants, which have been found to differ cross-linguistically. In particular, it examines the timing of voicing and oral and nasal air flow during production of voiceless nasal consonants in both word-initial and word-medial positions in Hakha Chin, a Tibeto-Burman language spoken in western Burma/Myanmar. Comparing three other Tibeto-Burman languages, Bhaskararao and Ladefoged (1991) found two distinct patterns in the production of voiceless nasal consonants, which varied by language, but reported general consistency among speakers of the same language. Results of the present study show considerable variation in the realization of voiceless nasals, both between individual speakers and even between individual tokens produced by the same speaker. This high degree of variability may be linked to the relatively weak acoustic cues for place of articulation among nasal consonants in general and for voiceless nasals in particular.

5aSC22. The effect of lexical competition on vowel duration before voiced and voiceless English stops. Eleanor Glewwe (Univ. of California, Los Angeles, 3125 Campbell Hall, Los Angeles, CA 90095-1543, eleanorglewwe@gmail.com)

One source of phonetic variation is lexical competition, which has been shown to cause both hyperarticulation, especially of vowel formants (e.g., Wright 2004), and reduction, manifested as shorter segment and word durations (Kilanski 2009, Gahl *et al.* 2012). Studies have found that lexical competition (e.g., having a minimal pair competitor, having more phonological neighbors) causes contrastive hyperarticulation of the initial stop voicing contrast in English: greater competition makes VOT longer for voiceless stops and/or shorter for voiced stops (Baese-Berk & Goldrick 2009, Nelson & Wedel 2017). I conducted a corpus study that looked for contrastive hyperarticulation of the final stop voicing contrast in English. The cue examined was preceding vowel duration. I did not find contrastive hyperarticulation: neither minimal pair competitor existence nor higher neighborhood density caused vowels to be longer before final [d] or shorter before final [t]. Instead, both competition metrics correlated with shorter vowel durations before [t] and [d]. This result is consistent with Goldrick *et al.* (2013)'s failure to find contrastive hyperarticulation of the final stop voicing contrast and their hypothesis that competition affects initial and final contrasts differently. It is also consistent with Gahl *et al*'s (2012) finding that high neighborhood density causes reduction.

5aSC23. The effect of domain-initial strengthening on voice onset time and vowel length on English oronyms. Marisha D. Evans (Linguist, Macalester College, 948 North St., Ste. 5, Boulder, CO 80304, mevans3@macalester.edu)

Oronyms are sets of words and phrases that contain the same phonemes but differ in word boundary (*atop*, *a top*). Past research established that domain-initial strengthening (DIS) affects segments at the beginning of prosodic domains. This research examines voice onset time and vowel length in English oronyms with the hope of finding a systematic strengthening of the target sounds word-initially. The word list consists of oronym pairs with the same phonemes and the same [vowel, voiceless stop, vowel] at the beginning of the oronym (e.g. *atop*/*a top*, *attack*/*a tack*). To elicit more natural-sounding tokens, participants created sentences with the target oronym at the beginning, or read already-prepared sentences. It is hypothesized that there will be a systematic strengthening of the initial segments (in the form of either a longer VOT or vowel length). This could mean speakers are fully aware of word boundaries and use DIS to help avoid lexical ambiguity when speaking.

5aSC24. The effect of cognitive load on the vowel onsets of older and younger adults. Richard J. Morris, Megan K. MacPherson, and Maria Roa (Commun. Sci. and Disord., Florida State Univ., 201 West Bloxham Rd., 612 Warren Bldg., Tallahassee, FL 32306-1200, richard.morris@cci.fsu.edu)

The purpose of the study was to determine the effect of cognitive load on the vowel timing of older and younger adults. It was hypothesized that the participants would have longer and less extensive second formant (F2) transitions in the heavier cognitive load condition. It was also hypothesized that the older adults would have longer and less extensive F2 transitions than the younger adults. Eight adults, four younger and four older, equally matched for sex participated. They completed a sentence-level Stroop task in two conditions, congruent and incongruent, that varied on cognitive load. Eight sentences were produced in each condition. Plosive-vowel syllables were selected for measurement before the color words, at the color words, and after the color words. All participants exhibited longer and more extensive vowel transitions before and at the color word in comparison to afterwards. In the incongruent condition, the color words had longer F2 transition durations. The older adults had longer F2 transition durations than the younger adults. The F2 transition frequency extent was greatest at the color word. The older adults had wider F2 transitions at the color word that also were wider for the incongruent words.

5aSC25. Imitation of prosodic contours in word shadowing. Cynthia G. Clopper (Ohio State Univ., 1712 Neil Ave., Oxley Hall 100, Columbus, OH 43210, clopper.1@osu.edu)

In a word shadowing task, participants are asked to repeat auditorily-presented words aloud, providing a speech "shadow" to the stimulus materials. Previous research has demonstrated that even in this kind of non-interactive, non-social task, participants' speech is more similar to the stimulus materials during shadowing than during a baseline reading task. These results are taken as evidence for implicit phonetic imitation during word shadowing. The current study examined imitation of prosodic contours in a word shadowing task. Participants were asked to first read a set of words aloud to establish their baseline production and then to repeat the same set of words after hearing them produced by a young female native speaker of Midwestern American English. This model talkers' word productions all involved a non-falling intonation contour (i.e., either a plateau or a rise), consistent with "list intonation" in reading. Although the participants produced more than 50% non-falling contours in their baseline productions, they produced significantly more non-falling contours in the shadowing task, suggesting imitation of the prosodic contours produced by the model talker. The results of an acoustic analysis of the prosodic contours to examine the phonetic detail of the imitation will also be discussed.

5aSC26. Individual strategies in adaptation to altered auditory feedback. Sarah Bakst (Commun. Sci. and Disord., Univ. of Wisconsin - Madison, 1500 Highland Ave., Madison, WI 53705, sbakst@wisc.edu), John F. Houde (Otolaryngol. - Head and Neck Surgery, UCSF, San Francisco, CA), Susan Lin, and Keith Johnson (Linguist, Univ. of California Berkeley, Berkeley, CA)

Speakers monitor themselves while talking. When they hear a real-time altered version of their speech, they will change their articulation so that when they hear their altered speech, it matches their acoustic target [Houde and Jordan (1998, Science 20;279(5354):1213–1216)]. The experiment presented here used the novel addition of ultrasound imaging to reveal how speakers (n = 30) change their articulations in response to two different formant perturbations: raising of F1 in "head" and F2 in "hood." Principal components analysis was used to identify speakers' individual strategies during adaptation. Some speakers use a single strategy for an entire adaptation block, while others change strategy. Speakers are also known to change production in a formant that was not altered [Katseff *et al.* (2010, JASA **127**(3), 1955)]. The ultrasound analysis shows that at least for some speakers, change in two formants is linked to independent and uncorrelated articulatory components and possibly serves a perceptual purpose in compensation, rather than being an unintended result of the compensatory response. Preliminary results (n = 4) of adaptation to raising F3 in "heard" will also be presented. Modeling with the Maeda and Manzara synthesizers [Bakst and Johnson (2016, JASA **140**(4), 3223)] correctly predicted speakers' articulatory strategies.

5aSC27. Acoustic cues to linguistic profiling? Machine learning of phonetic features of African American English. Meisam K. Arjmandi, Laura Dilley, and Suzanne Wagner (Dept. of Communicative Sci., Michigan State Univ., East Lansing, MI 48824, ldilley@msu.edu)

Listeners can readily differentiate words spoken in an African American English (AAE) dialect from a Standard American English (SAE) dialect, even in absence of distinctive morphosyntactic features. However, it is still unclear what acoustic-phonetic cues listeners utilize to rapidly distinguish AAE from SAE. This study investigates the informativeness of various acoustic-phonetic cues to the characterization of AAE dialect. V and VC sequences (with C = /n/, /m/, /l/, /r/) from speech of 7 female speakers (4 SAE and 3 AAE), recorded during sociolinguistic interviews, were randomly selected and acoustically analyzed, controlling for coarticulatory context. Acoustic cues of F1, F2, F3, F4 formant trajectories, formant bandwidth, pitch variation, duration, intensity, and voice quality measures (e.g., harmonic-to-noise ratio, jitter, shimmer, and spectral slope) were measured in these segments to identify the extent of their contribution to separating AAE and SAE. The results from machine learning modeling of acoustic cues demonstrate that speech within an AAE dialect entails distinct acoustic characteristics and voice quality compared to SAE speech. These separate acoustic patterns between AAE and SAE dialect indicate the need for including dialect-specific acoustic cues both in automatic speech recognition applications and clinical assessments of speech-language disorders.

5aSC28. Modeling geographic variation in pronunciation of United Kingdom English. Katherine Henley (Linguist, Univ. of Georgia, 142 Gilbert Hall, Athens, GA 30602, keh71529@uga.edu) and John S. Coleman (Linguist, Philology and Phonet., Univ. of Oxford, Oxford, United Kingdom)

This paper details an investigation of geographic variation of UK English pronunciation based on formant trajectories of the diphthongs /aɪ/ and /aʊ/. Thousands of audio samples of these vowels were retrieved from the spoken portion of the British National Corpus (BNC), and their F1 and F2 values were compared across 89 locations within the UK. The study was designed with an expectation of a spectrum of pronunciations existing across the UK with monophthongal realizations primarily found in the Northern and Western areas of the country and more diphthongization occurring in the South. After searching the BNC for phonemically aligned tokens of "five" and "house," functional data, modal, and geographic analyses were conducted. We found a strong correlation between speaker distance from the dataset's northernmost location and diphthongization of /aɪ/. This finding suggests that the variation in pronunciation of the vowel in "five" is more continuous than isoglossic. By investigating formant trajectories of vowels involved in the English Great Vowel Shift, we have captured variation in pronunciation that exists across the UK as a result of English sound change.

5aSC29. Shifting from the shift: Loss of dialect distinction in the US. Monica Nesbitt (Linguist and Lang., Michigan State Univ., Wells Hall - MSU, 619 Red Cedar Rd., East Lansing, MI 48824, nesbit17@msu.edu)

Historically, /æ/ in the Northern Cities Shift (NCS) dialect area is realized as raised, lengthened, and diphthongized in all consonantal contexts (Boberg and Strassel 2000) and that pre-oral /æ/ is realized with as much nasalization as pre-nasal /æ/ (Plichta 2004). Recent studies suggest that young speakers in the dialect area are adopting a nasal pattern for /æ/ such that only tokens before nasal consonants are raised and fronted in phonetic space (in Lansing, MI (Wagner *et al*. 2016), in Syracuse, NY (Driscoll and Lape 2015), in Upstate NY (Thiel and Dinkin 2017), in Detroit, MI (Acton *et al*. 2017)). Through acoustic analysis of 1310 /æ/ tokens produced by 26 speakers born and raised in Lansing, MI (date of birth 1991–1997), the current study finds that younger speakers are not simply lowering and retracting this vowel, they are rejecting all phonetic components of the NCS /æ/ system. For these speakers, only pre-nasal /æ/ tokens are realized as long, diphthongal, and nasalized, while pre-oral tokens are short, monophthongal, and have little to no nasalization. Vowel quality for /æ/ in the NCS dialect area is thus undistinguishable from that in the western and midland states in the US (e.g., California and Kansas).

5aSC30. Using speech stereotypes to assert identity and affinity with a minority group. Auburn Lutzross (Linguist, Univ. of California Berkeley, 2435 Grant St., Apt. 2, Berkeley, CA 94703, alutzross@berkeley.edu)

Perception-based studies on female speech have demonstrated clear stereotypes associating acoustic qualities and sexual orientation (e.g., Camp 2009, Munson *et al*. 2005, Pierrehumbert *et al*. 2004). However, production studies on female speech are rare and inconclusive: though male speakers have been shown to vary their use of stereotypically gay speech according to context (Podesva 2011), female speakers have not been studied. Likewise, there has been little consideration of how other personal attributes can intersect with sexual orientation to affect use of stereotypical speech. This study addresses the following for female speakers: (1) how personal attributes interact with sexual orientation in phonetic variation, (2) why straight or bisexual speakers might take on features of stereotypically lesbian speech, and (3) how these personal factors affect speech in different contexts and the resulting inter-speaker and inter-context variation. Speakers were recorded in both reading and interview speech modes. ANOVA of phonetic variation revealed that for straight and bisexual speakers, their "familiarity with Queer culture" was the most influential attribute on both speech modes. However, there was no such effect for lesbian speakers. I argue that speakers used stereotypical lesbian speech patterns to express outgroup affinity.

5aSC31. A sociophonetic analysis of gay male speech stereotypes in Buenos Aires, Argentina. Ellis Davenport (Linguist, Macalester College, 1600 Grand Ave., St. Paul, MN 55105, edavenpo@macalester.edu)

The present study expands academic knowledge on the phonetic cues that index "gay speech" beyond the English language by examining speech produced by gay Buenos Aires Spanish (porteño) speakers. Research has been conducted on gay speech in other parts of Latin America or on the Spanish language in general (Pahis 2017, Ezquerra 2015, Mack 2011, Mendes 2007, Sívori 2005), but not on the porteño dialect. Acoustic features (vowel quality, pitch, speech rate, and degree of aspiration of pre-consonantal and syllable-final /s/) corresponding to public stereotypes on porteño gay speech were measured. Speech from gay and straight men, as well as actors interpreting gay & straight roles in film and television, was analyzed. It is hypothesized that gay men will have higher pitch and that vowels be more clearly articulated than straight counterparts, that pre-consonantal and syllable-final /s/ will be aspirated less frequently and that pitch contours will be more pronounced in the speech of gay men. In addition, the speech of actors playing gay roles will align more closely with the stereotypes of gay speech than with actual phonetic properties, in accordance with Cartei and Reby 2011.

5aSC32. Understanding the speech cues to bisexuals. Mariya Yoshovska (Linguist, Macalester College, 1600 Grand Ave., St. Paul, MN 55105, myoshovs@macalester.edu)

The vast majority of research on speech patterns in the queer community has focused on gay and lesbian speakers. This study aims to expand our knowledge of speech in the queer community by focusing on the speech cues of adult bisexual speakers. For this study, adult lesbian, gay, bisexual, and straight cisgender American English speakers recorded the "Fire passage," the "Rainbow passage," and a description of their hometown. Vocal characteristics, such as duration, pitch, voice quality, formants, and speech rate, were measured to observe the differences between speech patterns of adult bisexual speakers and other queer and straight identities. The results suggest that bisexual and lesbian female speakers have naturally lower average pitch while straight speakers go up in pitch. Bisexual speakers maintain their pitch range between natural and read speech, while lesbian speakers have a wider pitch range during natural speech and straight speakers during read speech. The speech rate of bisexual and straight speakers decreases in natural speech, while it increases for lesbian speakers. Sentence-final phonation results were not conclusive between speaker identities. It is hypothesized that bisexual speakers produce less fronted /u/ and /ɑ/, higher /ɛ/ (low F1) and a more back /o/ (low F2).

5aSC33. Growth in the accuracy of preschool children's /r/ production: Evidence from a Longitudinal Study. Mara Logerquist (Speech-Language-Hearing Sci., Univ. of Minnesota, Minneapolis, MN), Hyuna A. Kim (Commun. Sci. and Disord., Univ. of Wisconsin, Madison, WI), Alisha B. Martell, Benjamin Munson (Speech-Language-Hearing Sci., Univ. of Minnesota, 115 Shevlin Hall, Minneapolis, MN, munso005@umn.edu), and Jan Edwards (Hearing and Speech Sci., Univ. of Maryland, College Park, MD)

The acquisition of American English /r/ is of particular theoretical and clinical interest because it is typologically rare among the world's languages, because it is one of the latest-acquired sounds by typically developing children, and because it is one of the sounds that is most likely produced in error by children with residual speech sound disorders. In this study, we examined the accuracy productions of /r/ and its most frequent substitute, /w/, by preschool children (n = 120) at two time-points: when they were between 3;3 (years;months) and 4;4 years old and again when they were 4;4 to 5;4 years. Accuracy was determined both by trained phonetic transcriber. Moreover, the accuracy of productions from the first time-point was rated by groups of naive listeners using a perceptual rating experiment similar to that described by Schellinger *et al*. (2016). Considerable variation in /r/ production was found at both time-points, and there was sizeable variation in the extent to which production improved from the first to the second time-point. Surprisingly, none of the predictor measures at the first time-point (speech perception, expressive and receptive vocabulary size) predicted growth in /r/ in this group.

5aSC34. Towards an inventory of prosodic contours produced by English-acquiring 2-year-olds: The use of H* and H + !H*. Jill C. Thorson (Commun. Sci. and Disord., Univ. of New Hampshire, 4 Library Way, Hewitt Hall, Dover, NH 03824, jill.thorson@unh.edu) and Stefanie Shattuck-Hufnagel (Speech Commun. Group, RLE, MIT, Cambridge, MA)

One viewpoint of early prosody is that utterances are highly simplified in comparison to adult models, in part, due to biological constraints such as incomplete control of pitch production (Lieberman, 1967; Snow, 2006). In contrast, research in Catalan and Spanish shows that early utterances (under 2 years) consist of the basic intonational categories (Prieto *et al*., 2012), including contours that would not be predicted under a biological approach. Our primary question asks what is the inventory of contours produced by American English-acquiring toddlers. Based on work showing toddlers to have near adult-like acoustic realizations of certain pitch accents (Thorson, 2015), we hypothesize that children will demonstrate sophisticated intonational configurations; here we analyze specific pitch accent varieties in

toddler speech. Spontaneous utterances of four 2-year-olds were analyzed. Pitch accents were annotated using ToBI (Beckman & Ayers, 1997) and acoustically analyzed. Results reveal three varieties of high pitch accents: (1) H*, rise in f0 on stressed syllable; (2) H+!H* complex-bitonal, higher f0 on preceding syllable, falling f0 throughout accented syllable; (3) H+!H* type-2, f0 plateau on preceding syllable continuing onto stressed syllable. Examining how toddlers use specific intonational elements helps to identify the prosodic contours that make up the child inventory.

5aSC35. Variation in English infant-directed speech. Isabelle Lin (Dept. of Linguist, Univ. of California Los Angeles, 3125 Campbell Hall, UCLA, Los Angeles, CA 90095-1543, isabellelin@ucla.edu), Adam J. Chong (Linguist, Queen Mary Univ. of London, London, United Kingdom), and Megha Sundara (Linguist, Univ. of California Los Angeles, Los Angeles, CA)

In adult-directed speech (ADS), words are rarely produced canonically. Infant-directed speech (IDS) has been argued to contain more canonical productions. However, recent analyses show that IDS is as variable as ADS. Then, how could infants learn to privilege canonical forms, as has been shown for adult listeners? Previous research on variation in IDS has focused on word-final productions. In this study, we investigate whether the extent of variation in IDS differs by the position of a segment in a word. We sampled IDS to 6 infants between 16 and 24-mo-old from the Providence corpus. Utterances with /t/, /d/, /n/, /s/ and /z/ were identified orthographically, forced-aligned, corrected, and transcribed by 3 phonetically-trained native speakers of English. This yielded 28,775 segment tokens in word initial, medial and final position. Results confirmed that IDS is at least as variable as ADS (canonical pronunciations $<$ 50%). However, variation was limited to coda positions; on average, over 90% of onsets were produced canonically compared to just 60% of codas. This positional difference could benefit category learning. Word-initial segments would bolster acquisition of canonical forms, and possibly support word segmentation, while word-final variation would encourage learning of phonetic variants resulting from processes in connected speech.

5aSC36. The influence of siblings on toddlers' mean length of utterance. Mark VanDam, Allison Saur (Speech & Hearing Sci., Washington State Univ., PO Box 1495, Spokane, WA 99202, mark.vandam@wsu.edu), Jenna Anderst (Com Sci Disord, East Wash Univ, Spokane, WA), Daniel Olds (Speech & Hearing Sci., Washington State Univ., Spokane, WA), and Paul De Palma (Comput. Sci., Gonzaga Univ., Spokane, WA)

Linguistic complexity is an indicator of language development in young children. Complexity of a child's linguistic productions have been shown to increase with development, but may be affected by factors such as disability or environmental variables. Here, we look into the role of family composition as a possible influence on a child's developing ability to use increasingly complex language. In particular, we ask if a toddler's mean length of utterance (MLU) is affected by the presence of siblings in the family and whether the sex of the child may play a role. MLU values were extracted from the public HomeBank database [http://homebank.talkbank.org] of transcribed natural child speech for both the target toddler and for siblings present in the recordings. Results indicate a main effect of increased MLU in children without siblings, but interaction effects suggest that differences may be driven by the boys without siblings alone. There was no correlation between the MLU of the target child and the MLU of the sibling. Findings are discussed in terms of family dynamics and joint attention.

5aSC37. Changes in vowel space characteristics during speech development based on longitudinal of measurements of formant frequencies. Brad H. Story, Kate Bunton, and Rebekkah Diamond (Speech, Lang., and Hearing Sci., Univ. of Arizona, 1131 E. 2nd St., P.O. Box 210071, Tucson, AZ 85721, bstory@email.arizona.edu)

During speech development, a child's vocal tract undergoes changes due to growth of anatomic structures. Such changes typically lower the formant frequencies, reshaping the [F_1,F_2] vowel space. Much of what is known about vowel space change, however, is based on cross-sectional formant measurements averaged over children in various age groups. The purpose of this study was to characterize changes in the vowel space of four children between the ages of 2 and 6 years. Longitudinally-collected audio recordings of four children (2F,2M) were selected from the Arizona Child Acoustic Database. Each child had been recorded every four months from ages 2-6 years, and produced a variety of words, phrases, vowel-vowel progressions, and occasional spontaneous speech. Formant frequencies (F_1 and F_2 only) were measured from the recordings using a spectral filtering technique. At each age increment, the formant frequencies for each child were plotted as vowel space density, where the "density" dimension indicates the relative tendency of a talker to produce sound in particular region of the vowel space. The change in location and shape of the density cloud during this period of development will be demonstrated. [Research supported by NIH R01-DC011275, NSF BCS-1145011.]

Session 5aSP

Signal Processing in Acoustics and Underwater Acoustics: Continuous Active Acoustics

Zachary J. Waters, Chair
Physical Acoustics - Code 7130, Naval Research Laboratory, 4555 Overlook Ave. SW, Bldg 2. Rm. 186, Washington, DC 20375

Invited Papers

8:00

5aSP1. Towards Doppler estimation and false alarm rejection for Continuous Active Sonar. Jeffrey R. Bates (Ctr. for Maritime Res. and Experimentation (NATO-STO), STO-CMRE, Viale San Bartolomeo 400, La Spezia, SP 19126, Italy, jeffrey.bates@cmre.nato.int), Doug Grimmett (SPAWAR Systems Ctr. Pacific, San Diego, CA), Gaetano Canepa, and Alessandra Tesei (Ctr. for Maritime Res. and Experimentation (NATO-STO), La Spezia, SP, Italy)

Linear frequency modulated (LFM) continuous active sonar (CAS) waveforms show promise for use in target tracking given that waveforms can be split into sub-waveforms (sub-bands), thereby increasing the target refresh rate. However, reducing the duration and bandwidth of the sub-band decreases the SNR/SRR, adversely affecting detection. We present a target detection technique in which sub-bands are averaged incoherently while exploiting the range bias error (commonly observed in large duration LFM CAS waveforms with significant Doppler) to significantly improve detection. Sub-band averaging, known to reduce the false alarm rate, also mitigates channel coherence losses while maintaining detectability in CAS. A method for performing incoherent averaging over many possible Doppler shifts and identifying contacts via clustering in the 3-dimensional range-bearing-Doppler parameter space will be described. Finally, the promising results obtained with this technique on data acquired by the 2016 Littoral CAS Multi-National Joint Research Project sea trial will be shown. [This work was funded by NATO Allied Command Transformation Future Solutions Branch under the Autonomous Security Network Programme and the LCAS MN-JRP.]

8:20

5aSP2. On the use of decision feedback equalization for continuous active sonar. Konstantinos Pelekanakis, Jeffrey R. Bates, and Alessandra Tesei (Res. Dept., Ctr. for Maritime Res. and Experimentation (NATO-STO), NATO STO CMRE, Viale San Bartolomeo 400, La Spezia, SP 19126, Italy, jeffrey.bates@cmre.nato.int)

Continuous Active SONAR (CAS) systems allow duty cycles up to 100% and so lower target association errors are possible as compared to Pulse Active Sonar (PAS). Large time-bandwidth product Linear Frequency Modulated (LFM) signals are the de facto standard for this type of SONAR systems, yet, these signals suffer from low processing gains when the ocean exhibits low-spatio temporal coherence. In this work, we depart from the typical sub-band matched filter processing and propose to adopt signal processing techniques used in underwater acoustic communications. In particular, we analyze Binary Phase Shift Keying (BPSK) signals via an adaptive Decision-Feedback Equaliser (DFE). The DFE is able to adapt to environmental changes based on the known transmitted bits. The operational bandwidth is 1500–4000 Hz and the BPSK signals are transmitted in two non-overlapping bands to avoid transmit interference during reception. The performance of the proposed system is demonstrated based on field data recorded during a sea experiment off the Coast of La Spezia in October 2017. The key result here is that the Doppler measurement update rate of the echoes is as fast as the bit rate of the transmitted BPSK signal.

8:40

5aSP3. Parametric bandwidth synthesis for active sonar. Jonathan B. Soli (Elec. and Comput. Eng., Duke Univ., 141 Providence Rd., Chapel Hill, NC 27514, jonsoli5@gmail.com) and Jeffrey Krolik (Elec. and Comput. Eng., Duke Univ., Durham, NC)

This talk presents an unconventional range processing technique applicable to linear FM pulsed and continuous active sonar. Parametric bandwidth synthesis (BWS) involves performing linear-predictive extrapolation using a low-order parametric autoregressive (AR) model to extend the bandwidth of received sonar returns. Synthetically increasing the bandwidth of echoes results in improved range resolution and greater pulse compression gain compared to standard matched filtering. It should be noted that despite the resolution improvement, BWS does not improve ranging accuracy beyond that of the transmitted signal. Using a low-order AR model for BWS enhances specular target returns and discrete scatterers, while not enhancing diffuse multipath, reverberation, and noise returns. BWS performance was verified in simulation and with sea trial data from the Littoral Continuous Active Sonar 2015 (LCAS-15) experiment. Processed results confirm that BWS improves range resolution of the echo repeater target, while suppressing multipath, diffuse reverberation, and noise. Analysis of the practical limits of BWS are also included.

9:00

5aSP4. Waveform performance and sidelobe reduction techniques for continuous active sonar (CAS). Matthew T. Adams, Brian B. Bishop, and Nicholas A. Rotker (MITRE Corp., 202 Burlington Rd., Mailstop S118, Bedford, MA 01730, mtadams@mitre.org)

The ability to conduct surveillance and tracking of targets over a wide area necessitates the use of a distributed sensing network and an appropriate processing scheme. One such technique that may be appropriate for the target tracking problem is continuous active sonar (CAS). In September 2016, an experiment was performed in Narragansett Bay, RI, to assess the performance of CAS for tracking a single unmanned underwater vehicle (UUV). PN sequence-coded chirps and LFM pulses of various bandwidths and center frequencies were transmitted from one transducer and received at two hydrophones in different locations. Following the experiment, several range-doppler sidelobe reduction techniques were experimented with to improve target detections and range-doppler estimations. During processing, waveform properties critical to sidelobe reduction performance were identified, and new waveforms have been chosen which exhibit these properties. A new experiment will be conducted in the coming months to demonstrate the performance of these waveforms, and results are expected to show improved detections and range-doppler estimations for various targets.

9:20

5aSP5. Demonstration of a compact quasi-monostatic autonomous underwater vehicle based continuous active sonar. Zachary J. Waters (Physical Acoust. - Code 7130, Naval Res. Lab., 4555 Overlook Ave. SW, Bldg 2. Rm. 186, Washington, DC 20375, zachary.waters@nrl.navy.mil)

Monostatic active sonar systems are typically operated in a pulsed transmit configuration, where an acoustic source is first activated to provide ensonification of the environment and then de-activated while scattered and ambient environmental acoustic energy is collected by a receiving sensor. In applications where the transmitter and receiver are in close proximity as with a monostatic configuration, it is typically impracticable to carry out the simultaneous transmission and reception of acoustic energy. Under such conditions, during continuous transmission, the source may saturate the receive sensors, potentially resulting in an overlap of target scattering returns from down-range with the direct acoustic coupling response from the source-to-receiver as well as reverberation, effectively blinding the sonar. Here, we explore the feasibility of techniques to disambiguate lower level target returns from a continuously transmitting autonomous underwater vehicle (AUV) based sonar. We conclude via theoretical studies supported by empirical analysis that the continuous transmission and reception of acoustic energy in a shallow-water waveguide is feasible for a realistic AUV based sonar system and overview demonstrations of these techniques to detect a bottomed target-object at-sea. [Work Supported by the Office of Naval Research.]

9:40

5aSP6. Inference regarding the state of a mobile underwater object from narrow band observations on a small aperture vertical array. Abner C. Barros (ECE Dept., Univ. of Massachusetts Dartmouth, 285 Old Westport Rd., Darmouth, MA 02747, abarros1@umassd.edu) and Paul J. Gendron (ECE Dept., Univ. of Massachusetts Dartmouth, North Dartmouth, MA)

A narrowband source ensonifies an area of interest in a refractive undersea environment. A small aperture vertical array is employed to infer the depth, range and speed of the acoustic scatterer by resolving the scattered direct and surface interacting wave fronts. Tracking the target by means of a continuous wave transmission is challenging due to the difficulty of inferring the frequencies and angles of the two returned closely spaced wave vectors. The propagation of sound in a refractive medium presents additional challenges for inversion of the wave vectors to range, depth, and speed. Joint posterior inference is made possible with a computational Bayesian Gibbs sampling scheme over all track parameters taking full advantage of the analytic tractability of the conditional densities of the received amplitudes and phases and of the ambient acoustic noise power. The conditional densities of the ordered wave vectors however are constructed numerically by 2-dimensional inverse quantile sampling. The inferred joint posterior density of the target state is obtained by constructing a numerical inverse transformation of the acoustic propagation model and provides posterior confidence intervals over speed, depth, and range. Simulation results demonstrate the approach at received signal to noise ratios (SNR) well below -9 dB and illuminate the limits of depth and speed estimation as a function of both depth and SNR.

Session 5aUWa

Underwater Acoustics: Underwater Soundscape: Measurement and Characterization

Timothy F. Duda, Chair
Woods Hole Oceanographic Institution, WHOI AOPE Dept., MS 11, Woods Hole, MA 02543

Contributed Papers

8:00

5aUWa1. Field verification of acoustic sources of geophysical survey in shallow water conditions. Sei-Him Cheong and Laura Palmer (Marine Wildlife Dept., Gardline Geosurvey Ltd., Endeavour House, Admiralty Rd., Great Yarmouth, Norfolk NR30 3NG, United Kingdom, sei-him.cheong@gardline.com)

Underwater noise generated from offshore survey is a growing concern and is known to have ecological consequences on the marine environment. As a conservation effort to reduce anthropogenic impact to the marine ecosystem, BOEM requires acoustic field verifications of sound sources used in a geophysical survey conducted in US water, to ensure the sound fields will not have detrimental effect to marine wildlife. This paper presents the field verification data from two geophysical surveys conducted in the east coast of USA in summer 2016. The survey locations were considered as area of high ecologically importance due to the close proximity to the migratory routes for a number of Baleen whale species including Fin and Northern Right whale. The recording of the survey sources were analysed spatially based on their spectral and temporal characteristics, to quantify the noise exposure experienced by a potential receptor. A comparative assessment between the sound isopleths obtained from field measurement and practical mitigation area was conducted. This information not only gives invaluable insight on the acoustic propagation in the local environment; it also provides extra confidence for the mitigation practice throughout the geophysical survey.

8:15

5aUWa2. Acoustic ground truthing of seismic noise in Chatham Rise, New Zealand. Sei-Him Cheong and Breanna Evans (Marine Wildlife Dept., Gardline Geosurvey Ltd., Endeavour House, Admiralty Rd., Great Yarmouth, Norfolk NR30 3NG, United Kingdom, sei-him.cheong@gardline.com)

Noise generated by seismic survey is widely recognised as a pervasive pollutant to marine ecosystem. Between the 31st January and 21st March 2016, a geophysical research survey was conducted in Chatham Rise, New Zealand, to collect seismo-acoustic data using a Sercel seismic streamer in order to ground-truth the underwater noise impact assessment, conducted according to the DOC (NZ) Seismic Survey Code of Conduct. Data were analyzed to determine the received sound level at a distance up to 3 km from the source array. This paper establishes the method to predict the impact radii in order to validate the results obtained using Gardline 360M predictive model. The aim was to provide confidence to the capability of predictive modelling for estimating the impact zone of a seismic sound source. Data showed that multipath reflections can fluctuate significantly according to the seafloor topography; however, a very consistent trend can be obtained from direct propagation to confidently establish mitigation radii. Results show that the employment of a seismic streamer for the establishment of effective mitigation radii is technically feasible and may be used as a tool to ground truth predictive modelling as part of mitigation plans to reduce the potential risk of acoustic trauma.

8:30

5aUWa3. The noise of rock n'roll: Incidental noise characterization of underwater rock placement. Rute Portugal, Sei-Him Cheong, Breanna Evans (Marine Wildlife Dept., Gardline GeosurveyLtd., Endeavour House, Admiralty Rd., Great Yarmouth, Norfolk NR30 3NG, United Kingdom, rute.portugal@gardline.com), and James Brocklehurst (Boskalis, Papendrecht, Netherlands)

Underwater noise is a growing concern to conservation and stock management efforts to which supra-national organizations (e.g., OSPAR or the European Union) and governments (e.g., USA) are beginning to respond by building catalogs of the noise introduced in the marine environment by human activity. Rock placement is a construction activity for which there is scarcely any data available. In order to fill the knowledge gap, opportunistic recordings were taken while the Gardline Mk 3 hydrophone array was deployed for Passive Acoustic Monitoring and mitigation for marine mammals. The recordings were analyzed for their spectral and temporal characteristics, a correlation analysis between the amount of rock placed and the intensity of sound produced was made and the suitability of the hydrophone array for the collection of this type of data was assessed.

8:45

5aUWa4. Study on single raindrop noise and energy conversion. Dajing Shang, Qi Li, Rui Tang, Fangzhou Deng, and Shu Liu (Underwater Acoust. Eng. Dept., Harbin Eng. Univ., Nangang District Nantong St. No.145, Harbin City, Heilongjiang Province 150001, China, shangdajing@hrbeu.edu.cn)

The measurement of rainfall in the ocean is more difficult than in land, but the noise signals produced by rainfall can be used for measuring the rainfall in the ocean. In this paper, a single raindrop measurement system was set up to measure the noise of the raindrop, the bubble sound are classified by the radius of the raindrop which was measured by the splash method. The mechanism of the initial impact and the bubble sound are analyzed, and the kinetic energy threshold and acoustic energy conversion efficiency of the single raindrops are also investigated. The results show that the bubble noise is the main noise compared with initial impact noise, the initial impact noise will become larger with the radius of raindrop, the acoustic energy of great and large raindrop is much larger than that of small one, but the tiny and medium raindrop have no acoustic energy. The kinetic energy threshold is not a constant, but it is proportional to the raindrop size.The acoustic energy conversion rate is about 1.04×10^{-3}% for small raindrops, 10^{-5}% for heavy rains and great raindrops, there are almost no acoustic energy for tiny raindrops and medium-sized raindrops.

Session 5aUWb

Underwater Acoustics: Underwater Acoustic Propagation: Models and Experimental Data

Timothy F. Duda, Chair
Woods Hole Oceanographic Institution, WHOI AOPE Dept. MS 11, Woods Hole, MA 02543

Contributed Papers

9:30

5aUWb1. Integrating the Biot-Stoll model with contact squirt flow and shear drag (BICSQS) in the Biot viscosity extended framework. Sri Nivas Chandrasekaran, Karen Brastad Evensen, Elisabeth Grønn Ramsdal, and Sverre Holm (Dept. of Informatics, Univ. of Oslo, Gaustadalléen 23B, Oslo 0373, Norway, srinc@ifi.uio.no)

The viscosity extended Biot theory [Sahay, Geophysics (2008)] incorporates pore fluid viscosity in the constitutive relation of the classical Biot theory. The strain rate term due to pore fluid viscosity turns the zero velocity slow S wave mode into a diffusive process. In this framework, it is claimed that fast P and S waves bleed energy when the slow S wave is generated at an interface or at a discontinuity. On the other hand, the Biot-Stoll model with contact squirt flow and shear drag (BICSQS) [Chotiros and Isakson, JASA (2004)] models the grain to grain contact as compression and shear relaxation processes and obtains a causal poro-elastic frame response. In this work, we incorporated the grain to grain contact physics in the constitutive relation of the Biot theory and integrated it into the viscosity extended Biot framework. With realistic parameters, the fast S wave phase velocity and attenuation in the integrated model show a significant difference over a range of frequencies when compared to BICSQS. Additionally, this integrated model may provide a more accurate match for the measured reflection coefficient.

9:45

5aUWb2. Sound propagation effects of near-seabed internal waves in shallow water. Timothy F. Duda, Andone C. Lavery, Ying-Tsong Lin, and Weifeng G. Zhang (Woods Hole Oceanographic Inst., WHOI AOPE Dept. MS 11, Woods Hole, MA 02543, tduda@whoi.edu)

Computational modeling is used to examine effects of near-seabed nonlinear internal waves. Waves of this type have been observed many times on the South New-England Shelf and elsewhere. The waves appear to be common when the dominant vertical density gradient zone is near the seabed. Many observations have been with profilers that alias the waves, or with instruments fixed to the seafloor that provide snapshots of passing waves. We have recently observed these features in greater detail using acoustic remote sensing from a ship, and we are now in a position to model their effects on sound propagation. Results are presented for many frequencies and many source/receiver/wave geometries. An important question remains about along-crest coherence of the waves, which would govern ducting and other anisotropic propagation effects. These waves are not as well studied as waves linked to a near-surface pycnocline, which can be studied remotely with EM via surface signature. The strong density gradients found in many shallow water areas allow the internal waves to have high frequencies, creating rapidly changing acoustics thereby shortening acoustic coherence time. The related strong sound-speed gradients, found both near the surface and near the seabed, are responsible for the important acoustic effects of the waves.

10:00

5aUWb3. Normal mode coupling through horizontally variable sound speed. Cathy Ann Clark (Sensors & Sonar Systems, NUWCDIVNPT, 1176 Howell St., B1320, R457, Newport, RI 02841, cathy.clark@navy.mil)

In conjunction with construction of a normal mode model specifically designed to compute acoustic propagation in shallow water environments, the coupling associated with horizontally variable sound speed is predicted using a closed form solution in a modified Pekeris wave guide with a hard bottom. The code enables analysis of the correspondence between mode summation and the corresponding wave propagation as represented by ray propagation. The closed form solution is being used to provide insights into horizontal mode coupling and as a benchmark for verification of coupling in the shallow water normal mode solution.

10:15

5aUWb4. Impact of salinity on predicted transmission loss. Jennifer Cooper and Michael Mandelberg (Appl. Phys. Lab., John Hopkins Univ., 11100 Johns Hopkins Rd., Mailstop 8-220, Laurel, MD 20723, jennifer.cooper@jhuapl.edu)

Often ocean sound speed profiles are estimated using a temperature profile measured, for example, with an expendable bathythermograph (XBT) combined with a database value for salinity. The prevalence of this approach is due to the added expense and limited availability of sensors that can simultaneously measure both temperature and salinity profiles, for example conductivity-temperature-depth (CTD) probes. When predicting acoustic performance of a sonar system, using the sound speed profile based only on measured temperature is not always sufficient. This study examines when and where the use of temperature-only profiles is adequate as well as potential improvements to typical sound speed profiles that can be made using limited conductivity measurements (i.e. measurement at a single depth or a small segment of the profile) or with database values based on oceanographic models or chosen based on matching the temperature profile rather than the geographically nearest value. The assessment of a profile's adequacy is based on how well modeled transmission loss (TL) matches the TL modeled using the measured temperature and salinity profile from CTD.

10:30–10:30 Break

10:30

5aUWb5. An immersed interface method for the solution of the wide-angle parabolic equation in range-dependent ocean environments. Roberto Sabatini (CNRS-LMA, Ecully cedex, France) and Paul Cristini (CNRS-LMA, 4, Impasse Nikola Tesla, CS40006, Marseille 13013, France, cristini@lma.cnrs-mrs.fr)

Taking into account accurately a complex topography, within the framework of the parabolic equation method in underwater acoustics, is a difficult

task. Several methods have been successfully proposed in the literature in order to improve the accuracy of numerical simulations. Nevertheless, despite the recent improvements, finding an accurate solution remains an open problem. In this presentation, we propose a novel approach for the treatment of irregular ocean bottoms within the framework of the wide-angle parabolic equation. This new approach is based on the immersed interface method originally developed by LeVeque and Li [SIAM J. Numer. Anal. 31(4), 1019–1044, (1994)]. It is intrinsically energy-conserving and allows to naturally handle generic range-dependent bathymetries. As an illustration of its capabilities, we provide some promising results based on the solution of different benchmark problems.

10:45

5aUWb6. Modeling acoustic wave propagation and reverberation in an ice covered environment using finite element analysis. Blake Simon and Marcia J. Isakson (Appl. Res. Labs., The Univ. of Texas at Austin, 1400 Briarcliff Blvd., Austin, TX 78723, blakesimon8@gmail.com)

A three-dimensional, longitudinally invariant finite element model of acoustic propagation and reverberation in an ice-covered shallow water waveguide has been developed. The ice is modeled as both an elastic medium and a pressure release surface. Transmission loss levels are calculated and compared for both assumptions of ice. Using Fourier synthesis, the time-harmonic acoustic pressure results are transformed into the time domain, and reverberation levels are then compared for both models. Finally, using a fully three-dimensional version of the finite element model, compressional-to-shear wave conversion at the elastic ice and water interface is characterized to inform propagation mechanisms of acoustic waves in Arctic ice sheets. [Work supported by ONR, Ocean Acoustics and the Robert W. Young Award for Undergraduate Student Research in Acoustics.]

11:00

5aUWb7. A characterization of the instantaneous wave front distortion characteristics in SW06 and ASIAEX 2001 data and strategies for exploiting lucky scintillations. Ivars P. Kirsteins (NUWCDIVNPT, 1176 Howell St, Newport, RI 02841, i.kirsteins@gmail.com)

Long line arrays are highly susceptible to signal wave front distortions caused by random medium effects like internal waves. Effects include large localization errors, biases, and loss of array gain. In earlier work, we had showed evidence from real data [Kirsteins and Ge, IEEE UASP workshop, Oct. 2017] that lucky scintillations, i.e., moments when the instantaneous signal wave front is relatively undistorted, occur regularly at much shorter time scales even during periods of strong internal wave activity that could potentially be exploited to improve array processing in environments with apparently poor spatial coherence. To better understand the phenomenon of lucky scintillations and how they can be used in array processing, we characterize in this paper the instantaneous or short time signal wave front distortion behaviors of horizontal line array (HLA) data from the Shallow Water 2006 (SW06) and ASIAEX 2001 experiments provided by the Woods Hole Oceanographic Institute, examining the time scales and the rate at which these lucky scintillations occur. Our analysis suggests an alternative array processing strategy by collecting data snap shots over a time interval matched to the ocean random medium time scales and using only the best snap shots in the estimator.

11:15

5aUWb8. Numerical study on radiation impedance of underwater sound sources in non-anechoic tank. Yiming Zhang, Rui Tang (Harbin Eng. Univ., Nantong St., Nangang District No. 145, Harbin 150001, China, zhangyiming1993@hrbeu.edu.cn), Qi Li, and Xin Feng (Harbin Eng. Univ., Harbin, Heilongjiang, China)

The radiation impedance is an important index to evaluate the characteristics of underwater sound sources. In this study, a numerical method for calculating the radiation impedance of underwater sound sources using ACTRAN is proposed. The radiation impedance is calculated by extracting the sound pressure and velocity on the surface of the sound sources. To verify the accuracy of the numerical method, the radiation impedance of a pulsating sphere source is calculated and the results are in good agreement with the analytical results. The radiation impedance of an elastic spherical shell in air, anechoic tanks, and non-anechoic tanks is calculated and analyzed. The results show that when the frequency is lower than the lowest normal frequency, the radiation impedance of the elastic spherical shell in anechoic tanks and non-anechoic tanks is consistent. When there are normal modes in non-anechoic tank, the radiation impedance of the elastic spherical shell varies with the normal frequencies. The proposed method can be used to calculate and analyze the radiation impedance of complex structures in different environments.

11:30

5aUWb9. Method for the measurement of the underwater transient sound characteristics *in a reverberation tank*. Xinyue Yu, Rui Tang, Qi Li, and Junming Zhang (Harbin Eng. Univ., Nantong Str., Nangang Dist No.145, Harbin 150001, China, tangrui@hrbeu.edu.cn)

A method to evaluate the sound power of transient sound sources in reverberation tanks is proposed and tested. The method is based on the steady state sound field characteristics in closed space, which are analyzed by normal-wave theory. To eliminate the interference caused by the boundary of the sound field, the volume integral of the spatial mean-square pressure in local space is measured and the sound power density of the non-anechoic is then obtained. Using the relationship between the sound field in the enclosed space and that of the free field, the sound power of the sound source can be obtained using the measured spatial mean-square pressure. The sound power of an impulsive sound generated by a spherical sound source is measured in a reverberation tank and the results are compared with the sound power measured in anechoic pool to verify the accuracy of the proposed method. The proposed method can be used to measure the radiated sound power of different types of transient sound sources in reverberation tanks.