

Session 5aBA

Biomedical Acoustics: Cavitation Control and Detection Techniques

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Invited Papers

8:00

5aBA1. Detection and control of cavitation during blood–brain barrier opening: Applications and clinical considerations. Meaghan A. O'Reilly, Ryan M. Jones, Alison Burgess, Cassandra Tyson, and Kullervo Hynynen (Physical Sci., Sunnybrook Res. Inst., 2075 Bayview Ave., Rm. C713, Toronto, ON M4N3M5, Canada, moreilly@sri.utoronto.ca)

Microbubble-mediated opening of the blood–brain barrier (BBB) using ultrasound is a targeted technique that provides a transient time window during which circulating therapeutics that are normally restricted to the vasculature can pass into the brain. This effect has been associated with increases in cavitation activity of the circulating microbubbles, and our group has previously described a method to actively control treatments in pre-clinical rodent models based on acoustic emissions recorded by a single transducer. Recently, we have developed a clinical-scale receiver array capable of detecting bubble activity through *ex vivo* human skullcaps starting at pressure levels below the threshold for BBB opening. The use of this array to spatially map cavitation activity in the brain during ultrasound therapy will be discussed, including considerations for compensating for the distorting effects of the skull bone. Additionally, results from pre-clinical investigations examining safety and therapeutic potential will be presented, and receiver design considerations for both pre-clinical and clinical scale systems will be discussed.

8:20

5aBA2. Passive acoustic mapping of stable and inertial cavitation during ultrasound therapy. Christian Coviello (Inst. of Biomedical Eng., Dept. of Eng. Sci., Univ. of Oxford, Oxford, United Kingdom), James Choi (Dept. of BioEng., Imperial College, London, United Kingdom), Jamie Collin, Robert Carlisle (Inst. of Biomedical Eng., Dept. of Eng. Sci., Univ. of Oxford, Oxford, United Kingdom), Miklos Gyongy (Faculty of Information Technol. and Bionics, Pazmany Peter Catholic Univ., Prague, Hungary), and Constantin C. Coussios (Inst. of Biomedical Eng., Dept. of Eng. Sci., Univ. of Oxford, Inst. of Biomedical Eng., Old Rd. Campus Res. Bldg., Oxford, Oxfordshire OX3 7DQ, United Kingdom, constantin.coussios@eng.ox.ac.uk)

Accurate spatio-temporal characterization, quantification, and control of the type and extent of cavitation activity is crucial for a wide range of therapeutic ultrasound applications, ranging from ablation to sonothrombolysis, opening of the blood-brain barrier and drug delivery for cancer. Passive Acoustic Mapping (PAM) is a technique that utilizes arrays of acoustic detectors, typically coaxially aligned or coincident with the therapeutic elements, to receive acoustic emissions outside the main frequency band of the therapy pulse. The signals received by each detector are then filtered in the frequency domain into harmonics and ultra/subharmonics of the fundamental therapeutic frequency and other broadband components, and subsequently beamformed using a multi-correlation algorithm, which uses measures of similarity between the signals rather than time-of-flight information in order to map sources of non-linear emissions in real time. 2D and 3D cavitation maps obtained using time exposure acoustics beamforming will be presented, and juxtaposed to the greater spatial resolution but increased computational complexity afforded by more advanced algorithms such as the Robust Capon Beamformer (RCB). The spatial correlation between cavitation maps produced using PAM and the associated therapeutic effect will also be discussed in the context of cavitation-enhanced ablation and drug delivery.

8:40

5aBA3. Image-guided sonothrombolysis in a stroke model with a cavitation delivery and monitoring system. Francois Vignon, William T. Shi (Ultrasound Imaging and Therapy, Philips Res. North America, 345 Scarborough Rd., Briarcliff Manor, NY 10510, francois.vignon@philips.com), Jeffry Powers (Philips Ultrasound, Bothell, WA), Feng Xie, Juefei Wu, Shunji Gao, John Lof, and Thomas R. Porter (Cardiology, Univ. of NE Medical Ctr., Omaha, NE)

Microbubbles (MB) and ultrasound (US) can dissolve intra-arterial thrombi. In order to reproducibly deliver the correct cavitation dose and ensure treatment efficacy and safety, we designed a therapeutic US mode with cavitation monitoring. Therapy delivery and recording of the MB signal are achieved with a sector imaging probe. Monitoring is achieved by spectrally analyzing the MB signal: ultraharmonics are a marker of stable cavitation (SC) and broadband noise characterizes inertial cavitation (IC). We used the system in a pig model. Thrombotic occlusions were created by injecting 4-hour old clots bilaterally into the internal carotids. Forty pigs were randomized to either 2.4 MI, 5 μ s pulses with MBs; 1.7 MI, 20 μ s pulses with MBs; and 2.4 MI, 5 μ s pulses without MBs. Angiographic recanalization rates were compared. Cavitation as a function of MI was estimated *in vivo*. Dominant SC started at an applied MI of 0.6

(0.3MI *in situ* after derating by skull attenuation). Dominant IC was estimated to start at an applied MI of 0.9 (0.6 *in situ*). Thus, all therapy settings were in the IC regime. The 2.4MI + MB setting was the most effective (100% recanalization) vs 38% for the 1.7MI + MB and 50% for 2.4 MI without MBs (both $p < 0.05$ compared to 2.4MI + MB). No signs of hemorrhage were found in any animal. In conclusion, higher IC levels are most effective for thrombus dissolution. Spectral analysis techniques can be used to plan and monitor the therapy.

9:00

5aBA4. Timing of high intensity pulses for myocardial cavitation-enabled therapy. Douglas Miller, Chunyan Dou (Radiology, Univ. of Michigan, 3240A Medical Sci. I, 1301 Catherine St., Ann Arbor, MI 48109-5667, douglm@umich.edu), Gabe E. Owens (Pediatrics, Univ. of Michigan, Ann Arbor, MI), and Oliver Kripfgans (Radiology, Univ. of Michigan, Ann Arbor, MI)

Ultrasound pulses intermittently triggered from an ECG signal can interact with circulating microbubbles to produce myocardial cavitation microlesions, which may enable tissue-reduction therapy. The timing of therapy pulses relative to the ECG was investigated to identify the optimal trigger point with regard to physiological response and microlesion production. Rats were anesthetized, prepared for ultrasound, placed in a heated water bath, and treated with 1.5 MHz focused ultrasound pulses aimed by 8 MHz imaging. Initially, rats were treated for 1 min with triggering at each of six different points in the ECG while monitoring blood pressure. Premature complexes, a useful indicator of efficacy, were seen in the ECG, except during early systole. Premature complexes corresponded with blood pressure pulses for triggering during diastole, but not during systole. Next, triggering at three of the time points, end diastole, end systole, or mid-diastole, was tested for the impact on microlesion creation. Microlesions stained by Evans blue dye were scored in frozen sections. There was no statistically significant variation in cardiomyocyte injury. The end of systole was identified as an optimal trigger time point which yielded ECG complexes and substantial cardiomyocyte injury, but minimal cardiac functional disruption.

9:20

5aBA5. Cavitation threshold determination —Can we do it? Gail ter Haar, John Civalè, Ian Rivens, and Marcia Costa (Phys., Inst. of Cancer Res., Phys. Dept., Royal Marsden Hospital, Sutton, Surrey SM2 5PT, United Kingdom, gail.terhaar@icr.ac.uk)

As clinical applications, which harness acoustic cavitation, become more commonplace, it becomes more and more important to be able to determine the threshold pressures at which it is likely to occur. In our studies, we have used a suite of different detection techniques in an effort to determine these thresholds. These include passive cavitation detection, transducer impedance monitoring, and visual appearance. Different methods of acoustic signal processing have been compared. The resultant cavitation thresholds will be discussed.

9:40

5aBA6. Monitoring boiling histotripsy with bubble-based ultrasound techniques. Vera Khokhlova (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, v.khokhlova@gmail.com), Michael Canney (INSERM U556, Lyon, France), Julianna Simon, Tatiana Khokhlova, Joo-Ha Hwang, Adam Maxwell, Michael Bailey, Oleg Sapozhnikov, Wayne Kreider, and Lawrence Crum (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

Cavitation phenomena have been always considered as a predominant mechanism of concern in mechanical tissue damage induced by therapeutic ultrasound. Corresponding methods have been developed to monitor cavitation. Recently, a new high intensity focused ultrasound technology, called boiling histotripsy (BH), was introduced, in which the major physical phenomenon that initiates mechanical tissue damage is vapor bubble growth associated with rapid tissue heating to boiling temperatures. Caused by nonlinear propagation effects and the development of high-amplitude shocks, this tissue heating is localized in space and can lead to boiling within milliseconds. Once a boiling bubble is created, interaction of shock waves with the cavity results in tissue disintegration. While the incident shocks can lead to cavitation phenomena and accompanying broadband emissions, the presence of a millimeter-sized vapor cavity in tissue produces strong echogenicity in ultrasound (US) imaging that can be exploited with B-mode diagnostic ultrasound. Various other methods of imaging boiling histotripsy, including passive cavitation detection (PCD), Doppler or nonlinear pulse-inversion techniques, and high speed photography in transparent gel phantoms are also overviewed. The role of shock amplitude as a metric for mechanical tissue damage is discussed. [Work supported by NIH EB007643, T32DK007779, and NSBRI through NASA NCC 9-58.]

10:00

5aBA7. Control of cavitation through coalescence of cavitation nuclei. Timothy L. Hall, Alex Duryea, and Hedieh Tamaddon (Univ. of Michigan, 2200 Bonisteel Blvd., Ann Arbor, MI 48109, hallt@umich.edu)

Therapeutic ultrasound in the form of SWL, HIFU, or histotripsy frequently generates cavitation nuclei (bubbles 1–10 μm radius), which can persist up to about 1 s before dissolving. These nuclei can attenuate and reflect propagation of acoustic fields reducing SWL efficiency, enhancing HIFU heating, or shifting the location of a histotripsy focal zone making procedures less predictable. Depending on their location, nuclei can also directly cause tissue damage when a high amplitude sound field causes them to undergo inertial cavitation. These undesirable effects can be reduced by using a low amplitude sound field ($MI < 1$) to stimulate coalescence of nuclei through primary and secondary Bjerknes forces. We will show nuclei coalescence significantly reduces sound field attenuation, improves SWL breakup of model kidney stones, and reduces collateral damage in soft tissues. We also show techniques for designing the non-focal acoustic fields for efficient coalescence with 3D printed acoustic lenses. Timothy Hall has a consulting arrangement with Histosonics, Inc., which has licensed intellectual property related to this abstract.

10:20–10:30 Break

5a FRI. AM

10:30

5aBA8. Ultraharmonic intravascular ultrasound imaging with commercial 40 MHz catheter: A feasibility study. Himanshu Shekhar, Ivy Awuor, Steven Huntzicker, and Marvin M. Doyley (Univ. of Rochester, 345 Hope-man Bldg., University of Rochester River Campus, Rochester, NY 14627, himanshuwaits@gmail.com)

The abnormal growth of the vasa vasorum is characteristic of life-threatening atherosclerotic plaques. Intravascular ultraharmonic imaging is an emerging technique that could visualize the vasa vasorum and help clinicians identify life-threatening plaques. Implementing this technique on commercial intravascular ultrasound (IVUS) systems could to accelerate its clinical translation. Our previous work has demonstrated ultraharmonic IVUS imaging with a modified clinical system that was equipped with a commercial 15 MHz peripheral imaging catheter. In the present study, we investigated the feasibility of ultraharmonic imaging with a commercially available 40 MHz coronary imaging catheter. We imaged a flow phantom that had contrast agent microbubbles (Targestar-P-HF, Targeson Inc., CA) perfused in side channels parallel to its main lumen. The transducer was excited at 30 MHz using 10% bandwidth chirp-coded pulses. The ultraharmonic response at 45 MHz was isolated and preferentially visualized using pulse inversion and digital filtering. Side channels with 900 μm and 500 μm diameter were detected with contrast-to-tissue ratios approaching 10 dB for clinically relevant microbubble concentrations. The results of this study indicate that ultraharmonic imaging is feasible with commercially available coronary IVUS catheters, which may facilitate its widespread application in preclinical research and clinical imaging.

10:45

5aBA9. A method to calibrate the absolute receive sensitivity of spherically focused, single-element transducers. Kyle T. Rich and T. Douglas Mast (Biomedical Eng., Univ. of Cincinnati, 3938 Cardiovascular Res. Ctr., 231 Albert Sabin Way, Cincinnati, OH 45267-0586, doug.mast@uc.edu)

Quantitative acoustic measurements of microbubble behavior, including scattering and emissions from cavitation, would be facilitated by improved calibration of transducers making absolute pressure measurements. In particular, appropriate methods are needed for wideband calibration of focused passive cavitation detectors. Here, a substitution method was developed to characterize the absolute receive sensitivity of two spherically focused, single-element transducers (center transmit frequencies 4 and 10 MHz). Receive calibrations were obtained by transmitting and receiving a broadband pulse between the two focused transducers in a pitch-catch, confocally aligned configuration, separated by a distance equal to the sum of the two focal lengths. A calibrated hydrophone was substituted to measure the pressure field in the plane of each receiver's surface. The frequency dependent receive sensitivity at the focus was then calculated for each transducer as the ratio of the receiver-measured voltage and the average hydrophone-measured pressure amplitude across the receiver surface. Calibrations were validated by generating an approximately spherically spreading, broadband pressure wave at the focus of each transducer using a 2-mm diameter transducer and comparing the absolute acoustic pressure measured by each focused transducer to that measured by a calibrated hydrophone.

11:00

5aBA10. Instigation and monitoring of inertial cavitation from nanoscale particles using a diagnostic imaging platform and passive acoustic mapping. Christian Coviello, James Kwan, Susan Graham, Rachel Myers, Apurva Shah, Penny Probert Smith, Robert Carlisle, and Constantin Cousios (Inst. of Biomedical Eng., Univ. of Oxford, ORCRB, Oxford OX3 7DQ, United Kingdom, christian.coviello@eng.ox.ac.uk)

Inertial cavitation nucleated by microbubble contrast agents has been recently shown to enhance extravasation and improve the distribution of anti-cancer agents during ultrasound (US)-enhanced delivery. However, microbubbles require frequent replenishment due to their rapid clearance and destruction upon US exposure and are unable to extravasate into tumor tissue due to their large size. A new generation of gas-stabilizing polymeric

cup-shaped nanoparticles, or "nanocups" (NCs), have been formulated to a size that enables exploitation of the enhanced permeability and retention effect for intratumoral accumulation. NCs provide sustained inertial cavitation, as characterized by broadband emissions, at peak rarefactional pressures readily achievable by diagnostic ultrasound systems. This enables the use of a single low-cost system for B-mode treatment guidance, instigation of cavitation, and real-time passive acoustic mapping (PAM) of the location and extent of cavitation activity during therapy. The significant lowering of the inertial cavitation threshold in the presence of NCs as characterized by PAM is first quantified *in-vitro*. *In-vivo* and *ex-vivo* results in xenograft-implanted tumor bearing mice further evidence the strong presence of inertial cavitation detectable in the tumor at diagnostic levels of US intensity, as confirmed by PAM images overlaid on B-mode in real-time.

11:15

5aBA11. Passive cavitation imaging with nucleic acid-loaded microbubbles in mouse tumors. Man M. Nguyen, Jonathan A. Kopechek, Bima Hasjim, Flordeliza S. Villanueva, and Kang Kim (Dept. of Medicine, Univ. of Pittsburgh, 3550 Terrace St., 562 Scaife Hall, Pittsburgh, PA 15261, manmnguyen@gmail.com)

Ultrasound-targeted microbubble (MB) destruction has been used to deliver nucleic acids to cancer cells for therapeutic effect. Identifying both the location and cavitation activities of the MBs is needed for efficient and effective treatment. In this study, we implemented passive cavitation imaging into a commercially available ultrasound open platform (Verasonics) for a 128-element linear array transducer, centered at 5 MHz, and applied it to an *in-vivo* mouse tumor model. Cationic lipid MBs were loaded with a transcription factor decoy that suppresses STAT3 signaling and inhibits tumor growth in murine squamous cell carcinomas. During systemic MB infusion, ultrasound pulses (4 or 20 cycles) were delivered with a 1-MHz single-element transducer (0.4–1.4MPa peak pressures). Channel data were offline beamformed, band-pass filtered, subtracted from reference images acquired without MBs, and co-registered with B-mode images. During MB infusion, harmonics and broadband emissions were detected in the tumor with both frequency spectra and cavitation images. For 4-cycle 0.4 MPa pulses, harmonic signals at 5 MHz and broadband signals 3–7 MHz were 23 dB and at least 5 dB greater with MBs than without MBs, respectively. These preliminary results demonstrate the feasibility of *in-vivo* passive cavitation imaging and could lead to further studies for optimizing US/MB-mediated delivery of nucleic acids to tumors.

11:30

5aBA12. Non-focal acoustic lens designs for cavitation bubble consolidation. Hedieh A. Tamaddoni, Alexander Duryea, and Timothy L. Hall (Univ. of Michigan, 2740 Barclay Way, Ann Arbor, MI 48105, alavi@umich.edu)

During shockwave lithotripsy, cavitation bubbles form on the surface of urinary stones aiding in the fragmentation process. However, shockwaves can also produce pre-focal bubbles, which may shield or block subsequent shockwaves and potentially induce collateral tissue damage. We have previously shown *in-vitro* that low amplitude acoustic waves can be applied to actively stimulate bubble coalescence and help alleviate this effect. A traditional elliptical transducer lens design produces the maximum focal gain possible for a given aperture. From experiments and simulation, we have found that this design is not optimal for bubble consolidation as the primary and secondary Bjerknes forces may act against each other and the effective field volume is too small. For this work, we designed and constructed non-focal transducer lenses with complex surface geometries using rapid prototyping stereolithography to produce more effective acoustic fields for bubble consolidation during lithotripsy or ultrasound therapy. We demonstrate a design methodology using an inverse problem technique to map the desired acoustic field back to the surface of the transducer lens to determine the correct phase shift at every point on the lens surface. This method could be applied to other acoustics problems where non-focused acoustic fields are desired.

5aBA13. Scavenging dissolved oxygen via acoustic droplet vaporization. Kirthi Radhakrishnan, Christy K. Holland, and Kevin J. Haworth (Internal Medicine, Univ. of Cincinnati, Cardiovascular Ctr. 3972, 231 Albert Sabin Way, Cincinnati, OH 45267, radhakki@ucmail.uc.edu)

Acoustic droplet vaporization (ADV) has been investigated for capillary hemostasis, thermal ablation, and ultrasound imaging. The maximum diameter of a microbubble produced by ADV depends on the gas saturation of the surrounding fluid. This dependence is due to diffusion of dissolved gases from the fluid into the perfluoropentane (PFP) microbubble. This study investigated the change in oxygen concentration in the surrounding fluid after ADV. Albumin-shelled PFP droplets in air-saturated saline (1:30, v/v) were continuously pumped through a flow system and insonified by a focused 2-MHz single-element transducer to induce ADV. B-mode image echogenicity was used to determine the ADV threshold pressure amplitude. The dissolved oxygen concentration in the fluid upstream and downstream of the insonation region was measured using inline sensors. Droplet size distributions were measured before and after ultrasound exposure to determine the ADV transition efficiency. The ADV pressure threshold at 2 MHz was 1.7 MPa (peak negative). Exposure of PFP droplets to ultrasound at 5 MPa peak negative pressure caused the dissolved oxygen content in the surrounding fluid to decrease from $88 \pm 3\%$ to $20 \pm 4\%$. The implications of oxygen scavenging during ADV will be discussed.

5aBA14. Effects of rose bengal on cavitation cloud behavior in optically transparent gel phantom investigated by high-speed observation. Jun Yasuda, Takuya Miyashita (Dept. of Commun. Eng., Tohoku Univ., 6-6-06-5 Aramakiyazaoba, Aoba, Sendai, Miyagiken 980-0801, Japan, j_yasuda@ecei.tohoku.ac.jp), Kei Taguchi (Dept. of Biomedical Eng., Tohoku Univ., Sendai, Japan), Shin Yoshizawa (Dept. of Commun. Eng., Tohoku Univ., Sendai, Japan), and Shin-ichiro Umemura (Dept. of Biomedical Eng., Tohoku Univ., Sendai, Japan)

Sonodynamic treatment is a non-thermal ultrasonic method using sonochemical effect of cavitation bubbles. Rose bengal (RB) is sonochemically active and reduces cavitation threshold and therefore has potential to be an agent for sonodynamic treatment. For the effectiveness and safety of the treatment, controlling cavitation is crucial. In our previous study, we have suggested high-intensity focused ultrasound (HIFU) employing second-harmonic superimposition, which can control cavitation cloud generation by superimposing the second harmonic onto the fundamental. In this study, to investigate the effects of RB on cavitation behavior, a polyacrylamide gel phantom containing RB was exposed to second-harmonic superimposed ultrasound and the generated cavitation bubbles were observed by a high-speed camera. The gel contained three different concentrations of RB, 0, 1, and 10 mg/L. The ultrasonic intensity and exposure duration were 40 kW/cm² and 100 μ s, respectively. The fundamental frequency was 0.8 MHz. In the results, the amount of the inception cloud became higher and the lifetime of bubbles became longer as the RB concentration increased at high reproducibility. The observed RB concentration dependence suggests that the amount of cavitation bubbles can be controlled using second-harmonic superimposition. The observed lifetime extension of bubbles can not only promote sonochemical but also enhance thermal bioeffect.

12:15–12:30 Panel Discussion

FRIDAY MORNING, 31 OCTOBER 2014

INDIANA E, 10:00 A.M. TO 1:00 P.M.

Session 5aED

Education in Acoustics: Hands-On Acoustics Demonstrations for Indianapolis Area Students

Uwe J. Hansen, Cochair

Chemistry & Physics, Indiana State University, 64 Heritage Dr., Terre Haute, IN 47803-2374

Andrew C. H. Morrison, Cochair

Natural Science Department, Joliet Junior College, 1215 Houbolt Rd., Joliet, IL 60431

Acoustics has a long and rich history of physical demonstrations of fundamental (and not so fundamental) acoustics principles and phenomenon. In this session “Hands-On” demonstrations will be set-up for a group of middle school students from the Indianapolis area. The goal is to foster curiosity and excitement in science and acoustics at this critical stage in the students’ educational development and is part of the larger “Listen Up” education outreach effort by the ASA.

Each station will be manned by an experienced acoustician who will help the students understand the principle being illustrated in each demo. Any acousticians wanting to participate in this fun event should email Uwe Hansen (uhansen@indstate.edu) or Andrew C. H. Morrison (amorriso@jjc.edu).

Session 5aNS

Noise: Transportation Noise, Soundscapes, and Related Topics

Alan T. Wall, Chair

Battlespace Acoustics Branch, Air Force Research Laboratory, Bldg. 441, Wright-Patterson AFB, OH 45433

Chair's Introduction—9:45

Contributed Papers

9:50

5aNS1. Traffic monitoring with noise: Investigations on an urban seismic network. Nima Riahi and Peter Gerstoft (Marine Physical Lab., Scripps Inst. of Oceanogr., 9500 Gilman Dr., MC 0238, La Jolla, CA 92093-0238, nriahi@ucsd.edu)

Traffic in urban areas generates not only acoustic noise but also much seismic noise. The latter is typically not perceptible by humans but could, in fact, offer an interesting data source for traffic information systems. To explore the potential for this, we study a 5300 geophone network, which covered an area of over 70 km² in Long Beach, CA, and was deployed as part of a hydrocarbon industry survey. The sensors have a typical spacing of about 100 m, which presents a two-sided processing challenge here: signals beyond a few receiver spacings from the sources are often strongly attenuated and scattered whereas nearby receiver signals may contain complicated near-field effects. We illustrate how we address this issue and give three simple applications: counting cars on a highway section, classifying different types of vehicles passing along a road, and measuring time and take-off velocity of aircraft at Long Beach airport. We discuss future work toward traffic monitoring and also possible connections with acoustical problems.

10:05

5aNS2. Impact of AMX-A1 military aircraft operations on the acoustical environment close to a Brazilian airbase. Olmiro C. de Souza and Stephan Paul (UFMS, Acampamento, 569, Santa Maria, Santa Maria 97050003, Brazil, olmirocz.eac@gmail.com)

Military aircraft operating on airbases usually have a considerable impact on the neighborhood. While the impact of civil aircraft operations can be modeled by commercially available software, the same is hardly possible for military aircraft as no EPNL data are available for such aircraft and flight path are not restricted to those used by civilian operations. Therefore, in this work, the noise impact of AMX-A1 aircraft operating at a Brazilian airbase was evaluated from measurements originally intended for calibration of the noise map of an university that is in the vicinity. From the data, it was possible to obtain $L_{Aeq10min}$ with and without jet noise to see how much does AMX operations influence in the total measurement. Sound exposure levels (SEL) were also calculated. It was found that depending of AMX procedure (Approach, Departure, Touch, Go, etc.), jet noise increases the $L_{Aeq10min}$ up to 10 dB and SEL values reaches 96, 6 dBA in sensitive areas. It will be discussed if A-weighted sound power level can be estimated from the data considering the aircraft as a point source in free field.

10:20

5aNS3. The effect of long-range propagation on contra-rotating open rotor en-route noise levels. Upol Islam (Inst. of sound and Vib. Res. (ISVR), Univ. of Southampton, Highfield Campus, Bldg. 13, Rm. 2009, Southampton, Hampshire SO17 1BJ, United Kingdom, ui1d11@soton.ac.uk)

The purpose was to calculate the en-route noise level produced by an advanced contra-rotating open rotor (CROR) powered aircraft. The en-route

noise is defined as the noise produced by an aircraft in high altitude operation (>3000 m) measured by a microphone 1.2 m above ground level. Calculations were performed for three different aircraft operating conditions—cruise, climb, and descent. For each calculation, the aircraft noise source was modeled as an isolated CROR engine. This noise model was determined from experimental measurements made in a transonic wind tunnel using a 1/6th-scale open rotor rig. En-route noise levels were calculated using the whole aircraft noise prediction code SOPRANO. The CROR noise model were input into SOPRANO and long-distance propagation was calculated using the ray-tracing code APHRODITE, which is implemented within SOPRANO. This ray tracing code requires atmospheric wind speed, wind direction, temperature, and humidity profiles, which were collected from historical data around Europe. The ray tracing method divides the atmosphere up into a number of layers. Meteorological parameters were assumed to vary linearly between the values specified at the boundaries of each layer. Numerous simulations are conducted using different atmospheres in order to assess the impact of atmospheric conditions on the en-route noise levels.

10:35

5aNS4. Gaps in the literature on the effects of aircraft noise on children's cognitive performance. Matthew Kamrath and Michelle C. Vigeant (Graduate Program in Acoust., Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, kamrath64@gmail.com)

In the past two decades, several major studies have indicated that chronic aircraft noise exposure negatively impacts children's cognitive performance. For example, the longitudinal Munich airport study (Hygge, Am. Psychol. Soc., 2002) demonstrated that noise adversely affects reading ability, memory, attention, and speech perception. Moreover, the cross-sectional RANCH study (Stansfeld, Lancet, 2005) found a linear correlation between extended noise exposure and reduced reading comprehension and recognition memory. This presentation summarizes these and other recent studies and discusses four key areas in need of further research (ENNAH Final Report Project No. 226442, 2013). First, future studies should account for all of the following confounding factors: socioeconomic variables, daytime and nighttime aircraft, road, and train noise, and air pollution. Second, multiple noise metrics should be evaluated to determine if the character of the noise alters the relationship between noise and cognition. Third, future research should explore the mitigating effects of improved classroom acoustics and exterior sound insulation. Finally, additional longitudinal studies are necessary: (1) to establish a causal relationship between aircraft noise and cognition; and (2) to understand how changes in the duration of the exposure and in the age of the students influence the relationship. [Work supported by FAA PARTNER Center of Excellence.]

10:50

5aNS5. Acoustic absorption of green roof samples commercially available in southern Brazil. Ricardo Brum, Stephan Paul (Centro de Tecnologia, Universidade Federal de Santa Maria, Rua Erly de Almeida Lima, 650, Santa Maria, RS 97105-120, Brazil, ricardozbrum@yahoo.com.br), Andrey R. da Silva (Centro de Engenharias da Mobilidade, Universidade Federal de Santa Catarina, Joinville, Brazil), and Tenile Piovesan (Centro de Tecnologia, Universidade Federal de Santa Maria, Santa Maria, RS, Brazil)

Previous investigations have shown that green roofs provide many environmental benefits, such as thermal conditioning, air cleaning, and rain water absorption. Nevertheless, information regarding acoustic properties, such as sound absorption and transmission loss is still sparse. This work presents measurements of the sound absorption coefficient of two types of green roofs commercially available in Brazil: the alveolar and the hexa system. Measurements were made in a reverberant chamber according to ISO-354 for different variations of both systems: the alveolar system with 2.5 cm of substrate with and without grass and 4 cm of substrate only. The hexa system was measured with layers of 4 and 6 cm of substrate without vegetation and 6 cm of substrate with a layer of vegetation of the sedum type. For all systems, high absorption coefficients were found for medium and high frequency limits ($\alpha \approx 0.7$) and low absorption in low frequencies ($\alpha \approx 0.2$). This was expected due to the highly porous structure of the substrate. The results suggest that the types of green roofs evaluated in this work could be a good approach to noise control in urban areas.

11:05

5aNS6. The perceived annoyance of urban soundscapes. Adam Craig, Don Knox, and David Moore (School of Eng. and Built Environment, Glasgow Caledonian Univ., 70 Cowcaddens Rd., Glasgow G4 0BA, United Kingdom, Adam.Craig@gcu.ac.uk)

Annoyance is one of the main factors that contribute to a negative view of environmental noise, and can lead to stress-related health conditions. Subjective perception of environmental sounds is dependent upon a variety of factors related to the sound, the geographical location, and the listener. Noise maps used to communicate information to the public about environmental noise in a given geographic location are based on simple noise level measurements and do not include any information regarding how perceptually annoying or otherwise the noise might be. This study involved subjective assessment by a large panel of listeners (N=200) of a corpus of 60 pre-recorded urban soundscapes collected from a variety of locations around Glasgow City Centre. Binaural recordings were taken at three points during each 24 hour period in order to capture urban noise during day, evening, and night. Perceived annoyance was measured using Likert and numerical scales and each soundscape measured in terms of arousal and positive/negative valence. The results shed light on the subjective annoyance of environmental sound in a range of urban locations around Glasgow, and form the basis for development of environmental noise maps, which more fully communicate the effects of environmental noise to the public.

11:20

5aNS7. What comprises a healthy soundscape for the captive Southern White Rhinoceros (*Ceratotherium simum simum*)? Suzi Wiseman (Environ. Geography, Texas State Univ.-San Marcos, 3901 North 30th St., Waco, TX 76708, sw1210txstate@gmail.com), Preston S. Wilson (Mech. Eng., Univ. Texas at Austin, Austin, TX), and Frank Sepulveda (Geophys., Baylor Univ., Killeen, TX)

Many creatures, including the myopic rhinoceros, depend upon hearing and smell to determine their environment. Nature is dominated by meaningful biophonic and geophonic information quickly absorbed by soil and vegetation, while anthropic urban soundscapes exhibit vastly different physical and semantic characteristics, sound repeatedly reflecting off hard

geometric surfaces, distorting and reverberating, and becoming noise. Noise damages humans physiologically, including reproductively, and likely damages other mammals. Rhinos vocalize sonically and infrasonically, but audiograms are unavailable. They generally breed poorly in urban zoos, where infrasonic noise can be chronic. Biological and social factors are studied, but little attention if any is paid to soundscape. We present a methodology to analyze the soundscapes of captive animals according to their hearing range. Sound metrics determined from recordings at various institutions can then be compared and correlations with the health and wellbeing of their animals can be sought. To develop this methodology we studied the sonic, infrasonic, and seismic soundscape experienced by the white rhinos at Fossil Rim Wildlife Center, one of the few U.S. facilities to successfully breed this species in recent years. Future analysis can seek particular parameters known to be injurious to human mammals, plus parameters known to invoke response in animals.

11:35

5aNS8. Shape optimization of acoustic horns using few design variables. Nilson Barbieri (Mech. Eng., PUCPR, Rua Imaculada Conceição, 1155, Curitiba, Paraná 80215-901, Brazil, nilson.barbieri@pucpr.br), Renato Barbieri (Mech. Eng., UDESC, Joinville, Santa Catarina, Brazil), Clebe T. Vitorino, and Key F. Lima (Mech. Eng., PUCPR, Curitiba, Brazil)

The main steps for design of the optimal geometry of acoustic horns employing numerical methods are: the definition of the domain and the restrictions and control of the boundary, the definition of the objective function and the frequency range of interest, the evaluation of the objective function value, and the selection of a robust optimization technique to calculate the optimal value. During the optimization process, the profile is changing continuously until obtaining the optimal horn profile. The main focus of this work was to obtain optimal geometries with the use of few design variables. Two different methods to control the horn profile during the optimization process are used: approximation of the contour of the horn with Hermite polynomials and sinusoidal functions. The numerical results show the efficiency of these methods and it was also found (at least from the engineering point of view) that the solution is not unique to the geometry of the horn to single-frequency. The results for the optimization for more than one frequency are also shown.

11:50

5aNS9. Parametric study of a PULSCO vent silencer. Usama Tohid (Eng., PULSCO, 17945 Sky Park Circle, Ste. G, Irvine, CA 92614, u.tohid@pulsco.com)

We have conducted a parametric study via numerical simulations of a PULSCO vent silencer. The overall objective is to demonstrate the existence of an optimum system performance for a given set of operating conditions by modifying the corresponding geometry of the device. The vent silencer under consideration consists of a perforated diffuser, the silencer body, and a tube module. The tube module consists of a set of tubes through which the working fluid passes. The flow tubes are perforated and surrounded with acoustic packing that is responsible for the attenuation. The mathematical model of the vent silencer is built upon Helmholtz equation for the plane wave solution, and the Delany-Bazley model for the acoustic packing. The geometrical parameters chosen for the parametric study include: the porosity of the diffuser and the flow tubes, the type of packing material used for the tube module, bulk density for the acoustic packing, and the hole diameter of the perforated diffuser and flow tubes. The equations of the mathematical model are discretized over the computational domain and solved with a finite element method. Numerical results in terms of transmission loss, for the system, indicate that diffuser hole size of 1/4" with porosity of 0.1, flow tube hole size of 1/8" with porosity of 0.23, packing density of 16 kg/m³ for TRS-10 and 100 kg/m³ for Advantex provided the optimum results for the chosen set of conditions. The numerical results were found to be in agreement with experimental data.

Session 5aPPa

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

Noah H. Silbert, Chair

*Communication Sciences & Disorders, University of Cincinnati, 3202 Eden Avenue, 344 French East Building,
Cincinnati, OH 45267*

All posters will be on display from 8:00 a.m. to 10:00 a.m. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 8:00 a.m. to 9:00 a.m. and contributors of even-numbered papers will be at their posters from 9:00 a.m. to 10:00 a.m.

Contributed Papers

5aPPa1. Is age-related hearing loss predominantly metabolic? Robert H. Withnell (Speech and Hearing Sci., Indiana Univ., Bloomington, IN) and Margarete A. Ueberfuhr (Systemic NeuroSci., Ludwig-Maximilians Univ., Großhaderner Str. 2, D-82152 Planegg-Martinsried, Munich, Germany, margarete.ueberfuhr@gmx.de)

Studies in animals have shown that age-related hearing loss is predominantly metabolic in origin. In humans, direct access to the cochlea is not usually possible and so non-invasive methods of assessing cochlear mechanical function are required. This study used a non-invasive assay of cochlear mechanical function, otoacoustic emissions, to examine a metabolic versus hair-cell-loss origin for age-related hearing loss. Three subject groups were examined: adult females with clinically normal hearing, adult females with age-related hearing loss, and adult males with noise-induced hearing loss. Contrasting otoacoustic emission input-output functions were obtained for the three groups, suggesting a causal relationship between age-related hearing loss and strial dysfunction.

5aPPa2. Further modeling of temporal effects in two-tone suppression. Erica L. Hegland and Elizabeth A. Strickland (Speech, Lang., and Hearing Sci., Purdue Univ., Heavilon Hall, 500 Oval Dr., West Lafayette, IN 47907, eheland@purdue.edu)

Two-tone suppression, a nearly instantaneous reduction in cochlear gain and a by-product of the active process, has been extensively studied both physiologically and psychoacoustically. Some physiological data suggest that the medial olivocochlear reflex (MOCR), which reduces the gain of the active process in the cochlea, may also reduce suppression. The interaction of these two gain reduction mechanisms is complex and has not been widely studied or understood. Therefore, a model of the auditory periphery that includes the MOCR time course was used to systematically investigate this interaction of gain reduction mechanisms. This model was used to closely examine two-tone suppression at the level of the basilar membrane using suppressors lower in frequency than the probe tone. Results were compared both with and without elicitation of the MOCR. Preliminary results indicate that elicitation of the MOCR reduces two-tone suppression when measured as the total basilar membrane response at the characteristic frequency (CF) of the probe. The purpose of this study was to investigate further by separating the frequency components of the basilar membrane response at CF to determine the excitation produced by the probe and by the suppressor with and without MOCR elicitation. [Research supported by NIH(NIDCD)R01 DC008327 and T32 DC00030.]

5aPPa3. Characterization of cochlear implant-related artifacts during sound-field recording of the auditory steady state response using an amplitude modulated stimulus: A comparison among normal hearing adults, cochlear implant recipients, and implant-in-a-box. Shruti B. Deshpande (Commun. Sci. & Disord., Univ. of Cincinnati, 3202 Eden Ave., Cincinnati, OH 45267-0379, balvalsn@mail.uc.edu), Michael P. Scott (Div. of Audiol., Cincinnati Children's Hospital Medical Ctr., Cincinnati, OH), Fawen Zhang, Robert W. Keith (Commun. Sci. & Disord., Univ. of Cincinnati, Cincinnati, OH), and Andrew Dimitrijevic (Commun. Sci. Res. Ctr., Cincinnati Children's Hospital, Dept. of Otolaryngol., Univ. of Cincinnati, Cincinnati, OH)

Recent work has investigated the use of electric stimuli to evoke auditory steady state response (ASSR) in cochlear implant (CI) users. While more control can be exerted using electric stimuli, acoustic stimuli present natural listening environment for CI users. However, ASSR using acoustic stimuli in the presence of a CI could lead to artifacts. Five experiments investigated the presence and characteristics of CI-artifacts during sound-field ASSR using amplitude modulated (AM) stimulus (carrier frequency: 2 kHz; modulation frequency: 82.031 Hz). Experiment 1 investigated differences between 10 normal hearing (NH) and 10 CI participants in terms of ASSR amplitude versus intensity and onset phase versus intensity. Experiment 2 explored similar relationships for an implant-in-a-box. Experiment 3 investigated correlations between electrophysiological ASSR thresholds (ASSRe) and behavioral thresholds to the AM stimulus (BTAM) for the NH and CI groups. Mean threshold differences (ASSRe-BTAM) were computed for each group and group differences were studied. Experiment 4 investigated the presence of transducer-related artifacts using masking. Experiment 5 investigated the effect of manipulation of intensity and external components of the CI on the ASSR. Overall, results of this study provide the first comprehensive description of the characteristics of CI-artifacts during sound-field ASSR. Implications for future research to further characterize CI-artifacts, thereby leading to strategies to minimize them are discussed.

5aPPa4. Baseline neurophysiological noise levels in children with auditory processing disorder. Kyoko Nagao (Biomedical Res., Nemours/Alfred I. duPont Hospital for Children, 1701 Rockland Rd., CPASS, Wilmington, DE 19803, knagao@nemours.org), L. Ashleigh Greenwood (Audiol. Services, Pediatric, Falls Church, VA), Raj C. Sheth, Rebecca G. Gaffney (Biology, Univ. of Delaware, Newark, DE), Matthew R. Cardinale (College of Osteopathic Medicine, New York Inst. of Technol., New York, NY), and Thierry Morlet (Biomedical Res., Nemours/Alfred I. duPont Hospital for Children, Wilmington, DE)

The current study examined the baseline neurophysiological responses between children with auditory processing disorder (APD) and the control group. Auditory event related potentials were recorded in 23 children with APD (ages 7–12 years, mean age = 8.9 years) and 25 age-matched control

children in response to a /da/ presented to each ear separately (right and left ear conditions). A no-sound condition was recorded as well. Baseline neurophysiological activity was measured as the root mean square amplitude of the 100 ms pre-stimulus period. Preliminary analysis of data from 19 children with APD and 13 controls indicated that the APD group showed significantly greater pre-stimulus amplitude than the control group in the left ear condition, $F(1, 30) = 4.415$, $p = 0.044$, but we did not find significant group differences in the no-sound and right ear conditions, $F(1, 30) = 2.237$, $p = 0.15$ and $F(1, 30) = 0.088$, $p = 0.77$, respectively. The results suggest that children with APD may need a longer time period to return to a resting state than control children when the left ear is stimulated. Hence, these results may indicate asymmetrical neural activities of the auditory pathways in APD.

5aPPa5. Speech spectral intensity discrimination at frequencies above 6 kHz. Brian B. Monson (Dept. of Pediatric Newborn Medicine, Brigham and Women's Hospital, Harvard Med. School, 75 Francis St., Boston, MA 02115, bmonson@research.bwh.harvard.edu), Andrew J. Lotto, and Brad H. Story (Speech, Lang., and Hearing Sci., Univ. of Arizona, Tucson, AZ)

Hearing aids and other communication devices (e.g., mobile phones) have made some recent efforts to extend their bandwidths to represent higher frequencies. The impact of this expansion on speech perception is not well characterized. To assess human sensitivity to speech high-frequency energy (HFE, defined here as energy in the 8- and 16-kHz octave bands), difference limens for HFE level changes in male and female speech and singing were obtained. Listeners showed significantly greater ability to detect level changes in singing vs. speech, but not in female vs. male speech. Mean differences limen scores for speech and singing were about 5 dB in the 8-kHz octave (5.6–11.3 kHz) but 8–10 dB in the 16-kHz octave (11.3–22 kHz). These scores are lower (better) than scores previously reported for isolated vowels and some musical instruments, and similar to scores previously reported for white noise.

5aPPa6. Duration perception of time-varying sounds: The role of the amplitude decay and rise-time modulator. Lorraine Chuen (Psych., Neurosci. & Behaviour, McMaster Univ., Psych. Bldg. (PC), Rm. 102, 1280 Main St. West, Hamilton, ON L8S 4K1, Canada, chuenll@mcmaster.ca) and Michael Schutz (School of the Arts, McMaster Univ., Hamilton, ON, Canada)

It is well known that ramped (rising energy) sounds are perceived as longer in duration than damped (falling energy) sounds that are time-reversed, but otherwise identical versions of one another (Schlauch, Ries & DiGiovanni, 2001; Grassi & Darwin, 2006). This asymmetry has generally been attributed to the under-estimation of damped sound duration, rather than the over-estimation of ramped sound duration. As previous literature most commonly employs exponential amplitude modulators, in the present experiment, we investigate whether altering the nature of this amplitude decay- or rise-time modulator (linear or exponential) would influence this typically observed perceptual asymmetry. Participants performed an adaptive, 2AFC task that assessed the point of subjective equality (PSE) between a standard tone with a constant ramped/damped envelope, and a comparator tone with a "flat," steady state envelope whose duration varied according to a 1-up, 1-down rule. Preliminary results replicated previous findings that ramped sounds are perceived as longer than their time-reversed, damped counterparts. However, for sounds with a linear amplitude modulator, this perceptual asymmetry is partially accounted for by ramped tone over-estimation, contrasting previous findings in the literature conducted with exponential amplitude modulators.

5aPPa7. Relationship between gap detection thresholds and performance on the Advanced Measures of Music Audiation Test. Matthew Hoch (Music, Auburn Univ., Auburn, AL), Judith Blumsack, and Lindsey Soles (CMDS, Auburn Univ., 1199 Haley Ctr., Auburn, AL 36849-5232, blumsjt@auburn.edu)

Considerable neurophysiological, neural imaging, and behavioral research indicates that auditory processing in musicians differs from that of non-musicians (e.g., Musacchia et al., 2007; Ohnishi et al., 2001; Pantev et

al., 1998; Parbery-Clark, 2009). Among the auditory skills in musicians that have been studied are gap detection measures of temporal acuity (Mishra & Panda, 2014; Payne, 2012). These studies typically have compared the gap detection thresholds of musicians and non-musicians. The present work relates gap detection performance to musical aptitude rather than to reported musical training history. In addition, in the present study, gap detection was measured under two different stimulus conditions: the within-channel (WC) condition (in which the sound that precedes the gap is spectrally identical to the sound following the gap) and the across-channel (AC condition) (in which the pre- and post-gap sounds are spectrally different. Results indicate a significant correlation between across-channel gap detection thresholds and musical aptitude and no correlation between within-channel performance and musical aptitude. These results have important implications for temporal acuity as it relates to musical aptitude.

5aPPa8. Modeling response times to analyze perceptual interactions in complex non-speech perception. Noah H. Silbert (Commun. Sci. & Disord., Univ. of Cincinnati, 3202 Eden Ave., 344 French East Bldg., Cincinnati, OH 45267, noah.silbert@uc.edu) and Joseph W. Houpt (Psych., Wright State Univ., Dayton, OH)

General recognition theory (GRT) provides a powerful framework for modeling interactions between perceptual dimensions in identification-confusion data. The linear ballistic accumulator (LBA) model provides powerful methods for analyzing multi-choice (2+) response time (RT) data as a function of evidence accumulation and response thresholds. We extend (static) GRT to the domain of RTs by fitting LBA models to RTs collected in two auditory GRT experiments. Although the mapping between the constructs of GRT (e.g., perceptual separability, perceptual independence) and the components of the LBA (e.g., drift rates, response thresholds) is complex, the dimensional interactions defined in GRT can be indirectly addressed in the LBA framework by testing for invariance of LBA parameters across appropriate subsets of the data. The present work focuses on correspondences between (invariance of) parameters in LBA and perceptual separability and independence in GRT.

5aPPa9. The effect of experience on environmental sound identification. Rachel E. Bash, Brandon J. Cash, and Jeremy Loebach (Psych., St. Olaf College, 1520 St. Olaf Ave., Northfield, MN 55057, bash@stolaf.edu)

The perception of environmental stimuli was compared across normal hearing (NH) listeners exposed to an eight-channel sinuswave vocoder and experienced bilateral, unilateral, and bimodal cochlear implant (CI) users. Three groups of NH listeners underwent no training (control), one day of training with environmental stimuli (exposure), or four days of training with a variety of speech and environmental stimuli (experimental). A significant effect of training was observed. The experimental group performed significantly better than exposure or control groups, equal to bilateral CI users, but worse than bimodal users. Participants were divided into low, medium and high-performing groups using a two-step cluster algorithm. High-performing members were only observed for the CI and experimental conditions, and significantly more low-performing members were observed for exposure and control conditions, demonstrating the effectiveness of training. A detailed item-analysis revealed that the most accurately identified sounds were often temporal in nature or contained iconic repeating patterns (e.g., a horse galloping). Easily identified stimuli were common across all groups, with experimental subjects identifying more short or spectrally driven stimuli, and CI users identifying more animal vocalizations. These data demonstrate that explicit training in identifying environmental stimuli improves sound perception, and could be beneficial for new CI users.

5aPPa10. The U.S. National Hearing Test, a 2013–2014 progress report. Charles S. Watson (Res., Commun. Disord. Technol., Inc., CDT, Inc., 3100 John Hinkle Pl, Bloomington, IN 47408, watson@indiana.edu), Gary R. Kidd (Speech and Hearing, Indiana Univ., Bloomington, IN), James D. Miller (Res., Commun. Disord. Technol., Inc., Bloomington, IN), Jill E. Pre-minger (Surgery, Univ. of Louisville, Louisville, KY), Alex Crowley, and Daniel P. Maki (Res., Commun. Disord. Technol., Inc., Bloomington, IN)

A telephone-administered screening test for sensorineural hearing loss was made publically available in the United States in September 2013. This test is similar to the digits-in-noise test developed by Smits and colleagues in the Netherlands, versions of which are now in use in most European

countries and in Australia. The test was initially offered in the United States for a small fee (\$8, then \$4) but after a year of promotion it became clear that either the fee or the complexity of paying it was inhibiting. During the first month in which the test was subsequently offered free of charge, 31,806 calls were made to the test line, of which 26,507 were completed tests. Analyses of test performance suggest that about 81% of the test takers had at least a mild hearing loss, and 40% had a substantial loss (estimated to be in excess of 45 dB PTA). Follow-up studies are being conducted to determine whether those who failed the test sought a full-scale hearing assessment, and whether those advised to obtain hearing aids did so. [Work funded by Grant No. 5R44DC009719 from the National Institute for Deafness and other Communication Disorders.]

FRIDAY MORNING, 31 OCTOBER 2014

MARRIOTT 1/2, 10:15 A.M. TO 12:15 P.M.

Session 5aPPb

Psychological and Physiological Acoustics: Perceptual and Physiological Mechanisms, Modeling, and Assessment

Anna C. Diedesch, Chair

Hearing & Speech Sciences, Vanderbilt University, Nashville, TN 37209

Contributed Papers

10:15

5aPPb1. Modest, reliable spectral peaks in preceding sounds influence vowel perception. Christian Stilp and Paul Anderson (Dept. of Psychol. and Brain Sci., Univ. of Louisville, 308 Life Sci. Bldg., Louisville, KY 40292, christian.stilp@louisville.edu)

Sensory systems excel at extracting predictable signal properties in order to be optimally sensitive to unpredictable, more informative properties. Studies of auditory perceptual calibration (Kiefte & Kluender, 2008 JASA; Alexander & Kluender, 2010 JASA) showed that when precursor sounds were filtered to emphasize frequencies matching the second formant (F_2) of the subsequent target vowel, vowel perception decreased its reliance on F_2 (predictable cue) and increased reliance on spectral tilt (unpredictable cue). Perceptual calibration occurred when reliable spectral peaks were 20 dB or larger, but findings in profile analysis and spectral contrast detection predict sensitivity to more modest spectral peaks. The present experiments tested identification of vowels varying in F_2 (1000–2200 Hz) and spectral tilt (-12-0 dB/octave), perceptually varying from /u/ to /i/. Listeners first identified vowels in isolation, then following a sentence filtered to add a reliable +2 to +15 dB spectral peak centered at F_2 of the target vowel. Changes in perceptual weights (standardized logistic regression coefficients) across sessions were indices of perceptual calibration. Vowel identification weighted F_2 significantly less when reliable peaks were at least +5 dB, but increases in spectral tilt weights were very modest. Results demonstrate high sensitivity to predictable acoustic properties in the sensory environment.

10:30

5aPPb2. Testing the contribution of spectral cues on pitch strength judgments in normal-hearing listeners. William Shofner (Speech and Hearing Sci., Indiana Univ., 200 S. Jordan Ave., Bloomington, IN 47405, wshofner@indiana.edu) and Marisa Marsteller (Speech, Lang. and Hearing Sci., Univ. of Arizona, Tucson, AZ)

When a wideband harmonic tone complex (wHTC) is passed through a noise vocoder, the resulting sounds can have harmonic structures with large peak-to-valley ratios in the spectra, but little or no periodicity strength in the

autocorrelation functions. Noise-vocoded wHTCs evoke simultaneous noise percepts and pitch percepts similar to those evoked by iterated rippled noises. We have previously shown that spectral cues do not appear to control behavioral responses of chinchillas to noise-vocoded wHTCs in a stimulus generalization task, but do appear to contribute to pitch strength judgments in normal-hearing listeners for noise-vocoded wHTCs relative to non-vocoded wHTCs. To further test the role of spectral cues, normal-hearing listeners judged the pitch strengths of noise-vocoded wHTCs relative to infinitely-iterated rippled noise (IIRN). Stimuli had harmonic structures with a fixed fundamental frequency of 500 Hz and were presented monaurally at 50 dB SL. Listeners' judgments of pitch strength evoked by vocoded wHTCs were generally consistent with peak-to-valley ratios of the stimuli. In order to reduce spectral cues and resolvability, stimuli were high-passed filtered. Pitch strength judgments of vocoded wHTCs were reduced following high-pass filtering. These findings suggest that spectral cues do contribute to pitch perception in human listeners.

10:45

5aPPb3. The role of onsets and envelope fluctuations in binaural cue use. G. Christopher Stecker and Anna C. Diedesch (Hearing and Speech Sci., Vanderbilt Univ. Medical Ctr., 1215 21st Ave. South, Rm. 8310, Nashville, TN 37232-8242, g.christopher.stecker@vanderbilt.edu)

Effective localization of real sound sources requires neural mechanisms to accurately extract and represent binaural cues, including interaural time and level differences (ITD and ILD) in the sound arriving at the ears. Many studies have explored the relative effectiveness of these cues, and how that effectiveness varies with the acoustical features of a sound such as spectral frequency and modulation characteristics. In particular, several classic and recent studies have demonstrated relatively greater sensitivity to ITD and ILD present at sound onsets and other positive-going fluctuations of the sound envelope. The results of those studies have clear implications for how spatial cues are extracted from naturally fluctuating sounds such as human speech, and how that process is altered by echoes, reverberation, and competing sources in real auditory scenes. Here, we review the results of several recent studies to summarize and critique the evidence for envelope-triggered

extraction of ITD and ILD across a wide range of spectral frequencies. A number of competing models for cue extraction in fluctuating envelopes are also considered in light of this evidence. [Work supported by NIH R01-DC011548.]

11:00

5aPPb4. Loudness of a multi-tonal sound field, consisting of either one two-component complex sound source or two simultaneous spatially distributed sound sources. Michaël Vannier and Etienne Parizet (Génie Mécanique Conception, INSA-Lyon, Laboratoire Vibrations Acoustique, 13, Pl. Jean Macé, Lyon 69007, France, michael.vannier@insa-lyon.fr)

The aim of the present study is to provide new elements about the perceived loudness of stationary complex sound fields and test the validity of current models under such conditions. The first part consisted in testing the hypothesis according which the directional loudness of a multi-component sound source could be fully explained by the directional loudness of each of its single components. In this way, the directional loudness sensitivities of a two-component complex sound source (third-octave noise bands centered at 1 kHz and 5 kHz) have been measured in the horizontal plane. Despite a previous equalization in loudness of each component to a frontal reference, a small effect of the azimuth angle on loudness still remained, partly disproving the assumption. In a second part, the influence of the spatial distribution of two sound sources on the global loudness was investigated (with the same two narrow-band noises). No effect has been found by Song (2007) for small incidence angles (10 and 30°). The present experiment extends this result for wide incidence angles and so, under highly dichotic listening situations. Finally, all the subjective data have been compared with the predictions from different models of loudness, and the results will be discussed.

11:15

5aPPb5. Computing interaural differences using idealized head models. Tingli Cai, Brad Rakerd, and William Hartmann (Phys. Astronomy, Michigan State Univ., 567 Wilson Rd., East Lansing, MI 48824, hartman2@msu.edu)

The spherical model of the human head, attributable to Lord Rayleigh, accounts for important features of observed interaural time differences (ITD) and interaural level differences (ILD), but it also fails to capture many details. To gain an intuitive understanding of the failures, we computed ITDs and ILDs for a succession of idealized shapes approximating the human head: sphere, ellipsoid, ellipsoid plus cylindrical neck, ellipsoid plus cylindrical neck plus disk torso. Calculations were done as a function of frequency (100–2500 Hz) and for source azimuths from 10 to 90 degrees using finite-element models. The computations were compared to free-field measurements on a KEMAR manikin. The spherical head model approximated many measured interaural features, but the frequency dependence tended to be too flat in both ITD and ILD. The ellipsoidal head produced greater variation with frequency and therefore agreed better with the measurements, reducing the RMS discrepancies in both ITD and ILD by 35%. Adding a neck further increased the frequency variation. Adding the disk torso further improved the agreement, especially below 1000 Hz, decreasing the ITD discrepancy by another 21%. The evolution of models enabled us to associate details of interaural differences with overall anatomical features. [Work supported by the AFOSR grant 11NL002.]

11:30

5aPPb6. Acoustic reflex attenuation in phon loudness measurements. Julius L. Goldstein (Hearing Emulations LLC, Ariel Premium, Hearing Emulations LLC, 8825 Page Ave., Saint Louis, MO 63114-6105, goldstein.jl@sbglobal.net)

Equal Loudness-level Contours, $ELC(f, L)$, represent the sound pressure level in dB SPL of tones at frequency f that are perceived by normal-hearing

listeners as equally loud as a 1 kHz tone at L dB SPL. Loudness is defined relatively as L phons (Fletcher & Munson, 1933). ELC measurements by Lydolf and Møller (1997) included in the current ISO standard (Suzuki & Takeshima, JASA vol. 116, 2004), show systematic increases in ELC growth rate with loudness above 60 phons and below 1 kHz, which suggests middle-ear attenuation by the acoustic reflex (AR). A steady-state ELC model was assembled including known mechanisms: (1) middle ear transmission modified by a head-related-transfer-function, (2) compressive cochlear amplification (CA) for signaling loudness, (3) a negative feedback model for AR attenuation by CA inputs exceeding AR threshold, and (4) attenuation of pressure-field stimuli by trans-ear drum static pressure. Model parameters were calculated from ELC data using minimum-square-error estimation. AR attenuation below 1 kHz depends on AR attenuation at the 1 kHz loudness reference frequency, but predicted ELCs are relatively insensitive to it. An earlier psychophysical study of AR attenuation, including 1 kHz, is consistent with subject-dependent model predictions (Rabinowitz & Goldstein, JASA vol. 54, 1973; Rabinowitz, 1977). [NIH-Funded.]

11:45

5aPPb7. Effects of tinnitus and hearing loss on functional brain networks involved in auditory and visual short-term memory. Fatima T. Husain, Kwaku Akrofi (Speech and Hearing Sci., Univ. of Illinois at Urbana-Champaign, 901 S. Sixth St., Champaign, IL 61820, husainf@illinois.edu), and Jake Carpenter-Thompson (Neurosci., Univ. of Illinois at Urbana-Champaign, Champaign, IL)

Brain imaging data were acquired from three subject groups—persons with hearing loss and tinnitus (TIN), individuals with similar hearing loss without tinnitus (HL) and those with normal hearing without tinnitus (NH)—to test the hypothesis that TIN and control subjects use different functional brain networks for short-term memory. Previous studies have provided evidence of a link between hearing disorders such as tinnitus and the reorganization of auditory and extra-auditory functional networks. Greater knowledge of this reorganization could lead to the development of more effective therapies. Data analysis was conducted on fMRI data obtained while subjects performed short-term memory tasks with low or high attentional loads, using both auditory and visual stimuli in separate scanning sessions. Auditory stimuli were pure tones with frequencies between 500 and 1000 Hz. Visual stimuli were Korean fonts, unfamiliar to the subjects. We found similar behavioral response across the three groups for both modalities and tasks. However, the groups differed in their brain response, with these differences being more marked for the auditory tasks and not for the tasks involving visual stimuli.

12:00

5aPPb8. Preliminary results of a two-interval forced-choice method for assessing infant hearing sensitivity. Lynne Werner (Speech & Hearing Sci., Univ. of Washington, 1417 North East 42nd St., Seattle, WA 98105-6246, lawerner@u.washington.edu)

Current methods for assessing infants' hearing are yes-no, single interval procedures. Although bias-free statistics can be used to describe the results of such procedures, with the limited number of trials typically available from an individual infant, use of these statistics can be problematic. A two-interval forced choice method based on infants' anticipatory eye movements toward an interesting visual event is currently under development. Preliminary results indicate that a high proportion of both 3- and 7-month-old infants achieve over 80% correct in the detection of a 70 dB SPL 1000 Hz tone presented through an insert earphone. Infants continue to perform better than expected by chance at levels as low as 25 dB SPL. Thus, a test method based on infant eye movements holds potential as an efficient behavioral method for assessing infant hearing.

Session 5aSC

Speech Communication: Speech Perception and Production in Challenging Conditions (Poster Session)

Alexander L. Francis, Chair

Purdue University, SLHS, Heavilon Hall, 500 Oval Dr., West Lafayette, IN 47907

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

Contributed Papers

5aSC1. A new dual-task paradigm to assess cognitive resources utilized during speech recognition. Andrea R. Plotkowski and Joshua M. Alexander (Speech, Lang., and Hearing Sci., Purdue Univ., 500 Oval Dr., West Lafayette, IN 47906, mitche99@purdue.edu)

Listening to ongoing conversations in challenging situations requires explicit use of cognitive resources to decode and process spoken messages. Traditional speech recognition tests are insensitive measures of this cognitive effort, which may differ greatly between individuals or listening conditions. Furthermore, most dual-task paradigms that have been devised for this purpose generally rely on secondary tasks like reaction time and recall that do not reflect real-world listening demands. A new task was designed to capture changes in both speech recognition and verbal processing across different conditions. Listeners heard two sequential sentences spoken by opposite gender talkers in speech-shaped noise. The primary task was a traditional speech recognition test, in which listeners immediately repeated aloud the second sentence in the pair. The secondary task was designed to engage explicit cognitive processes by requiring listeners to write down the first sentence after holding it in memory while listening to and repeating back the second sentence. Test sentences consisted of lists from the PRESTO test (Gilbert *et al.* 2013, *J. Am. Acad. Audiol.* vol. 24, pp. 26–36) that were carefully modified to help ensure list-equivalency. Psychometric results from the revised PRESTO sentence lists and from the new dual-sentence task will be reported.

5aSC2. Vocal effort, coordination, and balance. Robert A. Fuhrman (Linguist, U Br. Columbia, 2613 West Mall, Vancouver, BC, Canada, robert.a.fuhrman@gmail.com), Adriano Barbosa (Electron. Eng., Federal Univ. of Minas Gerais, Belo Horizonte, Brazil), and Eric Vatikiotis-Bateson (Linguist, U Br. Columbia, Vancouver, BC, Canada)

Manipulating speaking and discourse requirements allows us to assess the time-varying correspondences between various subsystems within a talker at different levels of vocal effort. These subsystems include fundamental frequency (F₀) and acoustic amplitude, rigid body (6D) motion of the head, motion (2D) of the body, and postural forces and torques measured at the feet. Analysis of six speakers has confirmed our hypothesis that as vocal effort increases coordination among sub-systems simplifies, as shown by greater correspondence (e.g., the instantaneous correlation) between the various time-series measures. However, at the two highest levels of vocal effort, elicited by having talkers shout to and yell at someone located appropriately far away, elements of the postural force, notably one or more torque components, often show a reduction in correspondence with the other measures. We interpret this result as evidence that talkers become more rigidly coordinated at the highest levels of vocal effort, which can interfere with their balance. Furthermore, the discourse type—shouting at someone to carry on a conversation vs. yelling at someone not expected to reply—can be associated with differing amounts of imbalance.

5aSC3. The gradient effect of transitional magnitude: A source of the vowel context effect. SANG-IM LEE-KIM (Linguist, New York Univ., 45-35 44th St. 1i, Sunnyside, NY 11104, sangim119@gmail.com)

Previous studies have shown that vocalic transitions play an important role in the identification of the consonantal places (e.g., Whalen 1981/1991, Nowak 2006, Babel & McGuire 2013). While it has been intermittently reported that the contribution of transitions may depend on vowel contexts, the common methodology, i.e., C-V cross-splicing, is too coarse to precisely identify the nature of this effect. In the present study, vocalic transitions are systematically manipulated and used as a gradient variable by incrementally removing the transitional period of the three vowels /u a e/ following alveopalatal sibilant /ç/ in Polish. In an identification task, native Polish speakers were given a choice between /ç/ and /s/ for stimuli with varying levels of palatal transitions. The results showed that participants' perception is gradient: greater transitions overall elicit more palatal responses in all vowel contexts. More importantly, it has been shown that the apparent vowel effect can be largely reduced to the relative magnitude of transitions that are specific to each vowel. The low and back vowels elicit greater palatal transitions providing more robust transitional cues in perception, while the high and front vowels elicit smaller or nearly zero palatal transitions providing less robust cues to the sibilants' place.

5aSC4. Adaptive compensation for reliable spectral characteristics of a listening context in vowel perception. Paul Anderson and Christian Stip (Psychol. and Brain Sci., Univ. of Louisville, 2301 S 3rd St., Louisville, KY, paul.anderson@louisville.edu)

When precursor sounds are filtered to emphasize frequencies matching F₂ of a subsequent target vowel, vowel perception decreases reliance on F₂ (predictable cue) and increases reliance on spectral tilt (unpredictable cue) and vice versa. Previously, initial cue weights and weight changes (i.e., perceptual calibration to reliable signal properties) were larger for F₂ than tilt, obscuring whether the magnitude of calibration reflects cue predictability or F₂'s status as a primary cue to vowel identity. Here, vowels varied from /u/ to /i/ in tilt (-12-0 dB/octave) and the full range of F₂ values (1000–2200 Hz) or a reduced range (1300–1900 Hz) designed to decrease F₂ cue weights, making tilt the primary cue for vowel identification. Vowels were presented in isolation, then following sentences filtered to match the target vowel's F₂ or tilt. In isolation, cue weights for F₂ were higher when identifying full-F₂-range vowels and higher for tilt when identifying reduced-F₂-range vowels. Weight changes (calibration) were comparable when the primary cue was predictable; this was also true for predictable secondary cues (tilt for full-F₂-range vowels, F₂ for reduced-F₂-range vowels). Perceptual calibration to reliable signal properties is an adaptive process reflecting cue predictability, not solely *a priori* cue use (e.g., F₂ over tilt).

5aSC5. An approach to the analysis of relations between syllable and sentence perception in quiet and noise in the Speech Perception Assessment and Training System: Preliminary results for ten hearing-aid users. James D. Miller (Res., Commun. Disord. Technol., Inc., 3100 John Hinkle Pl, Ste 107, Bloomington, IN 47408, jamdmill@indiana.edu)

Logistic functions relating abilities to identify syllable onsets, nuclei, and codas in quiet and noise as a function of snr are measured. Syllable perception is the product of these individual abilities. It is found that syllable perception in noise is highly correlated with syllable perception quiet. The relation of sentence perception in the SPATS sentence task with SPATS syllable constituent perception is examined. As shown years ago at the Bell Labs, only modest levels of syllable identification are needed to support nearly perfect levels of sentence perception. Here, it is found that sentence perception in quiet and noise is correlated with syllable perception in quiet, the use of inherent context provided by syllable perception (Boothroyd and Nittrouer (1988)), and with the use of situational context, independent of syllable perception. Finally, the effects of speech perception training on these relations are examined for each of the ten hearing-aid users studied. [Work supported by NIH/NICD Grant R21/R33DC011174 "Multi-site Study of the Efficacy of Speech Perception Training for Hearing-Aid Users," C. S. Watson, PI. Data supplied by cooperating sites: Medical University of South Carolina, J. Dubno, Site PI; University of Memphis, D. Wark, Site PI; and University of Maryland, S. Gordon-Salant, Site PI.]

5aSC6. Neural-scaled entropy predicts the effects of nonlinear frequency compression on speech perception. Varsha Hariram and Joshua Alexander (Speech Lang. and Hearing Sci., Purdue Univ., 500 Oval Dr., West Lafayette, IN 47907, vhariram@purdue.edu)

Signal processing schemes used in hearing aids, such as nonlinear frequency compression (NFC) recode speech information by moving high-frequency information to lower frequency regions. Perceptual studies have shown that depending on the dominant speech sound, compression occurs and the amount of compression can have a significant effect on perception. Very little is understood about how frequency-lowered information is encoded by the auditory periphery. We have developed a measure that is sensitive to information in the altered speech signal in an attempt to predict optimal hearing aid settings for individual hearing losses. The Neural-Scaled Entropy (NSE) model examines the effects of frequency-lowered speech at the level of the inner hair cell synapse of an auditory nerve model [Zilany *et al.* 2013, Assoc. Res. Otolaryngol.]. NSE quantifies the information available in speech by the degree to which the pattern of neural firing across frequency changes relative to its past history (entropy). Nonsense syllables with different NFC parameters were processed in noise. Results are compared with perceptual data across the NFC parameters as well as across different vowel-defining parameters, consonant features, and talker gender. NSE successfully captured the overall effects of varying NFC parameters across the different sound classes.

5aSC7. Tempo-based segregation of spoken sentences. Gary R. Kidd and Larry E. Humes (Speech and Hearing Sci., Indiana Univ., 200 S. Jordan Ave., Bloomington, IN 47405, kidd@indiana.edu)

The ability to make use of differences in speech rhythms to selectively attend to a single spoken message in a multi-talker background was examined in a series of studies. Sentences from the coordinate response measure corpus provided a set of stimuli with a common rhythmic framework spoken by several talkers at similar speaking rates. Subjects were asked to identify two key words spoken in a "target" sentence identified by a word (call sign) near the beginning of the sentence. The target talker was always in the same male voice and either two or six background talkers were presented in different voices (half male and half female). The rate of the background talkers was manipulated to create natural sounding speech that preserved the original pitch and speech rhythms at faster and slower speaking rates. Unaltered target sentences were presented in the presence of faster, unaltered, or slower competing sentences. Performance was poorest with matching target and background tempos, with substantial increases in performance as the tempo differences increased. Modification of the target-sentence rate confirmed that the effect is due to the relative timing of target and background speech, rather than the properties of target-modified background speech. [Work supported by NIH-NIA.]

5aSC8. Perceptual versus cognitive speed in a time-compressed speech task. Michelle R. Molis, Frederick J. Gallun, and Nirmal Srinivasan (National Ctr. for Rehabilitative Auditory Res., Portland VA Medical Ctr., 3710 SW US Veterans Hospital Rd., Portland, OR 97239, michelle.molis@va.gov)

Time-compression retains the information-bearing spectral change present in uncompressed speech, although at a rate that may outstrip cognitive processing speed. To compare the relative importance of perceptual and cognitive processing speed, we compared the understanding of (1) time-compressed stimuli expanded in time via gaps with (2) uncompressed stimuli where spectral change information was removed. We hypothesized that, despite the initial compression, the compressed and expanded stimuli would be more intelligible as it would retain relatively more information-bearing spectral change. Participants were somewhat older listeners (mid-1950s to mid-1960s) with normal hearing or mild hearing loss. Stimuli were spoken seven-digit strings time-compressed via pitch synchronous overlap and add (PSOLA) at three uniform compression ratios (2:1, 3:1, and 5:1). In gap insertion conditions, the total duration of the compressed stimuli was restored via introduction of periodic gaps. This produced signal-to-gap ratios of 1:1, 1:2, and 1:4. For comparison, segments of unaccelerated strings, equal to the duration of the inserted gaps, were zeroed out resulting in the same signal-to-gap ratios. Listeners identified the final four digits of the strings presented in quiet and in a steady-state, speech-shaped background noise (SNR +5). Our hypothesis was supported for the fastest compression rates. [Work supported by VA RR&D.]

5aSC9. Information-bearing acoustic changes are important for understanding vocoded speech in a simulation of cochlear implant processing strategies. Christian Stilp (Dept. of Psychol. and Brain Sci., Univ. of Louisville, 308 Life Sci. Bldg., Louisville, KY 40292, christian.stilp@louisville.edu)

Information-bearing acoustic changes (IBACs) in the speech signal are important for understanding speech. This was demonstrated with cochlea-scaled entropy for cochlear implants (CSE_{CI}), which measures perceptually significant intervals of noise-vocoded speech (Stilp *et al.*, 2013 JASA; Stilp, 2014 JASA; Stilp & Goupell, 2014 ASA). However, vocoding does not necessarily mimic CI processing. Some CI processing strategies present acoustic information in all channels at all times (e.g., CIS) while others present only the n -highest-amplitude channels out of m at any time (e.g., ACE). Here, IBACs were explored in a simulation of ACE processing. Sentences were divided into 22 channels spanning 188–7938 Hz and noise-vocoded. In each 1-ms interval (simulating 1000 pulses/second stimulation rate), only the eight highest-amplitude channels were retained. CSE_{CI} was calculated between 1-ms or 16-ms sentence segments, then summed into 80-ms intervals. High- CSE_{CI} or low- CSE_{CI} intervals were replaced by speech-shaped noise. Consistent with previous studies, replacing high- CSE_{CI} intervals impaired sentence intelligibility more than replacing an equal number of low- CSE_{CI} intervals. Importantly, performance was comparable when 1- or 16-ms IBACs were replaced by noise. Results reveal the perceptual importance of IBACs on rapid timescales after simulated ACE processing, indicating this information is likely available to CI users for understanding speech.

5aSC10. Talker intelligibility across clear and sinewave vocoded speech. Jeremy Loebach, Gina Scharenbroch, and Katelyn Berg (Psych., St. Olaf College, 1520 St. Olaf Ave., Northfield, MN 55057, loebach@stolaf.edu)

Talker intelligibility was compared across clear and sinewave vocoded speech. Ten talkers (5 female) from the Midwest and Western dialect regions recorded samples of 210 meaningful IEEF sentences, 206 semantically anomalous sentences, and 300 MRT words. Ninety-three normal hearing participants provided open set transcriptions of the materials presented in the clear over headphones. Forty-one different normal hearing participants provided open set transcriptions of the materials processed with an eight-channel sinewave vocoder. Transcription accuracy was highest for clear speech compared to vocoded speech, and for meaningful sentences, followed by anomalous sentences and words for both conditions. Weak talker effects were observed for the meaningful sentences in the clear (ranging from 97.7% to 98.2%), but were more pronounced for vocoded versions

(68.5% to 85.5%). Weak talker effects were observed for semantically anomalous sentences in the clear (89.4%–93.3%), but more variability was observed across talkers in the vocoded condition (54.4%–73.7%). Finally, stronger talker effects were observed for clear and vocoded MRT words (83.8%–95.6%, 46.3%–59.0%, respectively). Talker rankings differed across stimulus conditions, as well as across processing conditions, but significant positive correlations between conditions were observed for meaningful and anomalous sentences, but not MRT words. Acoustic and dialect influences on intelligibility will be discussed.

5aSC11. Vowels of four-year-old children with cerebral palsy in Mandarin-learning environment. Li-mei Chen (Foreign Lang. and Lit., National Cheng Kung Univ., Tainan, Taiwan), Yu Ching Lin (Physical Medicine and Rehabilitation, National Cheng Kung Univ., Tainan, Taiwan), Wei Chen Hsu, and Meng-Hsin Yeh (Foreign Lang. and Lit., National Cheng Kung Univ., 1 University Rd, Tainan 701, Taiwan, myonaa@gmail.com)

Characteristics of vowel productions of children with cerebral palsy (CP) were investigated with data from two 4-year-old children with CP and two typically-developing (TD) children in Mandarin-learning environment. Clear vowel productions from picture naming and natural conversation in three 50-minute audio recordings of each child were transcribed and analyzed. Seven parameters were examined: vowel duration of /a/, F2 slope in transition of CV sequence, cumulative change of F2 for vowel /a/, degree of nasalization in oral vowel (A1-P1), percent of jitter, percent of shimmer, and the signal to noise ratio (SNR). Major findings are: (1) The CP group showed shorter vowel duration of /a/; (2) TD group has larger F2 slope in CV transition; (3) No obvious differences were found between TD and CP groups in cumulative change of F2 for vowel /a/, degree of nasalization (A1-P1), and voice perturbation (percent of jitter, percent of shimmer, and SNR). Further study with more participants and with careful data selection can verify findings of this study in search for valid parameters to characterize vowel production of children with CP.

5aSC12. Effects of depression on speech. Saurabh Sahu and Carol Espy-Wilson (Elec. and Comput. Eng., Univ. of Maryland College Park, 8125 48 Ave., Apt 101, College Park, MD 20740, ssahu89@umd.edu)

In this paper, we are investigating the effects of depression on speech. The motivation comes from the fact that neuro-physiological changes associated with depression affect motor coordination and can disrupt the articulatory precision in speech. We use the database collected by Mundt *et al.* (J. Neurolinguist. vol. 20, no. 1, pp. 50–64, Jan. 2007) in which 35 subjects were treated over a 6 week period and study how the changes in mental state are manifest in certain acoustic properties that correlate with the Hamilton Depression Rating Scale (HAM-D), which is a clinical assessment score. We look at features such as the modulation frequencies, aperiodic energy during voiced speech, vocal fold jitter and shimmer, and other cues that are related to articulatory precision. These measures will be discussed in detail.

5aSC13. Pitch production of a Mandarin-learning infant with cerebral palsy. Meng-Hsin Yeh, Li-mei Chen (Foreign Lang. and Lit., National Cheng Kung Univ., 1 University Rd., Tainan 701, Taiwan, myonaa@gmail.com), Chyi-Her Lin, Yuh-Jyh Lin, and Yung-Chieh Lin (Pediatrics, National Cheng Kung Univ., Tainan, Taiwan)

In this study, pitch production were investigated in two Mandarin-learning infants at 6 months of age, an infant with cerebral palsy (CP) and a typically developing (TD) infant. Words with distinct tones in Mandarin differ in meaning. In order to produce a correct tone, having good control of the respiratory and the laryngeal mechanisms are necessary. Thus, producing a correct tone and reaching intelligibility for children with CP is considered to be relatively difficult. In previous studies, Kent and Murray (1982) pointed out that falling contours predominated in infant vocalizations at 3, 6, and 9 months. A study by Chen *et al.* (2013) with 4-year-old children indicated that the mean pitch duration of CP children is 1.3–1.8 times longer than TD counterparts. In adults, Jeng, Weismer, and Kent (2006) found that the pitch slopes of Mandarin in CP adults are smaller than in healthy adults. Three measures were employed in this current study and the major findings are:

(1) Both TD and CP infants produced more falling than rising pitch; (2) The mean duration of pitch in CP is 2.3 times longer than that of TD; (3) The pitch slope in CP is smaller than that of TD.

5aSC14. Linear and non-linear acoustic voice analysis of Persian speaking Parkinson's disease patients. Fatemeh Majdinasab (Speech Therapy, Tehran Univ. of Medical Sci., Tehran, Iran), Maryam Mollashahi, Mansour Vali (Medical Eng., k.N. toosi Univ. of Technol., Tehran, Iran), and Hedieh Hashemi (Dept. of Commun. Sci. & Disord., Univ. of Cincinnati, Cincinnati, OH, hashemihedieh@yahoo.com)

Purpose: Many studies have analyzed acoustic voice characteristics (AVC) of Parkinson's disease patients (PDP) by linear or non-linear methods. The aim of this study is to compare the linear and non-linear approaches in acoustic voice analysis of Persian speaking PDPs. Method: This cross sectional, non-experimental study was done on 27 (15 males, 12 females) PDP and 21 healthy age-sex matched subjects (11 males, 10 females). Patients were chosen from attendants of movement disorders clinic using convenience sampling. All of patients evaluated in "on" medication period. AVC consisting average fundamental frequency (f0), standard deviation of f0, mean percentage of jitter, shimmer, and HNR in prolongation of all Persian vowels /a, e, i, o, u /. PRAAT 5.1.17 software (as a linear tool) and MATLAB (as a non-linear method) used to evaluate AVC. Result: There was not any significant difference between PDPs and normal subjects except for jitter /æ / (0.041) and / e / (0.021). According to non-linear characteristics of Wavelet entropy coefficient, and by mother wavelet with coif1 (in MATLAB), all of AVC of patients differentiated from normal. Conclusion: It seems that non-linear analysis is more detailed method to discriminate dysarthric voice from normal voice. Keywords: Acoustic voice analysis, Parkinson's disease, linear, nonlinear.

5aSC15. Vowel development in children with Down and Williams syndromes. Ewa Jacewicz, Robert A. Fox (Dept. and Speech and Hearing Sci., The Ohio State Univ., 1070 Carmack Rd., 110 Pressey Hall, Columbus, OH 43210, jacewicz.1@osu.edu), Vesna Stojanovik, and Jane Setter (Dept. of Clinical Lang. Sci., Univ. of Reading, Reading, United Kingdom)

Down syndrome (DS) and Williams syndrome (WS) are genetic disorders resulting from different types of genetic errors. While both disorders lead to phonological and speech motor deficits, particularly little is known about vowel production in DS and WS. Recent work suggests that impaired vowel articulation in DS likely contributes to the poor intelligibility of DS speech. Developmental delays in temporal vowel structure and pitch control have been found in children with WS when compared to their chronological matches. Here, we analyze spontaneous speech samples produced by British children with DS and WI and compare them with typically developing children from the same geographic area in Southern England. We focus on the acquisition of fine-grained phonetic details, asking if children with DS and WS are able to synchronize the phonetic and indexical domains while coping with articulatory challenges related to their respective syndromes. Phonetic details pertaining to the spectral (vowel-inherent spectral change) and indexical (regional dialect) vowel features are examined and vowel spaces are derived from formant values sampled at multiple temporal locations. Variations in density patterns across the vowel space are also considered to define the nature of the acoustic overlap in vowels related to each syndrome.

5aSC16. Prosodic characteristics in young children with autism spectrum disorder. Laura Dilley, Sara Cook, Ida Stockman, and Brooke Ingersoll (Michigan State Univ., Dept. of Communicative Sci., East Lansing, MI 48824, ldilley@msu.edu)

The prosody of high-functioning adults and adolescents with autism spectrum disorder (ASD) has been reported to differ from that of typically developing individuals. The present study investigated whether young children under eight years old with ASD differ in prosodic characteristics compared with neurotypical children matched on expressive language ability. Seven children with ASD (38–93 months) and seven neurotypical children (20–30 months) were recorded during naturalistic interactions with a parent. Naïve listeners ($n = 18$) were recruited to rate utterances for: (i) age, (ii)

percentage of intelligible words, (iii) pitch, (iv) speech rate, (v) degree of animation, and (vi) certainty of diagnosis. An acoustic analysis was also conducted of speech rate and fundamental frequency (F0). Results of the rating task showed no statistically significant difference on any measure except estimated age. However, children in the ASD group had a significantly lower mean, maximum, and minimum F0 than children in the control group; there was no significant difference between groups for speech rate. These findings may indicate that speech characteristics alone are unlikely to be a sufficient early sign of an ASD diagnosis.

5aSC17. Speech production changes and intelligibility with a real-time cochlear implant simulator. Lily Talesnick (Neurosci., Trinity College, 300 Summit St., Hartford, CT 06106, lily.talesnick@trincoll.edu) and Elizabeth D. Casserly (Psych., Trinity College, Hartford, CT)

Subjects hearing their speech through a real-time cochlear implant (CI) simulator alter their production in multiple ways, e.g., reducing speaking rate and constricting F1/F2 vowel space. The motivations behind these alterations, however, are currently unknown. Two possibilities are that the changes in speech are due to the influence of a direct feedback loop in which the subject is adjusting speech production to minimize acoustic “error,” or that the changes could reflect the indirect influence of a high cognitive load (stemming from the challenge of hearing through the real-time CI simulator). We explored these two possibilities by conducting a playback experiment in which 35 naïve listeners assessed the intelligibility of speech produced under conditions of normal versus vocoded feedback. Intelligibility of vocoded isolated word stimuli in each condition was tested in both a two-alternative forced choice task (“Which recording is easier to understand?”) and an open-set word recognition task. Listeners found normal-feedback speech significantly more intelligible in both tasks (p 's < 0.0125), suggesting that speakers were not adjusting for direct error correction, but rather due to the influence of an intervening factor, e.g., high cognitive load. Confusion matrix analyses further illuminate the perceptual consequences of the effects of CI-simulated speech feedback.

5aSC18. Hearing and hearing-impaired children's acoustic-phonetic adaptations to an interlocutor with a hearing impairment. Sonia Granlund, Valerie Hazan (Speech, Hearing & Phonetic Sci., Univ. College London (UCL), Rm. 326, Chandler House, 2 Wakefield St., London WC1N 1PF, United Kingdom, s.granlund@ucl.ac.uk), and Merle Mahon (Developmental Sci., Univ. College London (UCL), London, United Kingdom)

In England, the majority of children with a hearing impairment attend mainstream schools. However, little is known about the communication strategies used by children when interacting with a peer with hearing loss. This study examined how children with normal-hearing (NH) and those with a hearing impairment (HI) adapt to the needs of a HI interlocutor, focusing on the acoustic-phonetic properties of their speech. Eighteen NH and 18 HI children between the ages of 9 and 15 years performed two problem-solving communicative tasks in pairs: one session was completed with a friend with normal hearing (NH-directed speech) and one session was done with a friend with a hearing impairment (HI-directed speech). As expected, task difficulty increased in interactions involving a HI interlocutor. HI speakers had a slower speech rate, higher speech intensity, and greater F0 range than NH speakers. However, both HI and NH participants decreased their speech rate, and increased their F0 range, mean F0 and the intensity of their speech in HI-directed speech compared to NH-directed speech. This suggests that both NH and HI children are able to adapt to the needs of their interlocutor, even though speech production is more effortful for HI children than their NH peers.

5aSC19. Objective speech intelligibility prediction in sensorineural hearing loss using acoustic simulations and perceptual speech quality measures. Emma Chiamarello, Stefano Moriconi, and Gabriella Tognola (Inst. of Electronics, Computers and TeleCommun. Eng., CNR Italian National Res. Council, Piazza Leonardo Da Vinci 32, Milan 20133, Italy, gabriella.tognola@ieiit.cnr.it)

A novel approach to objectively predict speech intelligibility in sensorineural hearing loss using acoustic simulations of impaired perception and

objective measures of perceptual speech quality (PESQ) is proposed and validated. Acoustic simulations of impaired perception with different types and degrees of hearing loss were obtained degrading the original speech waveforms by spectral smearing, expansive nonlinearity, and level scaling. The CUNY NST syllables were used as test material. PESQ was used to measure perceptual quality of the acoustic simulations thus obtained. Finally, PESQ scores were transformed into predicted intelligibility scores using a logistic function. Validation of the proposed objective method was performed by comparing predicted intelligibility scores with subjective measures of intelligibility of the degraded speech in a group of ten subjects. Predictive intelligibility scores showed good correlation ($R_{\text{sup}} > 2 < /sup > 0.7$) with subjective intelligibility scores and a low error in the prediction (RMSE = 0.14). The proposed approach could be a valuable aid in real clinical applications where it is needed to measure speech intelligibility and might be of some help in avoiding time-consuming experimental measurements. In particular, this method might be valuable in the characterization of the sensitivity of new speech tests for screening and diagnosing of hearing loss, or in the assessment of the performance of novel algorithms of speech enhancement for a target hearing impairment.

5aSC20. Identification of dialect cues by dyslexic and non-dyslexic listeners. Robert A. Fox (Speech and Hearing Sci., The Ohio State Univ., 110 Pressey Hall, 1070 Carmack Rd., Columbus, OH 43210-1002, fox.2@osu.edu), Gayle Long, and Ewa Jacewicz (Speech and Hearing Sci., The Ohio State Univ., Columbus, OH)

Spoken language encodes two different forms of information: linguistic (related to the message) and indexical (e.g., speaker's age, gender, and regional dialect). However, some speech-language impairments (such as dyslexia) can reduce a listener's ability to process both linguistic and indexical speech cues. For example, Perrachione *et al.* (Science, 333, 2011) demonstrated that individuals with dyslexia were less able to identify new voices than were control listeners. This study examines the ability of listeners with and without dyslexia to identify speaker dialect. Eighty listeners—40 adults and 40 children (20 in each group were dyslexic, 20 were not; 40 were males and 40 were females)—listened to a set of 80 sentences produced by English speakers from Western North Carolina or central Ohio and were asked to identify which region the speaker came from. Results demonstrated that adult listeners were significantly better at dialect identification and that listeners with dyslexia were significantly poorer at dialect identification. More notably, there was a significant age by listener group interaction—the improvement in dialect identification between adults and children was significantly smaller in listeners with dyslexia. This indicates that an initial limitation in language learning can inhibit long-term development of speaker-specific phonetic representations.

5aSC21. Individual differences in the lexical processing of phonetically reduced speech. Rory Turnbull (Linguist, The Ohio State Univ., 222 Oxley Hall, 1712 Neil Ave., Columbus, OH 43210, turnbull@ling.osu.edu)

There is widespread evidence that phonetically reduced speech is processed slower and more effortfully than unreduced speech. However, individual differences in degree and strategies of reduction, and their effects on lexical access, are largely unexplored. This study explored the role of autistic traits in the production and perception of reduced pronunciation variants. Stimuli were recordings of words produced in either high reduction (HR) or low reduction (LR) contexts, extracted from sentences produced by talkers ranging in autism-spectrum quotient (AQ) scores. The reductions in these stimuli were generally small temporal differences, rather than segmental-level alterations such as /t/-flapping. Listeners completed a lexical decision task with these stimuli and the autism-spectrum quotient (AQ) questionnaire. Confirming previous research, the results demonstrate that response times (RTs) to reduced words were slower than to unreduced words. No other effects on RT were observed. In terms of response accuracy, LR words were responded to more accurately than HR words, but this pattern was only observed for temporally reduced words. This LR word accuracy benefit was larger for listeners with more autistic personality traits. These results suggest that individuals differ in the extent to which unreduced speech provides a perceptual benefit.

5aSC22. Change of static characteristics of Japanese word utterances with aging. Mitsunori Mizumachi and Kazuto Ogata (Dept. of Elec. Eng. and Electronics, Kyushu Inst. of Technol., 1-1 Sensui-cho, Tobata-ku, Kitakyushu, 805-8440, Japan, mizumach@ecs.kyutech.ac.jp)

Acoustical characteristics of elderly speech have been investigated in the various viewpoints. Elderly speech can be subjectively characterized by roughness, breathiness, asthenic, and hoarseness. Those characteristics have been individually explained in both medical science and speech science. In particular, the hoarseness, which is caused by a physiological problem with an aged vocal cord, is the most well-known static properties of elderly speech. Change of the hoarseness is quantitatively investigated with aging. Japanese phonetically-balanced 543 word utterances were collected with the cooperation of 153 speakers, whose ages ranged from 20 to 89 years old. Acoustical characteristics of the word utterances were examined in the viewpoints of age and auditory impression. In the static acoustical analysis of Japanese vowels /a/, /e/, /i/, /o/, and /u/, it is confirmed that energy in the high frequency region rises with aging. There is a remarkable energy lift over 4 kHz, and the amount of the energy lift is proportion to the degree of subjective hoarseness.

5aSC23. Effect of formant characteristics on older listeners' dynamic pitch perception. Jing Shen (Commun. Sci. and Disord., Northwestern Univ., 2240 Campus Dr., Evanston, IL 60208, jing.shen@northwestern.edu), Richard Wright (Linguist, Univ. of Washington, Seattle, Washington), and Pamela Souza (Commun. Sci. and Disord., Northwestern Univ., Evanston, IL)

Previous research suggested a large inter-subject variability in dynamic pitch perception among older individuals (Souza *et al.*, 2011). Although data from younger listeners with normal hearing indicate temporal and spectral variations in complex formant characteristics may influence dynamic pitch perception (Green *et al.*, 2002), the present study examines this interaction in an aging population. The stimulus set includes two monophthongs that have static formant patterns and two diphthongs that have dynamic formant patterns. The fundamental frequency at the midpoint in time of each vowel is kept consistent, while the ratio of start-to-end frequency varies in equal logarithmic steps. Older adults with near-normal hearing are tested using an identification task, in which they are required to identify the pitch glide as either "rise" or "fall." An experimental task of AX discrimination is also included to verify the identification data. Results to date show inter-subject variability in dynamic pitch perception among listeners with good static pitch perception. Better pitch glide perception with monophthong than diphthong is observed in those individuals who perform poorly in general. The findings suggest a connection between individual abilities to perceive dynamic pitch and to extract the cues from fundamental and formant frequencies. [Work supported by NIH.]

5aSC24. Sentence recognition in older adults. Kathleen F. Faulkner (Dept. of Psychol. and Brain Sci., Indiana Univ., 1101 E 10th St., Bloomington, IN 47401, katieff@indiana.edu), Gary R. Kidd, Larry E. Humes (Speech and Hearing Sci., Indiana Univ., Bloomington, IN), and David B. Pisoni (Dept. of Psychol. and Brain Sci., Indiana Univ., Bloomington, IN)

Many older adults report difficulty when listening to speech in background noise. These difficulties may arise from some combination of factors, including age-related hearing loss, auditory sensory processing difficulties, and/or general cognitive decline. To perform well in everyday noisy environments, listeners must quickly adapt, switch attention, and adjust to multiple sources of variability in both the signal and listening environments. Sentence recognition tests in noise have been useful for assessing speech understanding abilities because they require a combination of basic sensory/perceptual abilities as well as cognitive resources and processing operations. This study was designed to explore several factors underlying

individual differences in aided speech understanding in older adults. We examined the relations between measures of speech perception, cognition, and self-reported listening difficulties in a group of aging adults (N=40, age range 60–86) and a group of young normal hearing listeners (N=28, age range 18–30). All participants completed a comprehensive battery of tests, including cognitive, psychophysical, speech understanding, as well as the SSQ self-report scale. While controlling for audibility, speech understanding declined with age and was strongly correlated with psychophysical measures, cognition, and self-reported speech understanding difficulties. [Work supported by NIH: NIDCD grant T32-DC00012 and NIA grant R01-AG008293 to Indiana University.]

5aSC25. Individual differences in speech perception in noise: A neuro-cognitive genetic study. Zilong Xie (Dept. of Commun. Sci. & Disord., The Univ. of Texas at Austin, 2504A Whitis Ave. (A1100), Austin, TX 78712, xzilong@gmail.com), W. Todd Maddox (Dept. of Psych., The Univ. of Texas at Austin, Austin, TX), Valerie S. Knopik (Div. of Behavioral Genetics, Rhode Island Hospital, Brown Univ. Med. School, Providence, RI), John E. McGeary (Providence Veterans Affairs Medical Ctr., Providence, RI), and Bharath Chandrasekaran (Dept. of Commun. Sci. & Disord., The Univ. of Texas at Austin, Austin, TX)

Previous work has demonstrated that individual listeners vary substantially in their ability to recognize speech in noisy environments. However, little is known about the underlying sources of individual differences in speech perception in noise. Noise varies in the levels of energetic masking (EM) and informational masking (IM) imposed on target speech. Relative to EM, release from IM places greater demand on selective attention. A polymorphism in exon III of the DRD4 gene has been shown to influence selective attention. Here we investigated whether this polymorphism contributes to individual variation in speech recognition ability. We assessed sentence recognition performance across a range of maskers (1-, 2-, and 8-talker babble, and speech-spectrum noise) among 104 young, normal-hearing adults. We also measured their working memory capacity with Operation Span Task, which relies on selective attention to update and maintain items in memory while performing a secondary task. Results showed that the long variant of the DRD4 gene significantly associated with better recognition performance in 1-talker babble conditions only, and that this relation was mediated by enhanced working memory capacity. These findings suggest that the DRD4 polymorphism can explain some of the individual differences in speech recognition ability, but is specific to IM conditions.

5aSC26. Potential sports concussion identification using acoustic-phonetic analysis of vowel productions. Terry G. Horner (Indiana Univ. Methodist Sports Medicine, 201 Pennsylvania Parkway, Ste. 100, Indianapolis, IN 46280, tghorner@hughes.net) and Michael A. Stokes (Waveform Commun., Indianapolis, IN)

Concussions impair cognitive function and muscle motor control; however, little is known about how this impairment affects speech production. In the present study, concussed athletes speech is recorded at the initial office visit and subsequent visits. The last recording, when the brain is determined to have recovered using present criteria, becomes the baseline. The vocabulary consists of seven h-vowel-d (hVd) words (who'd, heed, hood, hid, had, hud, and heard) produced three times each for a total of 21 productions. The study is focused on vowel characteristics and the limited coarticulatory effects of the hVd vocabulary make it ideal for the study. Duration measurements are made by experimenter analysis and the formant measurements are made using the automatic speech recognition engine ELBOW. The preliminary comparisons from the subjects completing the protocol show formant drift for three or more of the seven vowels, and duration is affected for each talker and each vowel. These results were anticipated since the impairment would affect articulatory movement and timing. The results as well as a discussion of development of an automated real-time concussion identification application will be presented.

5aSC27. Thai phonetically balanced word recognition test: Reliability evaluations and bias and error analysis. Adirek Munthuli (Elec. and Comput. Eng., Thammasat Univ., Khlong Luang, Pathumthani, Thailand), Chutamanee Onsuwan (Linguist, Thammasat Univ., Dept. of Linguist, Faculty of Liberal Arts, Thammasat University, Khlong Luang, Pathumthani 12120, Thailand, consuwan@hotmail.com), Charturong Tantibundhit (Elec. and Comput. Eng., Thammasat Univ., Khlong Luang, Pathumthani, Thailand), and Krit Kosawat (Thailand National Electronics and Comput. Technol. Ctr., Khlong Luang, Pathumthani, Thailand)

Word recognition score (WRS) is one of the measuring techniques used in speech audiometry, a part of a routine audiological examination. The test's accuracy is crucial and largely depends on the test materials. With emphasis on phonetic balance, test-retest reliability, inter-list equivalency, and symmetrical phoneme occurrence, Thammasat University Phonetically Balanced Word Lists 2014 (TU PB'14) were created with five different lists, each with 25 Thai monosyllabic words. TU PB'14 reflects Thai phoneme distribution based on large-scale written Thai corpora, InterBEST [1]. To evaluate its validity and test-retest reliability, the lists were given at five intensity levels (15–55 dB HL) in test and retest sessions to 30 normal-hearing subjects. The differences in performance between the two sessions are not significantly large and correlation coefficients at the linear regions are all positive. Analysis of listeners' errors, including sequence recurrences, was carried out. Errors occurred predominantly in the case of initials, followed by finals and lexical tones. Confusion patterns of initials, finals, and tones

are in line with those found for Thai speech sounds in noise condition. Interestingly, vowels are found to be most resistant to confusion. Finally, possible effect of lexical frequency is examined and discussed.

5aSC28. Talker variability in spoken word recognition: Evidence from repetition priming. Yu Zhang and Chao-Yang Lee (Ohio Univ., W239 Grover Ctr., Ohio University, Athens, OH 45701, yz137808@ohio.edu)

The effect of talker variability on the processing of spoken words is investigated using short-term repetition priming experiments. Prime-target pairs, either repeated (e.g., queen-queen) or unrelated (e.g., bell-queen), were produced by the same or different male speakers. Two interstimulus intervals (ISI, 50 and 250 ms) were used to explore the time course of repetition priming and voice specificity effects. The auditory stimuli were presented to 40 listeners, who completed a lexical decision task followed by a talker voice discrimination task. Results from the lexical decision task showed that the magnitude of priming was attenuated in the different-talker condition, indicating a talker variability effect on spoken word recognition. In contrast, the talker variability effect on priming did not differ between the two ISIs. Talker voice discrimination was faster and more accurate for nonword targets, but not for word targets, indicating a lexical status effect on voice discrimination. Taken together, these results suggest that talker variability affects recognition of spoken words, and that the effect cannot be simply attributed to non-lexical voice discrimination.

FRIDAY MORNING, 31 OCTOBER 2014

INDIANA F, 8:00 A.M. TO 12:30 P.M.

Session 5aUW

Underwater Acoustics: Acoustics, Ocean Dynamics, and Geology of Canyons

John A. Colosi, Cochair

Department of Oceanography, Naval Postgraduate School, 833 Dyer Road, Monterey, CA 93943

James Lynch, Cochair

Woods Hole Oceanographic, MS # 11, Bigelow 203, Woods Hole Oceanographic, Woods Hole, MA 02543

Chair's Introduction—8:00

Invited Papers

8:05

5aUW1. What do we know and what do we need to know about submarine canyons for acoustics? James Lynch, Ying-Tsong Lin, Timothy Duda, Arthur Newhall (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., MS # 11, Bigelow 203, Woods Hole Oceanographic, Woods Hole, MA 02543, jlynch@whoi.edu), and Glen Gawarkiewicz (Physical Oceanogr., Woods Hole Oceanographic Inst., Woods Hole, MA)

Acoustic propagation and scattering in marine canyons is an inherently 3-D problem, both for the environmental input (bottom topography and geology, biology, and physical oceanography) and the acoustic field. In this talk, we broadly examine what our knowledge is of these environmental fields, and what their salient effects should be upon acoustics. Examples from recent experiments off the United States and Taiwan will be presented, along with other historical data. Three dimensional acoustic modeling results will also be presented. Directions for future research will be discussed.

8:25

5aUW2. Ocean dynamics and numerical modeling of canyons and shelfbreaks. Pierre F. Lermusiaux (MechE, MIT, 77 Mass Ave., Cambridge, MA 02139, pierrel@mit.edu, Patrick Haley, Chris Mirabito (MIT, Cambridge, MA 02139), Timothy Duda, and Glen Gawarkiewicz (WHOI, Woods Hole, MA)

Multiscale ocean dynamics and multi-resolution numerical modeling of canyons and shelfbreaks are outlined. The dynamics focus is on fronts, currents, tides, and internal tides/waves that occur in these regions. Due to the topographic gradients and strong internal field gradients, nonlinear terms and non-hydrostatic dynamics can be significant. Computationally, a challenge is to achieve accurate simulations that resolve strong gradients over dynamically significant space- and time-scales. To do so, one component are high-order schemes that are more accurate for the same efficiency than lower-order schemes. A second is multi-resolution grids that allow optimized refinements, such as reducing errors near steep topography. A third are methods that allow to solve for multiple dynamics, e.g., hydrostatic and non-hydrostatic, seamlessly. To address these components, new hybridizable discontinuous Galerkin (HDG) finite-element schemes for (non)-hydrostatic physics including a nonlinear free-surface are introduced. The results of data-assimilative multi-resolution simulations are then discussed, using the primitive-equation MSEAS system and telescoping implicitly two-way nested domains. They correspond to collaborative experiments: (i) Shallow Water 06 (SW06) and the Integrated Ocean Dynamics and Acoustics (IODA) research in the Middle Atlantic Bight region; (ii) Quantifying, Predicting and Exploiting Uncertainty (QPE) in the Taiwan-Kuroshio region; and (iii) Philippines Straits Dynamics Experiment (PhilEx).

8:45

5aUW3. Internal tides in canyons and their effect on acoustics. Timothy F. Duda, Weifeng G. Zhang, Ying-Tsong Lin (Appl. Ocean Phys. and Eng. Dept., Woods Hole Oceanographic Inst., WHOI APOE Dept. MS 11, Woods Hole, MA 02543, tduda@whoi.edu), and Aurelien Ponte (Laboratoire de Physique des Océans, IFREMER-CNRS-IRD-UBO, Plouzane, France)

Internal gravity waves of tidal frequency are generated as the ocean tides push water upward onto the continental shelf. Such waves also arrive at the continental slope from deep water and are heavily modified by the change in water depth. The wave generation and wave shoaling effects have an additional level of complexity where a canyon is sliced into the continental slope. Recently, steps have been taken to simulate internal tides in canyons, to understand the physical processes of internal tides in canyons, and also to compute the ramifications on sound propagation in and near the canyons. Internal tides generated in canyons can exhibit directionality, with the directionality being consistent with an interesting multiple-scattering effect. The directionality imparts a pattern to the sound-speed anomaly field affecting propagation. The directionality also means that short nonlinear internal waves, which have specific strong effects on sound, can have interesting patterns near the canyons. In addition to the directionality of internal tides radiated from canyons, the internal tide energy within the canyons can be patchy and may unevenly affect sound.

9:05

5aUW4. An overview of internal wave observations and theory associated with canyons and slopes. John A. Colosi (Dept. of Oceanogr., Naval Postgrad. School, 833 Dyer Rd., Monterey, CA 93943, jacolosi@nps.edu)

Topographic environments such as canyons and slopes are known to be regions of complex internal-wave behavior associated with wave generation, propagation, and dissipation. Much of this anomalous behavior stems from the kinematic constraint that internal waves must maintain their angle of propagation with respect to the horizontal even after interaction with a sloping boundary. In canyons or on slopes, waves propagating in from deep water or generated locally (mostly by tidal flows) either reflect back out to sea or intensify in energy density as they propagate up slope. In particular, wave intensification can lead to nonlinear phenomena including steepening, breaking, and dissipation. This talk will provide an overview of internal wave observations, modeling, and theory in canyons and on slopes with a particular emphasis on acoustically relevant aspects of the wave field.

9:25

5aUW5. Fiery ice from the sea: Marine gas hydrates. Ross Chapman (Earth and Ocean Sci., Univ. of Victoria, 3800 Finnerty Rd., Victoria, BC V8P5C2, Canada, chapman@uvic.ca)

Marine gas hydrates are cage-like structures of water containing methane or some higher hydrocarbons that are stable under conditions of high pressures and low temperatures. The hydrate structures are formed in sediments of continental margins and are found worldwide. The stability zone extends to about 200 m beneath the sea floor, and hydrates exist in several different forms within the zone, from massive ice-like features at cold seeps on the sea floor to finely distributed deposits in sediment pores over extensive areas. The base of the stability zone is characterized by a strong acoustic impedance change from high velocity hydrated sediments above to low velocity gas below. This acoustic feature generates a strong signal in seismic surveys called the Bottom Simulating Reflector, and it is widely used as an indicator of the presence of hydrates. This paper reviews the current knowledge of hydrate systems from research carried out on the Cascadia Margin off the west coast of Vancouver Island, and in the Gulf of Mexico. The hydrate distributions are different in each of these areas, leading to different effects in acoustic reflectivity.

9:45

5aUW6. South China Sea upper-slope sand dunes acoustics experiment. Ching-Sang Chiu, Ben Reeder (Dept. of Oceanogr., Naval Postgrad. School, 833 Dyer Rd., Rm. 328, Monterey, CA 93943-5193, chiu@nps.edu), Linus Chiu (National Sun Yat-sen Univ., Kaohsiung, Taiwan), Yiing Jang Yang, and Chifang Chen (National Taiwan Univ., Taipei, Taiwan)

Very large subaqueous sand dunes were discovered on the upper continental slope of the Northeastern South China Sea. The spatial distribution and scales of these large sand dunes were mapped by two multibeam echo sounding (MBES) surveys, one in 2012 and the other in 2013. These two surveys represented two pilot cruises as part of a multiyear, US-Taiwan collaborative field study designed to characterize these sand dunes, the associated physical processes and the associated acoustic scattering physics. The main experiment will be carried out in 2014. The combination of MBES, coring and acoustic transmission data obtained from the two pilot cruises have

provided vital initial knowledge of (1) the spatial and temporal scales of the sand dunes from objective analysis, (2) the geoaoustic properties of the dunes based on forward modeling to matching the measured levels, and (3) the anisotropy and translational variability of the transmission loss based on a signal energy analysis of the repeated 1–2 kHz and 4–6 kHz FM signals transmitted by a calibrated sound source towed along two circular tracks, each surrounding a receiver. The results from the pilot cruises are presented and discussed. [The research is sponsored by the US ONR and the Taiwan NSC.]

10:05–10:20 Break

10:20

5aUW7. Three dimensional underwater acoustic modeling on continental slopes and submarine canyons. Ying-Tsong Lin, David Barclay, Timothy F. Duda, and Weifeng Gordon Zhang (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., Bigelow 213, MS#11, WHOI, Woods Hole, MA 02543, ytlin@whoi.edu)

Underwater sound propagation on slopes and canyons is influenced jointly and strongly by the complexity of topographic variability and ocean dynamics. Some integrated ocean and acoustic models have been developed and implemented to investigate such joint acoustic effects. In this talk, an integrated numerical model employing a time-stepping three-dimensional (3D) parabolic-equation (PE) acoustic modeling method and the Regional Ocean Modeling System (ROMS) is presented. Numerical examples of sound propagation and ambient noise in Mid-Atlantic Bight area with realistic environmental conditions are demonstrated. The sound propagation model reveals the focusing of sound due to concave canyon seafloor and the different level of temporal variability of focused and unfocused sound. The ambient noise model is constructed for surface wind generated noise, and the model shows the azimuthal dependency of noise field and its spatial coherence structure. Lastly, a simple sonar performance prediction is made to investigate the variability of the probability of detection in these complex underwater environments. [Work supported by the ONR.]

10:40

5aUW8. Three-dimensional effects in the sound propagation in area of coastal slope. Boris Katsnelson (Marine GeoSci., Univ. of Haifa, 1, Universitetskaya sq, Voronezh 394006, Russian Federation, katz@phys.vsu.ru) and Andrey Malykhin (Phys., Voronezh Univ., Voronezh, Russian Federation)

Coastal slope (wedge) in the ocean is well known “canonical” problem for analysis of manifestation of the horizontal refraction (3d effects) in a shelf zone. In given paper the following effects are reviewed:(1) Spatial variability of the sound field in given area. Areas of one-path and multipath propagation, shadow zones, and caustics in horizontal plane, their properties in dependence on the frequency, influence of bottom parameters;(2) Interference structure of the sound field in the horizontal plane in dependence on mode number and frequency;(3) Distribution of the sound field in vicinity of curvilinear coastal line, for example, gulf, bay, peninsula etc. (shadow zones, multipath area, and whispering gallery waveguide in horizontal plane);(4) Temporal variability of signals due to frequency dependence of the horizontal refraction and in turn pulse compression/decompression and time reversal in multipath area;(5) time-frequency diagrams; Mentioned and other effects can change properties of bottom and surface reverberation, scattering, noise field distribution, attenuation in area of coastal wedge. The corresponding estimations are presented. [Work was supported by BSF, Grant 2010471, RFBR-NSFC Grant 14-05-91180.]

Contributed Papers

11:00

5aUW9. Analytic prediction of acoustic coherence time scales in continental-shelf environments with random internal waves. Zheng Gong, Tianrun Chen (Mech. Eng., Massachusetts Inst. of Technol., 5-435, 77 Massachusetts Ave., Cambridge, MA 02139, zgong@mit.edu), Purnima Ratilal (Elec. and Comput. Eng., Northeastern Univ., Boston, MA), and Nicholas C. Makris (Mech. Eng., Massachusetts Inst. of Technol., Cambridge, MA)

An analytical model derived from normal mode theory for the accumulated effects of range-dependent multiple forward scattering is applied to estimate the temporal coherence of the acoustic field forward propagated through a continental-shelf waveguide containing random three-dimensional internal waves. The modeled coherence time scale of narrow band low-frequency acoustic field fluctuations after propagating through a continental-shelf waveguide is shown to decay with a power-law of range to the $-1/2$ beyond roughly 1 km, decrease with increasing internal wave energy, to be consistent with measured acoustic coherence time scales. The model should provide a useful prediction of the acoustic coherence time scale as a function of internal wave energy in continental-shelf environments. The acoustic coherence time scale is an important parameter in remote sensing applications because it determines (i) the time window within which standard coherent processing such as matched filtering may be conducted, and (ii) the number of statistically independent fluctuations in a given measurement period that determines the variance reduction possible by stationary averaging.

11:15

5aUW10. Modeling three dimensional environment and broadband acoustic propagation in Arctic shelf-basin region. Mohsen Badiey, Andreas Muenchow, Lin Wan (College of Earth, Ocean, and Environment, Univ. of Delaware, 261 S. College Ave., Robinson Hall, Newark, DE 19716, badiey@udel.edu), Megan S. Ballard (Appl. Res. Labs., Univ. of Texas, Austin, Delaware), David P. Knobles, and Jason D. Sagers (Appl. Res. Labs., Univ. of Texas, Austin, TX)

Rapid climate change over the last decade has created a renewed interest in the nature of underwater sound propagation in the Arctic Ocean. Changes in the oceanography and surface boundary conditions are expected to cause measurable changes in the propagation and scattering of low frequency sound. Recent measurements of a high-resolution three-dimensional (3-D) sound speed structure in a $50 \times 50 \text{ km}^2$ region in an open-water shelf-basin region of the Beaufort Sea offer a unique and rare opportunity to study the effects of a complex oceanography on the acoustic field as it propagates from the deep basin onto the continental shelf. The raw oceanography data were analyzed and processed to create a 3-D sound speed field for the water column in the basin-slope-shelf area. Recent advances in both 2-D and 3-D acoustic modeling capability allow one to study the effects of the range- and azimuth-dependent water column layers on the frequency-dependent acoustic modal structure. Of particular interest is the nature of the 3-D and mode-coupling effects on the frequency response induced by the oceanography. The results will likely be useful in designing acoustic experiments with serious logistical constraints in the rapidly changing Arctic Ocean.

11:30

5aUW11. Underwater jet noise simulation based on a Large Eddy Simulation/Lighthill hybrid method. GuoQing Liu (School of Naval Architecture and Ocean Eng., Huazhong Univ. of Sci. and Technol., Wuhan 430074, China China, liugq_2010@163.com), Tao Zhang, YongOu Zhang (School of Naval Architecture and Ocean Eng., Huazhong Univ. of Sci. and Technol., Wuhan, Hubei Province, China), Huajiang Ouyang (School of Eng., Univ. of Liverpool, Liverpool, United Kingdom), and Xu Li (School of Naval Architecture and Ocean Eng., Huazhong Univ. of Sci. and Technol., Wuhan, China)

In recent years, extensive researches about the numerical method for aeroacoustics noise simulation have been made. However, the research of hydrodynamic noise develops slowly. In this paper, a hybrid method of combining Large Eddy Simulation (LES) and Lighthill's acoustic analogy theory is established to compute the hydrodynamic noise, which is based on the preliminary study of the method for aerodynamic noise prediction under low Mach number. First, the model of three-dimensional underwater jet is determined by an experimental model. Meanwhile, the CFD mesh and the acoustic mesh are both prepared. Then, the flow field of underwater jet is simulated with LES. The characteristics of turbulent flow are analyzed by the pressure difference and the uniformity coefficient of velocity. After that, the noise of underwater jet is simulated using the theory of Lighthill's acoustic analogy. Finally, the solutions obtained by the hybrid method are compared with the experimental data available in open literature. In conclusion, the sound pressure level at the observation point agrees well with the experimental data. The LES/Lighthill hybrid method is able to compute the underwater jet noise and the hydrodynamic noise.

11:45

5aUW12. Formation sparse aperture antenna arrays based on the sequence Costas. Igor I. Anikin (Concern CSRI Elektropribor, JSC, 30, Malaya Posadskaya Ul., St. Petersburg 197045, Russian Federation, anikin1952@bk.ru)

To obtain high spatial resolution in sonar, ultrasonic image, radar, seismic, and radio astronomy use active antenna arrays that contain a large number of elements. To reduce the cost of such an antenna arrays used with sparse aperture. In this approach, the antenna array is partitioned into several subarrays. Geometric size subarray equivalent equidistant placement $N_c * N_c$ elements. Subarrays filled N_c elements arranged according to the sequence Costas N_c -th order. Each filled subarrays own Costas sequence. As a result, the number of elements in the array is reduced in times N_c . Form the beam pattern in the main sections close to the plane shape of the beam pattern of equidistant antenna array, and the directivity factor is almost independent of frequency band. The upper frequency is reduced in the directivity factor $(\pi \cdot N_c)/2$ times as compared with the plane equidistant antenna array. Using a decaying distribution can be reduced in amplitude level of the side lobe in the principal planes. Thus, by setting the order of the Costas sequence can in each case to optimize the degree of reduction of the number of elements in the array antenna at a predetermined directivity factor.

12:00

5aUW13. Ultra low frequency electromagnetic underwater sound source. Wei Lu and Yu Lan (College of Underwater Acoust. Eng., Harbin Eng. Univ., 145 Nantong St., Nangang District, Harbin 150001, China, luwei@hrbeu.edu.cn)

A detail analysis is presented of one ultra low frequency sound source which is smaller and lighter than conventional piezoelectric ultra low frequency sound source. This sound source is single piston vibration using the electromagnetic principle. The radiation characteristic of sound source is researched by single piston radiation model in low frequency. The dynamic characteristic such as resonant frequency, vibrant displacement of sound source is researched by analytic method and finite element method. In analytic method, the electricity-magnetism, magnetism-force, and force-vibration conversion models of sound source are established by differential equations in different coupling physical fields, and the dynamic characteristics based on the conversion models are simulated by combining and solving differential equations using the MATLAB/SIMULINK. In finite element method, using transient solver of electromagnetic analysis finite element software Ansoft, the dynamic characteristics of sound source are solved. Optimizing the dynamic characteristic of sound source by adjusting magnetic circuit, drive coil, and elastic component parameters, the resonant frequency and radiated sound power of sound source are determined. One prototype sound source design, in which the source level is 184 dB in frequency 73 Hz by calibration, is fabricated that demonstrated proof-of-concept.

12:15

5aUW14. Simplex underwater acoustic communications using passive time reversal. Lin Sun, Haisen Li, Bo Zou, and Ruo Li (College of Underwater Acoust. Eng., Harbin Eng. Univ., Harbin Eng. University, No.145 Nantong St., Nangang District, Harbin City, Heilongjiang Province., Harbin 150001, China, sunlinhrb@sina.com)

The spatial-temporal compression, which is achieved through using simple time reversal (TR) process, can reduce the inter-symbol interference and increase the signal strength. The active TR needs two-way propagation, so it cannot be used in simplex underwater acoustic communications. Based on the one-way propagation property of passive TR, a simplex underwater acoustic communication method using passive TR is proposed. The proposed method is considered in two scenarios: uplink transmission from a single send-only element to an array and downlink transmission from an array to a single receive-only element. The principle of proposed method is analyzed in theory and the performance of proposed method is verified through experiment. Results demonstrate that passive TR process can improve the output signal-to-noise ratio and decrease the bit error rate, so the performance of proposed method is superior to that of simplex acoustic communication method without using passive TR.