Session 5aAA

Architectural Acoustics: A Variety of Interesting Research and Observations in Architectural Acoustics

David S. Woolworth, Chair
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Chair's Introduction—8:00

Contributed Papers

8:05

5aAA1. A potential new and better method for measuring transmission loss in the field. Paul D. Schomer (Schomer and Assoc., Inc., 2117 Robert Dr., Champaign, IL 61821, schomer@SchomerAndAssociates.com)

In a recent study, transmission loss (TL) measurements were made from outdoors-to-indoors and indoors-to-outdoors of a house. These results agree with one another within 0.6 dB. The agreement achieved in this recent study is believed to be because the indoor measurements for the indoor-to-outdoor TL were made at the party-wall surface of the reverberant space. This current paper demonstrates what amounts to a form of pressure doubling at the surfaces of the room containing the reverberant field. It is this higher level that must be used in the TL calculation from indoors-to-outdoors; not the reverberant field measured interior to the room. The actual increase in level for this reverberant-field pressure enhancement appears to be close to + 2.7 dB, which is consistent with measurements of free-field pressure doubling on a hard surface, which are theoretically + 6 dB, but typically measured to be + 5 to + 5.5 dB. This factor of + 2.7 dB for reverberant-field pressure doubling should be applicable to all measurements that use a reverberant space such as laboratory facilities that measure transmission loss from a reverberant room to a more absorptive space.

8:20

5aAA2. Comparison of interior noise levels produced by rain impinging on several commercial roof constructions. Logan D. Pippitt, Michelle L. Huey, and Robert C. Coffeen (Architecture, Univ. of Kansas, Marvin Hall, 1465 Jayhawk Blvd., Lawrence, KS 66045, ldpippitt@gmail.com)

Noise caused by rain on commercial roofs is an ever-occurring problem. Due to climate changes, rainfall seems to have become more sporadic and intense, making the effects of rain noise on the interior environment an acoustical issue of intensifying importance. Noise produced within architectural spaces by rain on roofs is difficult to quantify due to varying rain intensity, water droplet size, droplet velocity, roof construction, and interior acoustical characteristics. However, it is possible to compare interior rain noise levels when rain conditions and the interior space conditions are constant. This paper compares (1) rain noise levels for several commercial roof constructions with and without suspended ceilings beneath the roof, (2) rain noise levels with roofs using typical rigid foam insulation versus mineral wool insulation. Noise level measurements and rain generation are in general conformance with ISO 140-18 "Laboratory measurement of sound generated by rainfall on building elements." The rain noise levels are presented along with similar rain noise measurements made in the same test facility that were the subject of a 2007 ASA paper. This research was conducted through the School of Architecture, Design and Planning at the University of Kansas.

8:35

5aAA3. Design, construction, and evaluation of a binaural dummy head. Maryam Landi, Vahid Naderyan (Dept. of Phys. and Astronomy & National Ctr. for Physical Acoust., Univ. of MS, NCPA, 145 Hill Dr., University, MS 38677, mlandi@go.olemiss.edu), and David S. Woolworth (Roland, Woolworth & Assoc., Oxford, MS, Oxford, MS)

Binaural-dummy-heads are often used as standard measurement devices where modeling of the human binaural hearing system is desired. The binaural-dummy-head imitates a human head (and torso) which is used in binaural recording as well as research areas such as hearing aids, sound localization, noise measurements, etc. Commercially available binaural heads are not economically efficient for some purposes. This paper outlines a less expensive binaural dummy head built using ANSI/ASA S3.36-2012 standard as a reference as part of an independent coursework. A hard plastic mannequin was used as head and torso, and the two ears were real human ear replicas casted out of water-based alginate gel. The complex Head-Related Transfer Functions (HRTF) of our dummy head were measured in an anechoic chamber to evaluate its spectral and directional properties and were compared to the same properties of the standard commercial dummy head.

8:50

5aAA4. ISO 717-1 introduction in Russia. Ilya E. Tsukernikov, Igor Shubin (Acoust. Lab., Res. Inst. of Bldg. Phys., Odoevskogo proezd, h.7, korp. 2, fl. 179, 21 Lokomotovny pr., Moscow 117574, Russian Federation, 3342488@mail.ru), Tatiana Nevenchannaya (Moscow State Univ. of Printing Arts, Moscow, Russian Federation), and Natalia Schurova (Acoust. Lab., Res. Inst. of Bldg. Phys., Moscow, Russian Federation)

Two single-number quantities are used now in Russia: weighted sound reduction index for rating air noise insulation by internal protecting designs of buildings and the quantity, characterizing sound insulation of external transparent protecting designs from noise, created by municipal transportation streams. For other sound insulation characteristics determined by International Standards ISO 10140-2, ISO 140-4, and ISO 140-5, the corresponding single-number quantities are not established in the Russian standard documents. Besides methods for determination of single-number quantities to be used differ by their various physical senses too. It causes expediency of introduction of International Standard ISO 717-1 in which the corresponding single-number quantities are entered for all spectral characteristics, put into practice of air noise insulation by building protecting designs, and universal methods of their determination are stated. In this paper, features of introduction of International Standard ISO 717-1 in Russia are considered. Comparison of the A-weighted noise spectra for various categories of the railway transportation, to be maintained on the Russian railways, to the sound level spectra, applied in the International Standard to calculate the spectral adaptation terms. By the concrete examples, the divergences in values of spectral adaptation terms received are shown and the corresponding recommendations are offered.

9:05

5aAA5. Characterization of ensemble rehearsal experiences at Brigham Young University. Kieren H. Smith, Tracianne B. Neilsen, Michael H. Denison, and Jeremy Grimshaw (Brigham Young Univ., Provo, UT 84602, kierenhs@gmail.com)

Musicians' ears are barraged with large quantities of sound almost constantly, a reality that is either augmented or diminished by room environments in which musicians practice and perform. In order to make recommendations for future renovations, the acoustics of ensemble rehearsal spaces within the Brigham Young University School of Music were measured. To quantify sound exposure during rehearsals, noise dosages and sound levels experienced by the musicians were measured in various positions in each of two major practice spaces. Measurements were taken during several two-hour rehearsals for major orchestras and band ensembles at BYU. Using data collected from noise dosimeters, spatial maps indicate the noise dosage throughout the rooms. Maximum and average sound levels experienced during the space of a rehearsal also offer insights into the sound environment. This data indicate which areas of the ensemble experience the greatest noise exposure. Reverberation time measurements taken within the rooms further illuminate potential acoustic deficiencies within the room. This preliminary noise environment and dosage highlights the need for future facility renovations and offers suggestions for short term acoustical treatments. EASE acoustic simulation software was used to determine the effectiveness of suggested renovations.

9:20

5aAA6. Reverberation theory obscures real physics in concert halls. James B. Lee (None, 6016 S. E. Mitchell, Portland, OR 97206, cadwal@macforcego.com)

Wallace Sabine posited sound in a room to be a uniform field, in equilibrium, varying but slowly with respect to time required to traverse the space: the "reverberant" field. It easy to demonstrate that such is not so, especially in rooms like occupied concert halls. But then, Sabine designed the Boston Symphony Hall, a paradigm of acoustic excellence. If that hall be regarded as a physics experiment, never has it been replicated by Sabine's followers, even with extensive emendations to his concept of reverberation. The real physics of concert halls involves non-equilibrium manifestation of physical acoustics with respect to bounding surfaces, particularly proximity effects on stage and resonant scattering about the audience.

9:35

5aAA7. Evaluating hospital quiet time from engineering, medical, and nursing perspectives. Jonathan R. Weber, Erica E. Ryherd (Durham School of Architectural Eng. & Construction, Univ. of Nebraska-Lincoln, 1110 S. 67th St., Omaha, NE 68182-0816, jonryanweber@gmail.com), Ashley Darcy Mahoney (Nell Hodgson Woodruff School of Nursing, Emory Univ., Atlanta, GA), Myra Rolfes, Heather Cooper (Neonatal Intensive Care Unit, Children's Healthcare of Atlanta, Atlanta, GA), and Brooke Cherven (Nursing Res. and Evidence Based Practice, Children's Healthcare of Atlanta, Atlanta, GA)

A healthy hospital soundscape is crucial to promote healing for patients and a healthy workplace for staff. Unfortunately, the occupant-generated sounds, building systems, and medical equipment required to care for patients can create high noise levels. Rising concern has led to an increase in hospital noise studies exploring noise reduction strategies. One major issue is the gap existing between the acoustical and medical contributions necessary to solving the problem. This talk will document how our team is focusing on bridging that gap through the interdisciplinary collaboration of individuals from architectural engineering, medicine, nursing, psychology, and statistics. The project includes an 18-month longitudinal study aiming to improve Neonatal Intensive Care Unit (NICU) soundscapes through the implementation of a Quiet Time (QT) evidence-based practice change. Detailed acoustic measurements and staff surveys were collected to document the objective and subjective effects of QT. The acoustical impact of QT on the soundscape and its occupants is currently being explored through the engineering and medical perspectives. In ongoing phases, infant physiological data are being analyzed to understand the infants' response to the altered soundscape resulting from QT. The collaborative efforts required to plan, execute, and evaluate this type of interdisciplinary study will be discussed.

9:50

5aAA8. Death by alarm: An error model of hospital alarms. Ilene Busch-Vishniac (School of Nursing, Johns Hopkins Univ., 200 Westway, Baltimore, MD 21212, buschvi@gmail.com)

On any given day in the United States, there are about 480,000 patients in hospital for reasons other than psychiatric care or rehabilitation, each generating, on average, about 135 clinical alarms per day. Studies have shown that over 90% of these alarms result in no action being taken. Alarm errors, either alarms that sound and receive no response or alarms that fail to sound when they should, number roughly 8 million per day yet data on adverse alarm impacts indicate about 200 alarm-related deaths per year and a total of a little more than 500 adverse impacts per year. A compelling conclusion from this data is that clinical alarms in hospital are very inefficient and ineffective tools for monitoring medical emergencies. Much attention has been dedicated to alarms recently, with the general goal of improving response to alarms in order to ensure no medical emergency is missed. While this work is of immediate use and is vitally important to the operation of the modern hospital, it focuses on minor changes to the existing systems rather than on trying to design the optimum system for the future. It is future alarm systems that we consider here, with an aim of designing a more effective and efficient system for use in hospitals in roughly 20 years.

10:05–10:20 Break

10:20

5aAA9. A method to evaluate nonideal effects of anechoic chambers on multiple-angle measurements. Michael H. Denison, K. J. Bodon, and Timothy W. Leishman (Phys., Brigham Young Univ., 485 S State St., Apt. 306, Provo, UT 84606, michael.denison23@gmail.com)

Anechoic chambers are typically qualified by comparing sound pressures at several radial distances from a sound source and verifying that they follow the spherical spreading law within specified tolerances. While this technique is useful, it may not sufficiently characterize free-field variations at fixed radial distances and numerous angular positions, as are commonly used for directivity, sound power, and other important acoustical measurements. This paper discusses a technique to detect angular field deviations in anechoic chambers. It incorporates a loudspeaker in an altazimuth mount, an adjustable-radius boom arm, and a precision microphone. The boom arm and microphone remain in line with the loudspeaker driver axis at a fixed radius while the system rotates to specified polar or azimuthal angle increments. In an ideal free-field environment, the frequency response function from the loudspeaker input to the microphone output should remain consistent—regardless of the system orientation. However, in typical anechoic chambers, they vary. Standard deviation calculations over many angles reveal frequency-dependent departures from the ideal, especially for narrow-band data. The results show the impact of these discrepancies for multiple-angle measurements and how they change with radial distance from the source.

10:35

5aAA10. Unexpected challenges of luxury residential building acoustics. Marlund E. Hale (Adv. Eng. Acoust., 663 Bristol Ave., Simi Valley, CA 93065, mehale@aol.com)

Building acoustics involving sound enhancements, noise abatement, and vibration control for luxury residential properties can present more than just technical, performance, and mitigation issues. The normal noise control issues dealing with mechanical, electrical, and plumbing (MEP) systems are common in commercial and multi-family residential projects. However, ultra-expensive, highly customized, single-family residences are a very different situation, both from the expectation of quality and the much larger and more varied types of noise producing systems. Such custom homes of entertainment and movie stars, corporate moguls, and sports personalities include features and systems that would never occur in the typical single-family home. Large in-home theaters with fully outfitted sound systems, indoor swimming pools, large entertainment centers, dance and reception halls, etc., present very challenging acoustical and noise control challenges. Many such estates also have commercial or industrial type mechanical systems, multi-fueled emergency power generators, and very large water storage and pumping systems that generate significant noise control issues both within the owner's property and for adjacent neighboring properties. This paper summarizes a few such challenging projects and reviews the often unusual situations that occur during concept and project design, specification and bidding phases, construction, redesign, change-orders, and essential inspections.

10:50

5aAA11. Acoustical evaluation of residential laneways and laneway housing. Rosa Lin and Maureen Connelly (School of Construction and the Environment, Br. Columbia Inst. of Technol., 3700 Willingdon Ave., NE03, Burnaby, BC V5G 3H2, Canada, rlin35@bcit.ca)

Laneway housing (LWH), a free-standing, small wood-frame house (1 to 1.5 stories high and less than 900sf in floor area), is an increasingly popular product of Vancouver B.C.'s high-density urban growth policy. LWH is subject to multiple noise concerns due to its small architectural form and siting in laneways typically designed for garage access and utilities. (Laneways are approximately 16 ft wide and 12 to 18 ft high.) Case study investigations examined residential laneways as potential urban canyons and the acoustical performance of LWH envelope. Methods employed include ASTM field measurements, software models (Odeon and AFMG Soundflow), and the National Research Council of Canada traffic noise model (comparable to the FHWA model). Metrics evaluated include rate of attenuation over propagation distance through a laneway, façade transmission loss, and room absorption. Results show that certain laneways function as urban canyons, measuring nearly 20 dBA higher in SPL at the LWH façade than in laneways without urban canyon characteristics. Indoor SPL due to traffic noise transmitting through the building envelope were above 45 dBA. At least half of the case studies investigated did not meet noise criteria for residential health. These investigations emphasize the need for acoustical specifications in design and construction guidelines for LWH.

11:05

5aAA12. A novel method for perceptual assessment of small room acoustics using rapid sensory analysis. Neofytos Kaplanis (Bang & Olufsen, Peters Bang vej 15, Struer, Denmark, neo@bang-olufsen.dk), Søren Bech (Electron. Systems, Aalborg Univ., Struer, Denmark), Tapio Lokki (Dept. of Media Technol., Aalto Univ., Helsinki, Finland), Toon van Waterschoot (Elec. Eng., ESAT-STADIUS/ETC, KU Leuven, Leuven, Belgium), and Søren H. Jensen (Electron. Systems, Aalborg Univ., Aalborg, Denmark)

Identifying and perceptually characterizing the physical properties of rooms is a fundamental step in understanding the acoustical qualities of a space. Over the last century, numerous studies have investigated the perceptual qualities in performance spaces, such as opera houses and concert halls. In smaller spaces, such as domestic environments, the research focus has been primarily steered toward sound reproduction within a room, rather than the transmission medium, the room. In this study, a new methodology is used to perceptually assess and characterize a range of acoustical properties within small rooms and car cabins. In-situ measurements were performed to obtain a range of possible acoustical settings, by varying physically the spaces under investigation. The measured responses were spatially analyzed and synthesized to reproduce the observed fields in the laboratory. Expert listeners were presented with auralized sound over a loudspeaker array and followed a rapid sensory analysis protocol. The elicited attributes and ratings are analyzed and possible links to the acoustical properties of these spaces are discussed. [*This study is a part of Marie Curie Network on Dereverberation and Reverberation of Audio, Music, and Speech. EU-FP7 under agreement ITN-GA-2012-316969.]*

11:20

5aAA13. Investigation of coprime scattering arrays as sparse sound diffusers. Kevin A. Melsert, Dane Bush, and Ning Xiang (Architectural Acoust., Rensselaer Polytechnic Inst., 110 8th St., Troy, NY 12180, melsek@rpi.edu)

Number-theoretic coprime sparse samplers have recently sparked research activities in the fields of signal processing and acoustics. The concept of coprime sparse sensing has been successfully implemented and experimentally tested in form of acoustic sparse microphone arrays, as shown in the publication [D. Bush and N. Xiang, J. Acoust. Soc. Am. 138, 447-456 (2015)]. The current study investigates how coprime theory can be applied to scattering incident sound. It also investigates how coprime array concepts can be applied to volumetric diffusers using solid cylinders as sparse array elements. These are arranged in pairs of subarrays whose number of cylinders are two mutually prime (coprime) integers in a sparse arrangement. Coprime scatter arrays are tested experimentally as sparse sound diffusers using one-fifth scale modeling and an acoustic goniometer to investigate coprime sparse diffuser effectiveness in real world applications.

11:35

5aAA14. Empirical measure of absorption and scattering properties of living wall plants and systems and predictive modeling of room acoustic benefits. Maureen R. Connelly, Daver Bolbolan, Masha Akbarnaejad, and Sepideh Daneshpanah (Construction and Environment, BCIT, 3700 Willingdon Ave., Bldg. NE03 Office 107, Burnaby, Vancouver, BC V5G3H2, Canada, maureen_connelly@bcit.ca)

This series of research projects investigate the acoustical characteristics of interior living walls and predicts how they can be used to positively benefit room acoustics. Scaled and full scale evaluations were executed in a reverberation chamber to validate test methods for absorption and scattering coefficients of soil/substrate and plant species (characterized by height, stem diameter, mass, leaf geometries and dimensions, and leaf area index (LAI)). Wall systems were evaluated over a gradient of plant coverage with monoculture and community planting. Evaluation indicates that evenly distributed pumice can act as the baseline on the scattering turn-table in a method to evaluate scattering coefficients of plant-specific foliage. Findings indicate that percentage of plant coverage is related to absorption coefficients (0.16–1.1) as averaged across all evaluated species. Only at low plant coverage do specific plant characteristics affect the absorption coefficients. The percentage of plant coverage is related to scattering coefficients (0.05–0.51) at 500 Hz and higher. LAI x Mass predicts absorption and scattering coefficients at mid-frequency (200–2500 Hz). Comparison of prediction and field studies identify that use of scattering coefficients improves the prediction of the beneficial use of living walls in room acoustics.

11:50

5aAA15. A hybrid method combining the edge source integral equation and the boundary element method for scattering problems. Sara R. Martín Román, U. Peter Svensson (Acoust. Res. Ctr., Dept. of Electronics and TeleCommun., Norwegian Univ. of Sci. and Technol., O.S. Bragstads plass 2a, Trondheim 7034, Norway, sara.martin@ntnu.no), Jan Šlechta (Univ. Ctr. for Energy Efficient Buildings, Czech Tech. Univ. in Prague, Bustehrad, Czech Republic), and Julius O. Smith (Ctr. for Comput. Res. in Music and Acoust. (CCRMA), Stanford Univ., Stanford, CA)

A hybrid method for acoustic scattering problems is studied in this paper. The boundary element method is combined with a recently developed edge diffraction based method [J. Acoust. Soc. Am. **133**, pp. 3681–3691, 2013]. Although the edge diffraction method has been shown to provide accurate results for convex, rigid objects at a very attractive computational cost, it has some numerical challenges for certain radiation directions. The hybrid method suggested here has a similar structure as the boundary element method (BEM): in a first step, the sound pressure is calculated on the surface of the scattering object, and in a second step, the scattered sound is obtained at any external receiver point. In this method, the edge diffraction based method is used for the first step, and then, the calculation of the scattered sound is performed à la BEM by means of the Kirkhoff–Helmholtz Integral equation. Several benchmark cases are studied, and the results are compared with different reference methods.

Session 5aMU

Musical Acoustics: General Topics in Musical Acoustics

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Contributed Papers

8:00

5aMU1. Real-time three-dimensional tongue motion during clarinet performance. Steven M. Lulich (Speech and Hearing Sci., Indiana Univ., 200 South Jordan Ave., Bloomington, IN 47405, slulich@indiana.edu), Sherman Charles (Linguist, Indiana Univ., Bloomington, IN), and Benjamin Lulich (The Cleveland Orchestra, Cleveland, OH)

Clarinetists teach and feel that they manipulate their tongue shape and position in order to properly "voice" music with ideal intonation and timbre, as well as for special effects such as portamento (pitch bending) and glissando. Two basic postures are typically described, which are frequently called the "ee" (or "er") and "ah" positions, or "voicings," due to their presumed similarity to tongue shapes during production of the speech sounds [i] (or [r]) and [a]. Although some two-dimensional imaging studies of tongue shape during clarinet performance have been reported, three-dimensional (3D) data have not been previously available. This study presents 3D tongue motions during performance by one professional clarinetist. Tongue images were acquired from a 3D/4D ultrasound system with synchronized audio recordings, and the images were aligned with a digitized impression of the performer's palate. Analyses of the data are ongoing, but initial results indicate that the "ee" and "ah" postures are based on the tongue shape but not on the tongue position. Further aspects of the tongue shape and position during clarinet "voicing" will also be discussed.

8:15

5aMU2. Resonant mode characteristics of a cajón drum and its effect on sound directivity. Michael H. Denison, K. J. Bodon, and Kent L. Gee (Phys., Brigham Young Univ., 485 S State St., Apt. 306, Provo, UT 84606, michael.denison23@gmail.com)

The cajón is a hand percussion instrument originally from Peru. Spanish for "large box" or "crate," the cajón is typically a box enclosure with a thin wooden plate at the front that serves as the playing surface. Many resonance modes are excited when a player strikes the front plate, as recently studied by Pehmoeller and Ludwidgsen [J. Acoust. Soc. Am. **138**, 1935 (2015)]. In this study, scanning laser Doppler vibrometry is coupled with high-resolution far-field directivity measurements to examine source vibration and the resultant sound field. Ties between the plate modal vibration and far-field sound radiation spatial patterns are shown. In particular, the non-uniform front plate normal velocities at higher-order resonances result in distinct nulls in the far-field directivity. Avoidance of these nulls yields recommended microphone locations for cajón sound amplification.

8:30

5aMU3. Investigating the effect of body geometry on the acoustics of electric guitars. Mark Rau and Gary Scavone (Music, CAML - McGill Univ., 2325 Maisonneuve West, Apt. 2, Montréal, QC H3H1L6, Canada, mark.rau@mail.mcgill.ca)

Three electric guitars of different body geometries are investigated in an attempt to characterize their tonal differences. One guitar has a solid body, one is fully hollow, and one is semi-hollow which represents a midway point between the first two. The transfer functions of the electromagnetic pickups are determined by inducing a known signal through the pickup and measuring the output. Input admittance measurements are taken at the bridge and nut of each guitar to show the vibrational modes of the bodies. Wire break measurements are taken for different notes on each guitar, which are analyzed in conjunction with the admittance measurements. The results demonstrate significant admittance peaks in the hollow and semi-hollow guitars that are not present in solid body electric guitars. As well, "dead" notes with shorter decay times are found to be correlated with these admittance peaks, thus indicating that the bodies of hollow and semi-hollow electric guitars play a role in their tonal characteristics.

8:45

5aMU4. Instrument identification and blending in vinyl records during the transition period from jazz to rock music. Jonas Braasch (School of Architecture, Rensselaer Polytechnic Inst., 110 8th St., Troy, NY 12180, braasj@rpi.edu)

Arnold Schering and others have partitioned music genres into traditions where the identification of individual instruments (Spaltklang) is the goal and others where instruments are blended to a homogeneous sound (Verschmelzung). Baroque music is often cited to emphasize the identifiability of individual instruments, whereas the fusion of orchestral instruments became one of the major goals in the romantic period. Historical vinyl record releases were used to investigate the extent to which the theory of identification and blending can be applied to 20th century popular music based on the fundamental principles of Auditory Scene Analysis. The identifiability of instruments was one of the major concerns for jazz combos, for example, in the Miles Davis sextet of the late fifties. Blending became more important in many types of rock music to create a huge sound, for example, in Phil Spector's Wall of Sound of the 1960s. Artificial reverberation, dynamic range compression, and other effects were now used to create loud and fused sound fields. Based on a psychophysical loudness model, it will be discussed how both approaches lend themselves to create loud sounding mixes to maximize the achieved perceived loudness for a given medium-dependent maximum signal level.

9:00

5aMU5. Vibration analysis of acoustic guitar string employing high-speed video cameras. Bozena Kostek (AudioAcoust. Lab., Gdansk Univ. of Technol., Natowicza, 11/12, Gdansk, Pomorskie 81-233, Poland, bokostek@audioacoustics.org), Piotr Szczuko, Jozef Kotus, Maciej Szczodrak, and Andrzej Czyzewski (Multimedia Systems Dept., Gdansk Univ. of Technol., Gdansk, Pomorskie, Poland)

A method of analysis and visualization of displacements of an acoustic guitar string is presented. Vibrations of the strings are recorded using high-speed cameras. The optical system used for the recording is applied in order to make it possible to observe the vibrations along the string. Images recorded with high-speed cameras are analyzed using digital signal processing algorithms in order to track the shape of deflections and displacement of strings, with a high spatial resolution, and to convert the acquired video data into an acoustic signal. The acoustic signal derived from the visual analysis is then compared with a reference signal which was recorded simultaneously using a measurement microphone. The research experiments are aimed principally at studying the phenomena related to energy transfer of vibrating strings to the body of the instrument. [This research study was supported by the grant, funded by the Polish National Science Centre, decision number DEC-2012/05/B/ST7/02151.]

9:15

5aMU6. A physical model of a highly nonlinear string and its use in the music composition *Quartet for Strings*. Edgar Berdahl, Stephen D. Beck, and Andrew Pfalz (Music, Louisiana State Univ., 102 New Music Bldg., Baton Rouge, LA 70803, eberdahl@ccrma.stanford.edu)

A real-time virtual string model is created that supports two modes of haptic force-feedback interaction. Using one haptic controller, the string can be plucked, and using a second haptic controller, the virtual string's pitch can be continuously varied by applying pressure to the string. Via both controllers, the vibrations of the virtual string can be felt while it is played. These functionalities are achieved by modeling the string as a finite sequence of interleaved masses and stiffening springs. For artistic purposes, a nonlinear spring force characteristic function is selected that limits the effective stiffness of each spring to a range between k and $(k + k_s)$. This function is $F(x) = kx + (k_s x^3)/(\beta^2 + x> + x^2)$, where x is the displacement of the nonlinear spring and β approximately adjusts the displacement at which the stiffness becomes significantly nonlinear. For small displacements, the string's undamped fundamental frequency is tuned by k, and for sufficiently large displacements, the string's undamped fundamental frequency is approximately tuned by $(k + k_s)$. The composition *Quartet for Strings* features the model tuned to four different scales (treble 1, treble 2, alto, and bass), each of which is played by a human performer according to the score of *Quartet for Strings*. For this composition, $(k + k_s)$ is set so that when the string is vibrated at sufficiently large displacements, the pitch increases by approximately an octave but not precisely an octave, in order to provide for a more complex pitch space.

9:30

5aMU7. Predicting the acoustical properties of 3d printed resonators using a matrix of impulse responses and mode interpolation. Romain Michon (Dept. of Music, Ctr. for Comput. Res. in Music and Acoust., Stanford Univ., Stanford, CA 94305-8180, rmichon@ccrma.stanford.edu) and John Granzow (School of Music, Theatre & Dance, Univ. of Michigan, Ann Arbor, MI)

Accurately predicting acoustical properties of 3D printed models is of interest to instrument designers who explore novel geometries. We introduce a technique to carry out these estimates using a database of impulse responses and mode interpolation. 3D models are organized as a function of their physical characteristics and placed into a multidimensional space/matrix. The models at the boundaries of this space define the limits of our prediction algorithm and they are produced using 3D printing. Impulse responses of these models are measured, and modal information is extracted from each object. Mode parameters are interpolated within the matrix to predict the frequency response of unprinted models that fall within the geometrical space of the test matrix. A physical model using modal synthesis also allows us to listen to the resulting resonator.

9:45

5aMU8. Physical modeling sound synthesis using embedded computers: More masses for the masses. Edgar Berdahl and Matthew Blessing (Music, Louisiana State Univ., 102 New Music Bldg., Baton Rouge, LA 70803, edgarberdahl@lsu.edu)

Physical modeling for sound synthesis is a technique in which musical acoustic equations are simulated by computer to synthesize sound. In prior decades, either offline simulation or powerful desktop or laptop computers were required in order to synthesize high-quality sound. However, increasingly small and relatively low power embedded computers are presently becoming available that can natively perform real-time simulations using floating-point computations. For example, the Raspberry Pi 2 is an embedded computer, which incorporates a quad-core 1GHz embedded microprocessor, and currently costs only US$35. This implies that physical modeling sound synthesis may become accessible to a wide range of people for many diverse applications. Furthermore, larger and larger numbers of virtual masses will be computable in real time. A poster with embedded Raspberry Pi 2 and amplified speaker is presented that uses Synth-A-Modeler to simulate a wide variety of physical models in real time.

Session 5aNS

Noise: Topics in Noise Control

Blaine M. Harker, Chair
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Contributed Papers

8:30

5aNS1. Aircraft carrier noise measurements of a high-performance fighter jet. Alan T. Wall, Richard L. McKinley, Michael R. Sedillo, Billy J. Swayne (Battlespace Acoust. Branch, Air Force Res. Lab., Bldg. 441, Wright-Patterson AFB, OH 45433, alantwall@gmail.com), Michael J. Smith, Allan C. Aubert (NAVAIR, Pax River Naval Base, MD), Robert E. Nantz, and Gregory J. Imhof (Joint Strike Fighter Integrated Test Force, Pax River Naval Base, MD)

The on-deck measurement of F-35C noise levels occurred during the DT-II sea trials aboard the USS Dwight D. Eisenhower in October 2015. The existence of aircraft carrier flight deck fighter noise data is extremely rare, with this data set being only the second of its kind in terms of its scope. Custom acoustic recording instrumentation was designed to obtain quality broadband noise measurements of high-amplitude signals, protected from extraneous noise due to high wind speeds, and shielded from intense electromagnetic interference from the multiple on-board radar systems. The data collected allow for the estimation of noise exposures at all pertinent flight deck locations where crewmembers are positioned. [Work supported by USAFRL through ORISE and F-35 JPO.]

8:45

5aNS2. Review study of main sources of noise generation at wheel-rail interaction. Qinan Li and Mohammad Mehdi Alemi (Mech. Eng., Virginia Tech, 1100 Houndschase Ln., Unit G, Blacksburg, VA 24060, liq001@vt.edu)

Railways are very economical and efficient long-distance transportation systems in many high-density populated countries. However, in terms of environmental safety, different countries announced an increasing traffic noise near residential area. The relative importance of wheel-rail noise to overall train noise plays a crucial role in developing railway transportation. Generally, there are three types of noise at wheel-rail interaction: rolling noise, impact noise, and squeal noise. Rolling noise refers to vertical vibration excitation on straight track due to the undulations of the wheel and rail surfaces. TWINS as the most advanced model up to date can provide total noise prediction within 2 dB compared to full-scale experiments. The impact noise happens when wheel is running on the rail surface discontinuities. Since linear noise generation assumption is employed at contact patch and non-linear contact element is ignored, impact noise due to large roughness are rarely developed and they need to be extended to the reasonable higher frequencies (5 kHz). The squeal noise is usually due to lateral excitation mechanism and always occurs on sharp radius curves. Due to the complexity of wheel-rail noise generation mechanisms, the almost all existing prediction models either have not been validated or they are bounded to predict noise under limited conditions and certain frequency ranges. In this paper, the differences between three main sources of noise generation at wheel-rail interaction will be reviewed.

9:00

5aNS3. A study on minimizing the noise from a trailing edge by the use of optimized serrations. Matt Brezina and Joana Rocha (Mech. and Aerosp. Eng., Carleton Univ., 1125 Colonel By Dr., Ottawa, ON K1S 5B6, Canada, matt.brezina@carleton.ca)

An optimization study is presented in which the noise produced by flow over the trailing edge of a flat plate is minimized via serrations. The theoretical optimum configuration is found by changing the geometry of the serrations with a non-linear gradient based algorithm. The theoretical noise produced by the trailing edge is determined using the semi-empirical model developed by Howe. As expected, the configuration of the trailing edge serrations that produced the least noise over all frequencies (from 20 Hz to 20 kHz) was found to be those with the smallest width (approaching zero) and largest root-to-tip distance (up to the assigned upper-bound). The optimum serration geometry for individual frequencies was found to be highly dependent on the upper bound given to the height. In addition to the optimization study, an analysis is presented of a method for modifying the noise model to capture the high-frequency noise increase (relative to a straight trailing edge) that is regularly observed in experiments. Results are presented for model validation by a comparison to the sound pressure level frequency spectrum produced experimentally.

9:15

5aNS4. Modeling systems of acoustic resonators for application in passive noise control. Matthew F. Calton and Scott D. Sommerfeldt (Brigham Young Univ., 560 E 400 N APT 4, Provo, UT 84606, mattcalton@gmail.com)

Acoustic resonators, such as Helmholtz and quarter wave resonators, are commonly used as narrowband attenuators in small enclosures. The widespread use of resonators has led to many analytical expressions and approximations for their response, especially for the ideal geometry case. While these resonators are limited in bandwidth, systems of resonators can be designed to obtain responses not otherwise attainable by a single element. Additionally, practical applications of these systems often require non-ideal geometries to be implemented in the design process. This research aims to incorporate equivalent circuit techniques to more accurately characterize the input impedance of a system of resonators. Using the calculated input impedance of the system, an optimal coupling location is determined using a model of the coupled source and enclosure. These calculations are compared to experimental results for validation.

9:30

5aNS5. Active control of noise radiated from an x-ray tube. Yin Cao, Kelli Succo, Scott D. Sommerfeldt (Dept. of Phys. and Astronomy, Brigham Young Univ., N203 ESC, BYU, Provo, UT 84602, kelli.fredrickson7@gmail.com), and Jonathan D. Blotter (Dept. of Mech. Eng., Brigham Young Univ., Provo, UT)

The application of implementing an active noise control system to globally attenuate noise radiated from a medical x-ray tube has been investigated. The noise radiated from the x-ray tube is characterized by the presence of numerous tonal peaks distributed over a broad frequency bandwidth. Furthermore, there is sufficient variability in the radiated field that

coherence in the acoustic field was also found to be a challenge. It was determined that a properly placed structural sensor could be used as a reference signal to achieve good coherence between the reference signal and error microphone. In order to achieve global control, speakers were placed in close proximity to areas that had been identified as source regions through the use of SLDV and acoustic intensity measurements. The frequency band below 1500 Hz was targeted, and it was found that effective attenuation could be achieved at over ten of the most prominent frequencies, resulting in attenuations in that bandwidth on the order of 7 dB. Some of the challenges encountered and results obtained will be discussed and presented.

9:45

5aNS6. Active control of a finite length line source by a novel secondary source. Qi Hu and Shiu-keung Tang (Dept. of Bldg. Services Eng., The Hong Kong Polytechnic Univ., ZS801, Block Z, Hung Hom Na, Hong Kong, qi.bs.hu@connect.polyu.hk)

Active control of noise in free space tends to be challenging, especially for the overall attenuation. This paper focuses on the control of a finite line noise source using active approach. Previous study by the author reveals that the introduction of a directional source, an axially oscillating circular piston, as the secondary control source will improve the overall performance of the active noise control system. A novel directional source is theoretically studied in this work, which consists of a core piston and an outer concentric annulus. The two components are axially oscillating with different phases and amplitudes, and these parameters are optimized to maximize the source directivity. This structure can be further extended to add more outer annuluses, which also could be analytically analyzed of the optimal parameters according to the similarity of the construction components. This exquisitely novel control source is adjusted to radiate a sound field analogous to the primary line source with multi-lobe property, and its physical size is restricted within a practical dimension.

10:00

5aNS7. Development of a direct aerodynamic drag measurement system for acoustic liners. Christopher Jasinski (Univ. of Notre Dame, 54162 Ironwood Rd., South Bend, IN 46635, chrismjasinski@gmail.com)

The objective of this paper is to establish the validity of a measurement apparatus for determining aerodynamic drag caused by acoustic liners. Conventional acoustic liners for ducted turbofan engine nacelles have been proven to reliably reduce engine noise in commercial aircraft for a narrow frequency range. Traditional acoustic liners consist of a porous facesheet, honeycomb core, and solid backing. As technology has developed, new liner designs, including variable-depth cores, show promise of reducing noise in a broad frequency range. Additionally, the next generation of aircraft design may allow for additional aircraft surface area to be covered by acoustic liners, further reducing aircraft noise perceived on the ground. Before these advanced acoustic technologies can make their way onto commercial aircraft fleets, the aerodynamic drag caused by the liners must be understood. Using the Mach 0.6 Wind Tunnel at the University of Notre Dame, a direct drag measurement system has been developed utilizing a linear force balance. In combination with indirect drag measurements taken at NASA Langley's Curved Duct Test Rig, measurement confidence has been established. This paper will detail the capabilities of the direct drag measurement system at Notre Dame and explore its intended future use in developing new liner technologies.

FRIDAY MORNING, 27 MAY 2016 — SALON H, 8:00 A.M. TO 12:15 P.M.

Session 5aPA

Physical Acoustics: General Topics in Physical Acoustics II

Michael B. Muhlestein, Chair
Mechanical Engineering, University of Texas at Austin, 3201 Duval Rd. #928, Austin, TX 78759

Contributed Papers

8:00

5aPA1. Acoustic hysteresis of varying cavity length of a bottle-shaped thermoacoustic prime mover with a neck to cavity ratio of 1:10. Emily Jensen and Bonnie Andersen (Phys., Utah Valley Univ., MS 179, 800 W University Pkwy, Orem, UT 84058, em.jensen88@gmail.com)

A previous study showed hysteresis in transition regions to overtones of bottle-shaped thermoacoustic prime movers with a neck to cavity diameter ratio of 1:2.4 while varying the cavity length. Hysteresis regions were studied with a neck to cavity diameter ratio of 1:10. The device consisted of a neck (5.15 cm long, 0.75" ID) with a heating element around it and a cavity (ID 3.75") with a sliding piston, allowing the cavity length to be varied up to 38 cm long. Copper mesh was used for the heat exchangers and were located about 30{\%} away from the top of the neck and 16 mg of steel wool served as the stack. A pressure sensor was connected to the center of the piston to measure acoustic pressure at the bottom of the cavity. Acoustic pressure, frequency, and hot and cold temperatures were recorded while both increasing and decreasing the cavity length from about 2 to 38 cm in increments of 0.2 cm over time intervals of 20 s with an input power of 10.5 W. Three transitions to overtones occurred at 6.8, 15.2, and 27.4 cm while pulling the piston out and at about 25.2, 15.6, and 6.0 cm while pushing the piston in. Frequencies and transition regions agreed with expected values. Both hot temperature and acoustic pressure increased during transition regions. This could be caused from multiple acoustic waves being produced.

8:15

5aPA2. Experimental demonstration of mitigating flame-sustained thermoacoustic oscillations by using an electrical heater. Dan Zhao and Xinyan Li (Aerosp. Eng. Div., Nanyang Technolog. Univ., 50 Nanyang Ave., Singapore, Singapore 639798, Singapore, zhaodan@ntu.edu.sg)

In this work, experimental investigation is conducted to mitigate flame-sustained thermoacoustic instability occurred in a conventional Rijke-type and bifurcating thermoacoustic systems by using an electrical heater. A premixed methane-burned flame is confined in the bottom part of the Rijke-type and bifurcating thermoacoustic systems to produce self-sustained thermoacoustic oscillations. As the electrical heater is placed downstream of the flame, it acts like a sound absorber and can even completely stabilize the

unstable thermoacoustic system. The damping performance of the heater depends strongly on its axial location and output power. To further validate our findings, experimental investigation of mitigating thermoacoustic oscillations in a bifurcating unstable system by using a heater is performed. It has a mother tube with a premixed methane-burned flame confined. The mother tube splits into two bifurcating branches with an angle of α. A temperature-controllable electrical heater (TCH) is enclosed in one of the bifurcating branches. It is shown that the bifurcating system is associated with thermoacoustic oscillations at around 190 Hz with a sound pressure level of 130 dB. However, when the heater placing at 15 cm away from the open end is actuated, sound pressure level is found to be reduced by approximately 50 dB.

8:30

5aPA3. Using helium as the working fluid in thermoacoustic engines. Nathaniel Wells and Bonnie Andersen (Phys., Utah Valley Univ., MS 179, 800 W University Pkwy, Orem, UT 84058, nateswells@gmail.com)

The efficiency of thermoacoustic engines can be improved by using helium as the working fluid because it reduces viscous losses, has a higher thermal conductivity and speed of sound. The engine in this study has a bottle-shaped resonator. The neck consists of a brass cylinder, closed at the top end and a copper cylinder, open at both ends, with copper mesh screens heat exchangers between them (ID of 1.91 cm and total length of 5.24 cm). A small amount of steel wool (20 mg) functions as the stack. The neck opens into an aluminum cavity (10 cm long with an ID of 4.13 cm). A combination of two types of heat-shrink tubing and Teflon were used to connect the brass and copper pieces. The engine was evacuated of air and backfilled with helium as much as the setup would tolerate. Using an input power of 14.8 W over intervals of 0.5–3 h, it was observed that the frequency decreased in time, indicating that the helium was leaking out slowly. From the frequency data, the volume fraction of helium was calculated, indicating that the engine was able to achieve 64% volume fraction of helium and decreased to 6%. The intensities of the sound over this range of volume fractions averaged at 155 W/m^2 compared with air at an average of 118 W/m^2.

8:45

5aPA4. Entrainment of two thermoacoustic engines. Orest G. Symko, Myra Flitcroft, Brenna Gillman, Ivan Rodriguez, and Cedric C. Wilson (Phys. & Astronomy, Univ. of Utah, 621 S. 1100 E., Salt Lake City, UT, wilson.cedric.c@gmail.com)

In order to build more powerful sources of sound for energy conversion, the synchronization of two thermoacoustic heat engines has been studied. Experiments were performed on engines in the acoustic frequency range of 2.6 kHz and also on very small engines in the ultrasonic range of 24 kHz. In both cases, the engines were mounted on a cylindrical cavity, and they were coupled mainly by the acoustic field in the surrounding air at one atmosphere. They were driven by heaters of resistance wire in contact with the hot heat exchanger. At a specific coupling between the 2.6 kHz, engines' synchronization occurred, and also for the 24 kHz devices; frequency pulling in each pair of engines led to a common frequency in each set, i.e., in-phase synchronization. The strength of the synchronization was determined as a function of detuning of engines. Mutual entrainment was observed at the onset of oscillations and this is attributed to drive by fluctuations. Moreover, as a result of synchronization, the critical temperature gradient for onset of oscillations was reduced from that of the individual values. Delays between two oscillators in the start up led to quenching of the generated sound output (oscillation "death" of Rayleigh).

9:00

5aPA5. Irregular reflection of weak acoustic shock pulses on rigid boundaries. Desjouy Cyril, Sébastien Ollivier (LMFA UMR CNRS 5509, Ecole Centrale de Lyon, Univ. Lyon 1, 36 av Guy de Collongue, Ecully 69134, France, sebastien.ollivier@univ-lyon1.fr), Olivier Marsden, Didier Dragna, Maria Karzova, and Philippe Blanc-Benon

The reflection of weak shockwaves on rigid boundaries at grazing angles was studied in different geometrical configurations. It was shown that even in the case of weak shocks, irregular reflection of N-waves can lead to the formation of a three shocks pattern with a Mach stem, a triple point above the rigid surface and an angle reflection which differs from the incident one. Experiments were done using spark generated spherical shock waves with peak pressure lower than 10 kPa. Reflection patterns of shocks were obtained using a Schlieren visualization setup. Both regular and irregular regimes of reflection were observed. Numerical simulations of the nonlinear propagation were also performed. They were based on the high-order finite difference solution of the two dimensional Navier-Stokes equations. Optical measurements were compared with the results of simulations. Both experimental and numerical results showed the growth of the Mach stem with the distance of propagation. In the case of a randomly or periodically rough surface, the Mach stem is shorter, and nonlinear interactions between reflected waves occurs. [This work was supported by the Labex Centre Lyonnais d'Acoustique of Université de Lyon, operated by the French National Research Agency (ANR-10-LABX-0060/ANR-11-IDEX-0007).]

9:15

5aPA6. Derivative skewness values of shock-containing noise waveforms. Brent O. Reichman, Kent Gee (Brigham Young Univ., 453 E 1980 N, #B, Provo, UT 84604, brent.reichman@byu.edu), Michael James, Alexandria Salton (Blue Ridge Res. and Consulting, LLC, Asheville, NC), and Tracianne B. Neilsen (Brigham Young Univ., Provo, UT)

Acoustic shocks present in military aircraft noise are often referred to as crackle, a component of jet noise that can significantly alter perception and annoyance. Quantifying the shock content within a waveform is important for gauging possible effects that these shocks may have. In the past, the derivative skewness has been used to quantify the steepening of waveforms throughout propagation, representing average behavior throughout the waveform. Though past use has been mostly qualitative, recent work has given physical meaning to derivative skewness values of sinusoids and provided recommendations on sampling rate and thresholds that can indicate significant shocks within the waveform. Using these recommendations, the derivative skewness of high-performance military aircraft noise is estimated for various locations and engine conditions. Derivative skewness values are compared with shock counting schemes based on various criteria. The comparisons show a strong relationship between high derivative skewness values and the strength and number of shocks within a waveform.

9:30

5aPA7. Comparison of rocket launch data to propagation model results. John Noble (Army Res. Lab., 2800 Powder Mill Rd., Adelphi, MD 20783, john.m.noble.civ@mail.mil)

NASA's Wallops Flight Facility launched medium lift rockets for experimental and space station resupply missions during 2013 and 2014. These launches have been a great opportunity to use the rocket-generated infrasound as a repeatable source to study long range propagation. Data from the US Array was used to compare the received amplitude from the rocket launches to a propagation model prediction using realistic atmospheric profiles for ranges out to 500 km from the launch point. The US Array was a distribution of infrasound and seismic sensors which range in a north/south strip across the United States and would periodically relocate further east over time. The results of this comparison will show how well the modeling was able to track the behavior in the measurements.

9:45

5aPA8. Quadspectral nonlinearity analysis of military jet aircraft noise waveforms. Kyle G. Miller (Phys. and Astronomy, Brigham Young Univ., 323 East 1910 South, Orem, UT 84058, kglenmiller@gmail.com), Kent L. Gee, Brent O. Reichman, Tracianne B. Neilsen (Phys. and Astronomy, Brigham Young Univ., Provo, UT), and Michael M. James (Blue Ridge Res. and Consulting, LLC, Asheville, NC)

Understanding the impact of jet noise can be improved by quantifying the nonlinearity in a signal with a single-microphone measurement. Based on the quadspectral Morfey-Howell indicator, a nonlinearity gain factor called ν_N has been derived from an ensemble-averaged, frequency-domain version of the generalized Burgers equation [Miller *et al.*, AIP Conf. Proc. **1685**, 090003 (2015)]. This gain factor gives a quantitative expression for the change in sound pressure level spectrum over distance. Past results show that ν_N accurately characterizes nonlinear evolution of waves in simulation

and model-scale jet data. Here, noise waveforms from a high-performance military jet aircraft are characterized using the ν_N indicator; results are compared with those from other indicators that have been used previously (e.g., derivative skewness, time-waveform steepening factor, etc.). Far field results show that the nonlinear gains at high frequencies (>1 kHz) tend to balance the absorption losses, thus establishing the characteristic $1/f^2$ spectral slope. Additionally, trends over angle, distance, and engine condition are explored. [Work supported by the AFRL SBIR program.]

10:00–10:15 Break

10:15

5aPA9. Modeling of the pressure distribution, acoustic streaming, and formation of bubble structures in an ultrasonic horn reactor. Yezaz A. Gadi Man and Francisco J. Trujillo (School of Chemical Eng., The Univ. of New South Wales, F10 Chemical Sci. Bldg., Kensington Campus, Sydney, NSW 2052, Australia, y.gadiman@unsw.edu.au)

A framework is developed to simulate the acoustic streaming and the formation of conical bubble structures (CBS) in ultrasonic horn reactors. In these reactors, acoustic pressure waves propagate from the vibrating solid horn to the liquid. On its movements away from the horn, sound is attenuated producing acoustic streaming, which is exacerbated by the presence of inertial acoustic bubbles that diminish the pressure amplitude many fold very near to the acoustic source. This is a complex Multiphysics phenomena occurring at different time and length scales. Inertial bubbles, which have sizes of the order of few micrometers, oscillate, grow, and collapse as function of time within the period of the acoustic wave while acoustic streaming is a time independent hydrodynamic movement of the liquid occurring through the totality of the reactor. The four main phenomena, solid vibration, acoustic pressure, streaming and bubble generation and distribution, are simulated in COMSOL Multiphysics in a stepwise fashion. The first two phenomena are coupled and solved in frequency domain, and the resultant acoustic pressure forces are applied to solve turbulent streaming and a transport equation in time-independent mode. Steps are repeated sequentially until a good agreement is found with experimental CBS, power, and streaming.

10:30

5aPA10. Quasi-emulsion between water and cavitation bubble cloud in an ultrasonic field. Lixin Bai, Weijun Lin, Jingjun Deng, Chao Li, Delong Xu, Pengfei Wu, and Lishuo Chen (Inst. of Acoust., Chinese Acad. of Sci., No. 21, Bei-Si-huan-Xi Rd., Beijing 100190, China, blx@mail.ioa.ac.cn)

The cavitation bubble distribution (cavitation structure) is spatially inhomogeneous in ultrasonic field. A quasi-emulsion phenomenon of cavitation structure is reported in this paper. The inception process of cavitation bubble cloud in a thin liquid layer (the thin liquid layer is trapped between a radiating surface and a hard reflector) was investigated experimentally with high-speed photography. It is revealed that cavitation bubble cloud can be considered as a uniform fluid (cloud), and water without cavitation can be considered as another uniform fluid (water). The conversion from cloud-in-water emulsion to water-in-cloud emulsion occurs with the increase in the number of bubbles. The formation and stability of cloud-water emulsions is analyzed theoretically. It is found that surface tension of cavitation bubble cloud played a leading role. Findings of this research proved that cavitation bubble clusters can be considered and investigated as a whole. [This work was supported by the National Natural Science Foundation of China (Grant No. 11174315).]

10:45

5aPA11. Dynamics of cavitating layer on a surface of quasi-empty cavity in a real heterogeneous liquid. Valeriy Kedrinskiy (Physical HydroDynam., Lavrentyev Inst. of HydroDynam., Russian Acad. of Sci., Lavrentyev Prospect 15, Novosibirsk 630090, Russian Federation, kedr@hydro.nsc.ru) and Ekaterina S. Bolshakova (Physical Faculty, Novosibirsk State Univ., Novosibirsk, Russian Federation)

The state dynamics of a heterogeneous medium surrounding the spherical cavity is considered, using the multi-phase mathematical model with an incompressible liquid component. The heterogeneous medium contains 1.5-micron bubbles with a density of 10^6 cm^{-3}. At $t = 0$, the pressure in the cavity is sharply decreased up to p (0) values forming instantly the dynamically changing decompression wave in the surrounding medium. Numerical analysis allowed to find the significant influence of p (0) values on the dynamics of the cavitating layer on the cavity surface. When p $(0) = 10^{-2}$ MPa, approximately 20 pulsations of bubble radii and their internal pressure are observed during the collapse time of 1 cm cavity (T is 0.9 ms). Most of them reach only its initial state with a pressure about 0.1 MPa. But at approaching to the time moment close to T value, a few pressure pulsations demonstrate the cumulative effect of over-compression up to the pressures of 0.6, 1.8, and 3 (in log-scale). The letter means that cavitating boundary layer accumulates the energy of high density. When the p $(0) = 10^{-4}$ MPa, only one pulsation synchronous with the cavity collapse as well as the over-compression effect are observed. [Work supported by RFBR, grant 15- 05-03336.]

11:00

5aPA12. Ultrasound directed self-assembly of user-specified patterns of nanoparticles dispersed in a fluid medium. John Greenhall, Fernando Guevara Vasquez, and Bart Raeymaekers (Mech. Eng., Univ. of Utah, 1495 e 100 s, Salt Lake City, UT 84112, john.greenhall@utah.edu)

We employ an ultrasound wave field generated by one or more ultrasound transducers to organize large quantities of nanoparticles dispersed in a fluid medium into two-dimensional user-specified patterns. To accomplish this, we theoretically derive a direct method of calculating the ultrasound transducer parameters required to assemble a user-specified pattern of nanoparticles. The computation relates the ultrasound wave field and the force acting on the nanoparticles to the ultrasound transducer parameters by solving a constrained optimization problem. We experimentally demonstrate this method for carbon nanoparticles in a water reservoir and observe good agreement between experiment and theory. This method works for any simply-closed fluid reservoir geometry and any arrangement of ultrasound transducers, and it enables using ultrasound directed self-assembly as a scalable fabrication technique that may enable a myriad of engineering applications, including fabricating engineered materials with patterns of nanoscale inclusions.

11:15

5aPA13. Real-time polymerization monitoring of a thermosetting resin around its glassy transition temperature. Nacef Ghodhbani, Pierre Marechal, and Hugues Duflo (LOMC, UMR 6294 CNRS, Université du Havre, 75 rue Bellot, Le Havre 76600, France, nacef.ghodhbani@univ-lehavre.fr)

Real-time ultrasonic monitoring is investigated to quantify changes in physical and mechanical properties during the manufacture of composite structures. In this context, an experimental transmission was developed with the aim to characterize a high temperature polymerization reaction and post-curing properties using an ultrasonic method. First, the monitoring of ultrasonic parameters of a thermosetting resin is carried out in an isothermal polymerization process at 160°C. During this curing, the resin is changing from its initial viscous liquid state to its final viscous solid state. Between those states, a glassy transition stage is observed, during which the physical properties are strongly changing, i.e., an increase of the ultrasonic velocity up to its steady value and a transient increase of the ultrasonic attenuation. Second, the ultrasonic inspection of the thermosetting resin is performed during a heating and cooling process to study the temperature sensitivity after curing. This type of characterization lead to identify the ultrasonic properties dependence before, during, and after the glassy transition temperature T_g. This study is composed of two complementary parts: the first is useful for the curing optimization, while the second one is fruitful for the post-processing characterization in a temperature range including the glassy transition temperature.

11:30

5aPA14. Yeast flocculation using acoustic agglomeration. Mark H. Holdhusen (Eng., Univ. of Wisconsin, Marathon County, 518 S 7th Ave., Wausau, WI 54401, mark.holdhusen@uwc.edu)

A major cause of haziness in beer is due to yeast suspended in the liquid. The most common methods used to flocculate the yeast and have it settle

out of the liquid solution are to use various chemicals, cold temperatures, and/or extended time. This research considers using acoustic agglomeration to cause the yeast to flocculate as a means to clarify beer. Acoustic agglomeration uses high intensity acoustic standing waves to clump fine particles together in order for them to become large enough to settle out of the fluid. In this work, an ultrasonic acoustic transducer will be implemented to achieve standing waves in the beer. In theory, these standing waves will cause the yeast particles to clump together and settle. The results from this approach will be compared to the traditional methods of yeast flocculation. This approach may lead to an increase in efficiency in clarifying beer without the use of undesirable chemicals. The preliminary work considered here is a proof of concept and will use visual inspection as a means of clarity comparison. Future work will use more in-depth laboratory analysis as a means of comparison of clarity.

11:45

5aPA15. Photoacoustic spectroscopy with SF_6, an optically thick greenhouse gas. Han Jung Park and Wittmann S. Murphy (Chemistry, Univ. of Tennessee at Chattanooga, 615 McCallie Ave., Chattanooga, TN 37403, hanjung-park@utc.edu)

Photoacoustic spectroscopy was used to test the photoacoustic properties of sulfur hexafluoride, an optically thick and a potent greenhouse gas. Detection of trace amounts of the gas was also implemented. The conditions in which the gas was tested, gas cell length, temperature, concentration, and power of the laser, were varied to determine their effect on the photoacoustic signal, and the ideal conditions to detect trace gas amounts. A detection limit of 2.86 ppb was determined for SF_6.

FRIDAY MORNING, 27 MAY 2016 SALON D, 7:55 A.M. TO 12:00 NOON

Session 5aPP

Psychological and Physiological Acoustics: Spatial Hearing

Ewan A. Macpherson, Chair

National Centre for Audiology, Western University, 1201 Western Rd., Elborn College 2262, London, ON N6G 1H1, Canada

Chair's Introduction—7:55

Contributed Papers

8:00

5aPP1. Listener head motion can degrade spatial selective auditory attention. Ewan A. Macpherson (National Ctr. for Audiol., Western Univ., 1201 Western Rd., Elborn College 2262, London, ON N6G 1H1, Canada, ewan.macpherson@nca.uwo.ca) and Blair K. Ellis (Health and Rehabilitation Sci. Graduate Program, Western Univ., London, ON, Canada)

Listeners can use spatial selective auditory attention (SSAA) to focus on one talker in a complex acoustic scene. Although the dynamic binaural cues generated by listener head motion improve sound localization, and might therefore enhance the perceptual differences between separated targets and distractors, the effect of head motion on SSAA is unknown. We measured listeners' ability to attend, with and without head motion, to a frontal target in the presence of two symmetrically separated distractors. During stationary trials, listeners visually fixated and oriented toward a target loudspeaker. On head-motion trials, listeners oscillated their heads at ~0.5 Hz with an amplitude of ~$\pm$40° while continuously directing their gaze toward the target. On each trial, three equal-intensity sequences of four spoken digits were presented simultaneously as target and distractors. Listeners reported the target sequence heard. With distractors at $\pm$22.5°, 86% of target digits were reported correctly without motion, but only 72% were reported correctly with head motion. Correspondingly, the percentage of distractor digits reported as targets increased from 11% to 23%. For widely separated distractors, there was no performance penalty for head motion. These results suggest that listeners cannot rapidly update the focus of their SSAA to compensate for head motion.

8:15

5aPP2. Restoring sensitivity to interaural timing differences in bilateral cochlear implant listeners using multi-electrode stimulation. Tanvi D. Thakkar, Alan Kan, and Ruth Y. Litovsky (Waisman Ctr., Univ. of Wisconsin-Madison, 1500 Highland Ave., Madison, WI 53705, tthakkar@wisc.edu)

Identification and discrimination of sound sources in complex auditory environments is facilitated for normal-hearing listeners from access to interaural time differences (ITDs). For patients fitted with bilateral cochlear implants (BiCI), binaural sensitivity is harder to achieve due to several factors, including asynchronous interaural processing. BiCI listeners have shown good ITD sensitivity with low rates of electrical stimulation; however, low-rate delivery of ITD cues is unrealistic for the high-rate pulsatile stimulation required to achieve good speech understanding. One solution is to present a mix of high- and low-rate stimulation on different electrodes, preserving both speech recognition and sound localization ability. In the present study, a binaural benefit was observed in a mixed-rate strategy under direct electrical stimulation: ITD sensitivity was measured in a two-alternative forced-choice discrimination task, using seven multi-electrode conditions. We hypothesized that by introducing low-rate ITDs at a few electrodes alongside high-rate ITDs at remaining electrodes provides sufficient ITD cues for sound localization. The present data suggests the binaural system of BiCI listeners can extract pertinent cues to achieve ITD sensitivity even when high-rate ITD information is presented at the majority of the cochlear locations. This lends to a possibility in implementation of ITD cues to current processing strategies.

8:30

5aPP3. Shared monaural and binaural temporal processing limits in bilateral cochlear implant listeners. Antje Ihlefeld (Biomedical Eng., New Jersey Inst. of Technol., 323 Martin Luther King Blvd., Fenster Hall, Rm. 645, Newark, NJ 07102, ihlefeld@njit.edu), Robert P. Carlyon (Cognition and Brain Sci. Unit, Medical Res. Council, Cambridge, United Kingdom), Alan Kan, Tyler H. Churchill, and Ruth Y. Litovsky (Waisman Ctr., Univ. of Wisconsin-Madison, Madison, WI)

Bilaterally implanted cochlear implant users were tested on monaural rate discrimination and binaural interaural time difference (ITD) discrimination, as a function of pulse rate, to examine the hypothesis that deterioration in performance at high rates occurs for the two tasks due to a common neural basis. For the rate discrimination task, pulse trains were presented to one electrode, located in the apical, middle, or basal part of the array, and in either the left or the right ear. In each two-interval trial, the standard stimulus had a rate of 100, 200, 300, or 500 pulses-per-second and the signal stimulus had a rate 35% higher. For the ITD discrimination task, performance between pitch-matched electrode pairs was measured for the same standard rates as in the rate discrimination task, and with an ITD of $+/-$ 500 μs. Sensitivity (d') on both tasks decreased with increasing rate. Results show that ITD scores for different pairs of electrodes correlated with the lower of the rate discrimination scores for those two electrodes. Statistical analysis, which partialed out overall differences between listeners, electrodes, and rates, supports the hypothesis that monaural and binaural temporal processing limitations are at least partly due to a common mechanism.

8:45

5aPP4. The perception of reverberation is constrained by environmental statistics. James Traer and Josh H. McDermott (Brain and Cognit. Sci., MIT, 77 Massachusetts Ave., Cambridge, MA 02139, jtraer@mit.edu)

Human sound recognition is robust to reverberation. We explored the hypothesis that this robustness is rooted in the ability to separate the contributions of a sound's source from that of reverberation. As the separation of source and filter from their convolution is inherently ill-posed, any such capacity should depend on prior assumptions about the nature of filter and/or source. We measured the distribution of real-world environmental impulse responses (IRs) and tested whether it constrains the ability of listeners to estimate source and filter from reverberant audio. We surveyed volunteers about the spaces they encountered during daily life, and measured IRs at each location. We found that the tails of real-world IRs decay exponentially, with decay rates consistently slower at low frequencies. We then synthesized IRs that were either faithful to, or deviated from, the observed distribution of real-world IRs. We assessed (a) the sense of reverberation conveyed by IRs, (b) discrimination of novel sound sources in reverberation, and (c) discrimination of IRs given only their convolution with sound sources. We found that human listeners can separately estimate the source and filter in reverberant conditions, but are strongly constrained by whether the filter conforms to the naturally occurring distribution.

9:00

5aPP5. Effect of visual and vestibular information on auditory space perception. Shuichi Sakamoto, Keishi Hanakago, Zhenglie Cui (Res. Inst. of Elec. Commun. and Graduate School of Information Sci., Tohoku Univ., 2-1-1 Katahira, Aoba-ku, Sendai, Miyagi 980-8577, Japan, saka@ais.riec.tohoku.ac.jp), Wataru Teramoto (Faculty of Letters, Kumamoto Univ., Kumamoto, Japan), Yôiti Suzuki (Res. Inst. of Elec. Commun. and Graduate School of Information Sci., Tohoku Univ., Sendai, Japan), and Jiro Gyoba (Graduate School of Arts and Letters, Tohoku Univ., Sendai, Japan)

We investigated how auditory space would be represented during linear self-motion and visually-induced self-motion (vection) (Teramoto *et al.*, 2012, 2014). The previous studies indicated that the subjective coronal plane (SCP) was displaced during forward self-motion, while during backward motion in the case of vection. The present study investigated how auditory space perception would be altered when both visual and vestibular information were presented. A random-dot pattern simulating linear self-motion was used. At the beginning of the trial, the random-dot pattern started to move at a velocity of ± 0.1 m/s. When an observer perceived vection, the velocity was changed at an acceleration of ± 0.2, 0, and 0.4 m/s^2. At the same time, the observer was moved forward at an acceleration of 0.2 m/s^2 by using a linear-motor-driven chair. A short noise-burst was presented from a loudspeaker when the observer moved 2 m. The observers indicated the direction in which the sound was perceived relative to their coronal plane. The results showed that the direction of the SCP shift was different only when the acceleration of both visual and vestibular information was identical. These results suggest that auditory space distortion effect occurs closely related to the integration of vestibular and visual information.

9:15

5aPP6. Sound-source enumeration by hearing-impaired adults. Michael A. Akeroyd (MRC Inst. of Hearing Res., Nottingham NG7 2RD, United Kingdom, maa@ihr.mrc.ac.uk), William M. Whitmer, David McShefferty, and Graham Naylor (Scottish Section, MRC/CSO Inst. of Hearing Res., Glasgow, United Kingdom)

To help measure the veracity with which the auditory world is heard by hearing-impaired listeners, we studied the ability to count the number of discrete sources that are present. In two experiments, stimuli were newly selected 5-s samples of concatenated sentences from 1 to 7 locations in simulated rooms of varying reverberation, presented from a circular loudspeaker array. In the first experiment, listeners responded with the number of locations heard after each presentation. In the second experiment, listeners heard two presentations, an interval with n sources (one talker per location) and an interval with $n+1$ sources; listeners responded which interval had more sources. Results corroborate recent work showing the maximum number of identifiable sources is roughly four. Asymmetry in hearing impairment reduced the ability to enumerate locations. Various conditions of presentation were tested: headphone and aided results were not markedly different from unaided results, and reverberation had only a modest inflating effect on the perceived number of sources. The ramifications for complex listening as well as the potential relation to cortical representations of auditory space are discussed. [Work supported by the MRC (U135097131) and the Chief Scientist Office (Scotland).]

9:30

5aPP7. Evaluating performance of hearing-impaired listeners with a visually-guided hearing aid in an audio-visual word congruence task. Elin Roverud (Boston Univ., 712 Cincinnati St., Lafayette, IN 47901, emroverud@gmail.com), Virginia Best, Christine R. Mason, Timothy Streeter, and Gerald Kidd (Boston Univ., Boston, MA)

Hearing-impaired (HI) individuals typically experience greater difficulty listening selectively to a target talker in speech mixtures than do normal-hearing (NH) listeners. To assist HI listeners in these situations, the benefit of a visually guided hearing aid (VGHA)—highly directional amplification created by acoustic beamforming steered by eye gaze—is being evaluated. In past work with the VGHA, performance of NH listeners was assessed on an audio-visual word congruence task [e.g., Roverud *et al.*, ARO 2015]. The present study extends this work to HI listeners. Three spoken words are presented simultaneously under headphones at each of three source locations separated in azimuth. A single word is printed on a monitor at an angle corresponding to one of the locations, indicating the auditory target word location. After each stimulus, listeners indicate whether the auditory target matches the printed word with a YES-NO response, ignoring the two masker words. Listeners visually track the printed word, which moves location trial-to-trial with a predetermined probability. Participants' eye movements, measured by an eye tracker, steer the VGHA amplification beam. Performance is compared for the VGHA, stimuli processed with natural binaural information (KEMAR HRTFs), and a hybrid condition combining both types of information. [Work supported by NIDCD.]

9:45

5aPP8. Temporal integration of dynamic binaural events. G. Christopher Stecker, Julie M. Stecker, Anna C. Diedesch, Nathan C. Higgins, and Sandra Da Costa (Hearing and Speech Sci., Vanderbilt Univ., 1215 21st Ave. South, Rm. 8310, Nashville, TN 37232, g.christopher.stecker@vanderbilt.edu)

Events occurring in different sensory dimensions (e.g., modalities) and within an interstimulus interval (ISI) of few hundred milliseconds typically

appear simultaneous. The temporal "window" of sensory integration can be estimated by increasing the ISI until events appear discrete. Here, we adapt the simultaneity-judgment paradigm to investigate the integration of binaural events within and across dimensions of interaural time and level difference (ITD and ILD). Noise bands (353.5–707 Hz) of 2s duration were presented over headphones to normal-hearing listeners. Each stimulus contained two 10-ms binaural-change events, during which the ITD or ILD changed by 500 μs or 6 dB. Events could occur in the same or different cue dimensions (ITD or ILD), but always agreed in direction (leftward or rightward). ISI varied over the range ±600 ms. Listeners indicated the number and direction of each perceived shift in the lateral image, e.g., one (left-to-right) at short ISI, or two (left-to-center, center-to-right) at long ISI. The threshold for reporting two events was shortest for pairs of ITD events, suggesting greater temporal fidelity for ITD change than for ILD change. Thresholds did not differ between ILD and cross-cue event pairs, suggesting no additional cost of cross-cue integration. [Work supported by NIH R01-DC011548.]

10:00–10:15 Break

10:15

5aPP9. Sensitivity to binaural cues beyond threshold as revealed by eye movements. Matthew Winn (Speech & Hearing Sci., Univ. of Washington, 1417 NE 42nd St, Seattle, WA 98105, mwinn83@gmail.com) and Alan Kan (Waisman Ctr., Univ. of Wisconsin-Madison, Madison, WI)

Binaural cues are paramount for sound localization along the azimuth. Studies on the perception of the critical binaural cues, interaural time and level differences (ITDs and ILDs, respectively), typically measure sensitivity at threshold using N-interval, forced choice paradigms. This approach gives little or no information regarding perceptual abilities beyond threshold. In this study, an anticipatory eye movement (AEM) paradigm (cf. McMurray, 2004) was used as a novel measure to study binaural cue sensitivity throughout a wide perceptual range. This paradigm is sensitive to gradient (rather than all-or-none) perception of auditory cues. Adults with normal hearing visually tracked the location of a ball that becomes hidden on a computer screen, and anticipated its motion and reappearance via eye movements guided by binaural cues. The auditory stimuli were 4.8-s 1/3-octave narrowband noises centered at different frequencies, and contained an ILD or ITD change, indicating the impending motion of the ball. Greater cue levels elicited systematically quicker and more accurate saccades. ILD-driven AEMs were elicited for all noise center frequencies, with greater sensitivity for higher frequency noises, consistent with ecological significance. Eye movements in this study suggest variation in binaural cue perception beyond threshold that is gradient in terms of latency and accuracy.

10:30

5aPP10. The role of early and late reflections on spatial release from masking. Nirmal Kumar Srinivasan, Meghan M. Stansell, Rachel E. Ellinger, Kasey M. Jakien, Sean D. Kampel, and Frederick J. Gallun (National Ctr. for Rehabilitative Auditory Res., 3710 SW US Veterans Hospital Rd., Portland, OR 97239, nirmal.srinivasan@va.gov)

It is well documented that older listeners have more difficulty understanding speech in complex listening environments than do younger listeners. Early reflections (occurring <50 ms following the direct sound) have been linked to improved speech intelligibility (Lochner and Burger, 1964), while later-arriving reverberant sound has been shown to limit speech understanding (Knudsen, 1929). However, we do not know how spatial release from masking (SRM) is affected by early and late reflections or how age and hearing loss interacts with the relative influences of each. SRM in two simulated reverberant environments was measured for listeners varying in age and hearing thresholds under three different signal processing conditions: (1) early reflections alone, (2) late reflections alone, and (3) all reflections. Results indicated that though all listeners performed better when only early reflections were present, the older hearing-impaired listeners benefited the most from the absence of late reflections. Effects of age and hearing loss on performance and SRM under these three different signal processing conditions will be discussed. [Work supported by NIH R01 DC011828.]

10:45

5aPP11. Discrimination and streaming of speech sounds based on differences in lateralization in the horizontal and median planes. Marion David and Andrew Oxenham (Psych., Univ. of Minnesota, 75 East River Parkway, Minneapolis, MN 55455, david602@umn.edu)

Understanding speech in complex backgrounds relies on our ability to perceptually organize competing voices into streams. Differences in fundamental frequency (F0) between voiced sounds are known to enhance stream segregation; less is known about the perceptual organization of unvoiced sounds such as fricative consonants. We showed previously that natural consonant-vowel (CV) pairs can be segregated based on F0 differences, despite lacking F0 cues in the fricative part. This study also used CVs, filtered by head-related impulse responses (HRIR) to simulate different positions in the horizontal and median planes. In the median plane, cues are limited to high frequencies, and so should affect fricatives more than vowels. Both discrimination, using a three-interval forced-choice task, and streaming, using a within- or across-stream repetition-detection task, were tested. The CV pairs were either held constant during a trial, or varied randomly. In the constant condition, any difference in spectrum would indicate a change in location along the median plane; in the variable condition, such differences would require listeners to extract spectral regularities from across widely varying spectra of the speech stimuli. Preliminary results suggest that discrimination and streaming are more challenging in the median than in the horizontal plane. [Work supported by NIH grant R01DC012262.]

11:00

5aPP12. Acoustic cues for determining head position in sound source localization when listeners move. William Yost (Speech and Hearing Sci., ASU, PO Box 870102, Tempe, AZ 85287, william.yost@asu.edu)

Recently, Yost *et al*. [J. Acoust. Soc. Am. **40**, 3293–3310 (2015)] showed in experiments involving rotating listeners that everyday world-centric sound source localization involves two sources of information: information about the auditory spatial cues and information about the position of the head. In this presentation, we will describe experiments in which sound has the potential to provide information about the position of the head, allowing for world-centric sound source location. We use conditions in which sound rotates along an azimuth circular array of loudspeakers as listeners also rotate in the azimuth plane at constant velocity with their eyes open or closed. The rotation conditions and the resulting perceptions of sound rotation will be described along with how those perceptions are altered when other sounds are present. [Research supported by the Air Force Office of Scientific Research, AFOSR.]

11:15

5aPP13. Binaurally integrated cross-correlation/auto-correlation mechanism (BICAM). Jonas Braasch (School of Architecture, Rensselaer Polytechnic Inst., 110 8th St., Troy, NY 12180, braasj@rpi.edu)

A new precedence effect model is described that can use a binaural signal to robustly localize a sound source in the presence of multiple reflections. The model also extracts the delays (compared to the direct sound source) and lateral positions of each of the distinct reflections. A second-layer cross-correlation algorithm is introduced on top of a first layer auto-correlation/cross-correlation mechanism to determine the interaural time difference (ITD) of the direct sound source component. The ITD is then used to time align two auto-correlation functions obtained from the left and right ear signals to gather information about the reflections and form a binaural activity pattern. The model is able to simulate psychoacoustic lateralization results for a simulated direct sound source and a single reflection also for cases where the reflection exceeds the intensity of the direct sound (Haas Effect). Using head-related transfer functions to spatialize the sound sources, the model can accurately localize a speech signal in the presence of two or more early side reflections and late reverberation. The model can handle reverberation times of 2 s and above. [This material is based upon work supported by the National Science Foundation under Grant No. 1320059.]

11:30

5aPP14. Squelch of room effects in everyday conversation. Aimee Shore, William M. Hartmann, Brad Rakerd (Michigan State Univ., Phys. Astronomy, 567 Wilson Rd., East Lansing, MI 48824, shoreaim@msu.edu), Gregory M. Ellis, and Pavel Zahorik (Univ. of Louisville, Louisville, KY)

When a conversation is recorded and then played back, listeners are aware of effects of the room that are not perceived in face-to-face conversation. The room effects, which are physically present, are said to be "squelched" under face-to-face conditions. Room effects include reverberation and coloration caused by reflections. The squelch effect has been attributed to the binaural nature of natural listening, based on binaural experiments with headphones [W. Koenig, J. Acoust. Soc. Am., {22}, 61–62 (1950)]. We report experiments on binaural and diotic listening to recordings of speech made at conversational distances in a room with normal frequency dependence of reverberation and a direct to reverberant power ratio between 1/2 and 1/3 at speech fundamental frequencies. Recordings were made with and without a head between the microphones. Listeners ranked the recordings in order of increasing perceived room effects. The data revealed a strong effect of distance, a weak effect of head diffraction, and an advantage for binaural listening somewhat smaller than the advantage of a factor of 2 in direct power. For a few listeners, binaural listening enhanced the room effects. The latter listeners apparently found that binaural differences produced especially prominent spatial effects. [Work supported by the AFOSR.]

FRIDAY MORNING, 27 MAY 2016 — SALON B/C, 8:00 A.M. TO 9:50 A.M.

Session 5aSCa

Speech Communication: Phonetics of Under-Documented Languages I

Amanda L. Miller, Cochair
Linguistics, The Ohio State University, 222 Oxley Hall, 1712 Neil Avenue, Columbus, OH 43210-1298

Richard Wright, Cochair
Linguistics, University of Washington, Box 352425, Seattle, WA 98195-2425

Benjamin V. Tucker, Cochair
Linguistics, University of Alberta, 4-32 Assiniboia Hall, Edmonton, AB T6G 2E7, Canada

Chair's Introduction—8:00

Invited Papers

8:05

5aSCa1. Under-documented languages expand phonetic typology. Ian Maddieson (Dept. of Linguist, UNM, Univ. of New Mexico, MSC03-2130, Albuquerque, NM 87131-0001, ianm@berkeley.edu)

The set of speech sounds known to be used in human languages continues to grow ever larger as more information becomes available on previously under-documented languages. In addition, the range of contrastive distinctions known to be employed continues to be enlarged. A brief survey of categories of sounds that have been added to phonetic typology as a result of work on such languages will be presented, followed by exemplification of the specific case of Yélî Dnye (ISO 693 yle). This language, spoken on Rossel Island, Papua New Guinea by about 3000 people, has large contrastive inventories of both consonants and vowels (58 consonants, 34 vowels). It is the only language known to include sets of doubly articulated labial-alveolar and labial-postalveolar plosives and nasals in its inventory (in addition to the more widespread category of labial-velars). Moreover, the stops contrast plain, prenasalized and nasally released categories, and some of them occur distinctively palatalized. Thus, there are at least nine consonant types not known from any other language. Yélî Dnye is also the only language known to have a contrast between oral and nasalized vowels following nasally released stops. Our knowledge of this language therefore enlarges our perspective on how human languages may differ.

8:30

5aSCa2. Acoustic realization of a distinctive, frequent glottal stop: The Arapaho example. Doug H. Whalen (Haskins Labs., 300 George St. Ste. 900, New Haven, CT 06511, whalen@haskins.yale.edu), Christian DiCanio (Haskins Labs., Buffalo, NY), Christopher Geissler, and Hannah King (Haskins Labs., New Haven, CT)

Complete closure of the glottis is typically treated as the canonical realization of glottal stop, but it has instead been found to be "quite unusual" in running speech. However, such evidence comes mostly from English, with non-phonemic glottal stops. How do glottal stops vary in a language where they are common and contrastive, as in Arapaho (ISO 639 arp)? Does distinctive and frequent use of

a glottal stop lead to more canonical productions? Moreover, glottalization is often used to mark prosodic boundaries; Are Arapaho phonemic glottal stops affected by boundary position? Glottal stops in an Arapaho corpus were classified on a scale of lenition and examined for duration, relative intensity, harmonics-to-noise ratio (HNR), and F0. Results show that glottal stops were seldom realized as a stop (only 25%) but instead mostly as glottalization. HNR was lower in glottalization than in adjacent vowels. Word-final glottal stops were more often realized with full closure than word-internal ones. The rarity of full glottal stops in English is also reflected in Arapaho: Greater use of these stops does not result primarily in canonical stop realizations. Moreover, glottal stop realization varies prosodically; thus, glottalization as a prosodic feature is not restricted to non-phonemic glottal stops.

8:50

5aSCa3. On the possible origin of voiceless implosives: Hints from Ese'eja (Takanan). Ddiier Demolin (ILPGA, LPP sorbonne nouvelle, 19 rue des Bernardins, Paris 75005, France, ddemolin@univ-paris3.fr) and Marine Wuillermet (Linguist, Radboud Universitei, Nijmegen, Netherlands)

Voiceless implosives are reported in a few languages of Africa, Mesoamerica, and Amazonia. Ese'eja (Takanan) has bilabial [ɓ̥] and [ɗ̥] alveolar voiceless implosives in its phonemic inventory (Vuillermet 2006). These sounds are realized with a complete closure of the vocal folds, a lowering of the larynx during the glottal closure associated with a lowering of the pressure (Po) inside the vocal tract behind the labial or alveolar closures. This is followed by a rapid larynx rising. The acoustic characteristics are: a period of silence and a short prevoicing preceding a strong final burst. There are two possibilities to explain the origin of voiceless implosives in Ese'eja. The first is that they are the consequence of the devoicing of a voiced implosive. The second is that they are due to the combination of a glottal closure and a voiced stop. The lowering of the glottis and Po is anticipating the articulation of a following low or back vowel during the glottal closure. Voiceless implosives have been described as preglottalized stops in other languages. (Dimmendaal 1986) emphasizes that in Lendu there is a possible auditory confusion between voiceless implosives and preglottalized stops. Aerodynamic and acoustic measurements confirm this hypothesis for Ese'eja.

9:10

5aSCa4. Interaction of pitch and phonation in Louma Oeshi. James Gruber (Dept of Linguist, Reed College, 3203 SE Woodstock Blvd., Portland, OR 97202-8199, james.gruber@reed.edu) and Sigrid Lew (Graduate Linguist Dept., Payap Univ., Chiang Mai, Thailand)

Louma Oeshi is an Akoid (Tibeto-Burman) language of Laos for which acoustic characteristics are undocumented with the exception of preliminary work by the present authors. This study focuses on the phonetic properties associated with Oeshi's three tones (high, mid, and low) and two registers (Tense, Lax), which fully intersect to yield a six-way suprasegmental contrast. Eight speakers were recorded in Phongsali Province, Lao PDR. Each spoke a 100 token word list, for which 30 tokens were repeated in a carrier sentence placing the token between high and mid-toned lax words. Decile measures of F0 and an array of measures reflecting phonation types (H1-H2, H1-A1, H1-A3, HNR, and SHR) were performed in Voicesauce (Shue *et al*. 2011) to capture dynamic values over the syllable duration. Our findings show a reliable three-way F0 contrast between tones and a single interaction with register such that High Tense words consistently fall (tones are otherwise level). Acoustic correlates of the Tense~Lax distinction are less clear. The picture that emerges is one where Tense register has variable phonetic manifestations — preglottalization of onsets, vocalic creaky voice, or a glottal stop coda — and the seemingly inconsistent acoustic results reflect these variable articulatory timing strategies.

9:30

5aSCa5. Investigating phonetic bases of sound patterns in Mangetti Dune !Xung: Contributions of high frame rate ultrasound to the description of endangered languages. Amanda L. Miller (Linguist, The Ohio State Univ., 222 Oxley Hall, 1712 Neil Ave., Columbus, OH 43210-1298, miller.5592@osu.edu)

High frame rate lingual ultrasound methods (Miller and Finch 2011) allow investigations of consonant and vowel kinematics outside of the laboratory. I summarize the results of a number of studies that investigate place and manner of articulation in consonants produced with the pulmonic and lingual airstream mechanisms in the endangered Namibian language Mangetti Dune !Xung. Results show that clicks, like dorsal stops, exhibit different postures of the posterior part of the tongue when they precede [i] and [ɑ]. The tongue dorsum and root are retracted in the production of all four coronal clicks when they precede [ɑ], but differ in their postures when they precede [i]. Further, tongue dorsum and root postures are less variable within click types before [ɑ], than they are preceding [i]. Clicks also differ in the timing of the anterior and posterior releases, resulting in different constrictions being adjacent to following vowels, thus leading to different co-articulation patterns. Timing patterns of the two releases also contribute to our understanding of a diachronic sound change from an abrupt palatal click to a fricated post-alveolar click in this family. This body of research illustrates how high frame rate ultrasound contributes to descriptions of sound patterns in under-described languages.

Session 5aSCb

Speech Communication: Phonetics of Under-Documented Languages II (Poster Session)

Amanda L. Miller, Cochair
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All posters will be on display and all authors will be at their posters from 10:05 a.m. to 12:00 noon.

Contributed Papers

5aSCb1. Acoustic phonetic study of phonemes and tonemes of spoken Punjabi language. Shyam S. Agrawal, Shweta Bansal, Shambhu Sharan, and Amritpal Singh (College of Eng., KIIT, Sohna Rd., Near Bhondsi, Gurgaon, Haryana 122102, India, dr.shyamsagrawal@gmail.com)

Punjabi language is one of the important languages among 22 official languages in India. It is spoken by about 105 million people in India. The present paper describes a study and results of detailed acoustic analysis of vowels, consonantal phonemes and tonemes as spoken by the speakers of Malwai dialect of Punjabi language. A database of 1500 words containing all the phonemes and tonemes, selected from a text corpus of 300, 000 words were used for the study. These words were recorded and segmented by using signal processing tools to analyze the samples of speech. Fundamental frequency, first three formants, and bandwidths for nasal and non-nasal vowels were measured. For the study of consonants, duration of sub-phonemic segments such as occlusion, burst, and VOT have been computed and compared. Special features of tonemes have been studied and compared with non-tonal phonemic segments. Tones are fully phonemic and words with similar spellings are distinguished by varying tones- low, mid, and high and corresponding accent marks. It has been observed that intonation plays a significant role in the discrimination of phonemes and tonemes. These results can be used to create PLS and phonetic dictionary of Punjabi speech.

5aSCb2. Consonant-tone interactions in Gengbe. Samson Lotven and Kelly Berkson (Dept. of Lingustics, Indiana Univ., 1021 E. Third St., Mem 322E, Bloomington, IN 47405, kberkson@indiana.edu)

Gengbe is an under-documented and understudied Gbe language spoken as a lingua franca in Southern Togo and Benin. Few resources for Gengbe exist, especially in the domain of empirical acoustic phonetic research. To this end, we present an overview of its consonant, vowel, and tonal inventories. Of particular interest is the fact that its two register tones, (L)ow and (H)igh, show systematic phonological variation based on the voicing of onset consonants. Like many other tone languages in Africa and beyond, voiced obstruents act as so-called "depressor consonants," triggering lower f0 on subsequent vowels than do their voiceless counterparts. This lowering is phonologized in many H tone contexts, resulting in a Rising tone in many (but not all) morphophonological environments. The distinction is phonetically present in low tone as well, where it perseverates across the entire vowel and results in a lower register L (by approx. 20 Hz). We survey the phonological environments where such lowering effects are realized, and probe the interaction between obstruent voicing, tone, vowel height, and nasality via instrumental acoustic analysis.

5aSCb3. Pronunciation change in SENĆOŦEN: A acoustic study of /k, k^w, k^w', q, q', q^w q^w'/ across generations of speakers. Sonya Bird (Linguist Dept., Univ. of Victoria, PO Box 1700 STN CSC, Victoria, BC V8W 2Y2, Canada, sbird@uvic.ca)

This paper presents an acoustic study of /k, k^w, k^w', q, q', q^w q^w'/ in SENĆOŦEN, a dialect of North Straits Salish. The "K series"—as these sounds are collectively called—are of particular concern among speakers, the perception being that both the uvular ~ velar and the plain ~ ejective contrasts may be disappearing. To understand how these contrasts are currently being realized, 12 speakers (3 speakers x 4 generations) were recorded pronouncing 20 isolated words (7 consonants x 2–3 words). Based on auditory impression and preliminary acoustic analysis (PLACE: spectral composition of bursts/frication; formant transitions into adjacent vowels; VOICING: VOT; jitter; shimmer; amplitude rise time), the velar ~ uvular contrast is indeed less consistent among younger speakers than among their elders; there is also a lot of variability in the plain ~ ejective contrast, among all speakers. In addition, the plain ~ ejective contrast interacts with the velar ~ uvular contrast for some speakers: /k kw/ and /q qw/ are merged and relatively fronted ([k kw]); /kw'/ and /qw'/ are merged and relatively backed ([qw']). These realizations likely reflect a combination articulatory/aerodynamic considerations and orthographic influences (e.g., /kw'/ = 'Q'; /qw/ = 'Ḱ').

5aSCb4. A preliminary acoustic analysis of vowels and obstruents in San Juan Quiahije Chatino. Jacob Heredos, Kaitlynn Milvert, and Hilaria Cruz (Indiana Univ., 1021 E 3rd St., Memorial Hall 322 E (Linguistics), Bloomington, IN 47405, jheredos@indiana.edu)

San Juan Quiahije (SJQ) Chatino is a language of the Zapotecan branch of the Otomanguean family, spoken by approximately 4000 people in the state of Oaxaca in southern Mexico. While acoustic analyses of other varieties of Chatino exist, empirical acoustic data related to the phonetic inventory of SJQ Chatino is slim to nonexistent. As such, the current work presents a preliminary investigation of the stop consonants and vowels of SJQ Chatino. Data are from one female native speaker of SJQ and include words produced in isolation as well as in running speech. F1 and F2 values are used to plot the five oral and four nasal vowels of SJQ. With regard to the consonant inventory, contrastive voicing has been lost in at least one variety of Chatino and is marginal in a number of other varieties. Our data confirm that SJQ retains the contrast in coronal stops in at least some contexts. Voice onset times are reported, and negative values are found after initial /n/.

5aSCb5. Consonantal timing in Eastern Armenian. Knar Hovakimyan (Linguist, Reed College, #1328, 3203 SE Woodstock Blvd., Portland, OR 97202, knhovakim@reed.edu)

Eastern Armenian is an under-documented language, particularly in the field of phonetics. The language treats onset and coda consonant clusters asymmetrically, breaking up onset clusters with a vowel, but not coda clusters. This study focuses on consonantal timing in word-initial and word-final consonant sequences in Armenian. Five speakers were recorded reading 50 pairs of words with representative consonant clusters and equivalent words without clusters. Results indicate that singleton consonants behave differently than complex consonants depending on their location within a word. Namely, the vowel-adjacent consonant in a complex coda is longer in duration than an equivalent singleton consonant, whereas in onsets the duration of consonants is stable regardless of whether or not there is a preceding consonant. This corresponds to a greater degree of overlap in codas than in onsets. This asymmetrical treatment is motivated by the need to preserve phonetic cues for distinguishing consonants. The 30-consonant inventory is greatly limited by neutralizations in coda position, but all consonantal contrasts are maintained in onset clusters. Since there are more consonants to be distinguished in onset position, stronger cues are required to identify them and so less overlap is tolerated.

5aSCb6. Voicing contrast in Ruwund. Didier Demolin (LPP-ILPGA, Université Sorbonne nouvelle, 19 rue des Bernardins, Paris 75005, France, ddemolin@univ-paris3.fr)

Ruwund has a set of voiceless and voiced stops in its phonemic inventory. Length measurements reveal that voiceless stops are realized with duration comparable to voiced stops. There is also a positive VOT varying between bilabial, alveolar, and velar places of articulation. Voiced stops are characterized by a negative VOT and have greater voicing intensity. What is striking in Ruwund is that voiceless stops are also produced with voicing of weak intensity. These sounds should therefore be described as voiced stops. The presence and amount of aspiration during the delay between the first burst and the onset of voicing is the cue that contributes to these sound's identification as voiceless. Repp [1] observed that the increase in the amplitude of aspiration noise relative to the following periodic vocalic portion increases the salience of this cue and helps the probability to classify these consonants into the voiceless category. A perceptual test m confirms this hypothesis. Removing the part lying between the first burst and the following voiced part accounting for the beginning of the following vowel, show that listeners do not recognize these stops as voiceless anymore but as voiced. Ruwund reveal therefore new subtleties of voicing distinction and VOT. [1] B. Repp, "Relative amplitude of aspiration noise as a voicing cue for syllable-initial stop consonants," Lang. Speech, **23**, 173–189 (1979).

5aSCb7. San Juan Quiahije Chatino: A look at tone. Colette Feehan, Kelly Berkson, Malgorzata Cavar (Lingustics, Indiana Univ., 1021 E. Third St. Memorial Hall 322, IU Bloomington, IN 47405-7005, cmfeehan@umail.iu.edu), and Hilaria Cruz (Linguist, Univ. of Kentucky, Lexington, KY)

San Juan Quiahije (SJQ) Chatino is an under-documented indigenous language spoken in Oaxaca, Mexico, by some 3000 speakers. Like other members of the Eastern Chatino branch, SJQ has a complex tonal system—in particular, four tone levels and 11 lexical tones (14 when considering sandhi contexts). Excellent linguistic investigations of SJQ phonology (Cruz 2004, 2011) and discourse analysis (Cruz & Woodbury 2014, Cruz 2014) exist, but there is virtually no empirical phonetic work. We present exploratory and descriptive investigation of the tones of San Juan Quiahije Chatino. Data from one female native speaker largely align with the tonal description provided in Cruz (2011): our findings confirm the presence of a multitude of tones, and reveal that several lexical tones are merged in isolation. Furthermore, while depressor consonant effects—wherein tonal targets are lower after voiced obstruents than after voiceless obstruents—have traditionally been discussed primarily in the context of African languages (Bradshaw 1999), we discover a consistent lowering of tone by approximately 25 Hz after voiced consonants. The effect is present in both low and high tone contexts and perseverates throughout the vowel.

5aSCb8. An acoustic analysis of the vowels and stop consonants of Bashkir. Kelly H. Berkson, Matthew C. Carter, and Christopher M. Robbins (Linguist, Indiana Univ., 1021 E 3rd St., Memorial Hall 322 E, Bloomington, IN 47405, cartermc@umail.iu.edu)

Bashkir is a language of the Volga-Kipchak branch of the Turkic language family, spoken by approximately 1.2 million ethnic Bashkirs primarily in the autonomous Republic of Bashkortostan, Russia. Minimal research has been conducted on Bashkir in English, and what research has been conducted in either Russian or English focuses on the morphophonemics, syntax, and semantics of the language: acoustic investigation of Bashkir, meanwhile, is nearly nonexistent. This study is a preliminary examination of the phonetics of Bashkir. Using data from a female native speaker in her early 50s from Ufa, Bashkortostan, we present instrumental analysis of vowels in both pre-stressed and stressed positions, as well as voice onset time measures for oral stops. The vocalic data, in particular, are surprising in multiple ways: while they largely align with descriptions of the Bashkir vowel space provided by previous sources, with mid vowels that are reduced in most positions and that also have a large, variable range in the vowel space, they also suggest that the realization of one vowel phoneme may differ substantially from previous descriptions.

5aSCb9. The acoustics of strengthened glides in Kirundi. Alexei Kochetov (Linguist, Univ. of Toronto, 100 St. George St., Sidney Smith 4076, Toronto, ON M5S 3G3, Canada, al.kochetov@utoronto.ca)

Strengthening of post-consonantal glides /w/ and /j/ to obstruents or nasals is cross-linguistically uncommon, especially when the process is synchronically conditioned. This study investigates acoustic correlates of post-consonantal glide strengthening in Kirundi, an eastern Bantu language spoken in Burundi. The materials included words with /w/ or /j/ preceded by labial obstruent or nasal consonants /p, b, v, m/—the contexts that have been previously described to trigger the process. The morphological context was also varied, with sequences occurring at prefix-root or suffix-root boundaries, or within a root. Four female and four male native speakers produced the word list in a carrier sentence three times. The analysis involved phonetic classification of strengthened glides, as well as their duration and spectral measurements (frication/burst spectra and following vowel formants). The results showed that the post-consonantal /w/ was consistently realized as an oral or nasal stop agreeing with the preceding consonant in voicing and nasality, consistently with previous descriptive accounts. The realization of the post-consonantal /j/, however, was considerably more variable across and within speakers, ranging from a palatal stop or nasal to a weakly fricated/nasalized glide, and showing some sensitivity to morphological structure.

5aSCb10. Acoustic analysis of Punjabi stress and tone (Doabi dialect). Kiranpreet Nara (Linguist, Univ. of Toronto, 100 St. George St., Toronto, ON M5S 3G3, Canada, kiranpreet.nara@mail.utoronto.ca)

An acoustic experiment was conducted to study the stress and tone systems of Punjabi, an under-documented Indo-Aryan language. Tone and stress are linked because tone associates with the stressed syllable (Bailey, 1914; Wells and Roach, 1980; Baart, 2003). The experiment was used determine the acoustic cues of stress and the tonal contours of the three Punjabi tones: default, rising, and falling. Five native speakers read a list of 85 words five times. Measurements of duration, intensity, f0 were made in PRAATand analyzed in SPSS. The Mixed Models analysis of normalized intensity and duration revealed that the acoustic cue of stress is the duration of the rhyme. A similar finding for Hindi, a closely related language, is reported in Nair *et al.* (2001). As for tone, the default tone has the smallest f0 range and the falling tone has the largest f0 range. Falling tone is realized entirely on the stressed syllable whereas for the rising tone, the phenomenon of peak delay is observed unless tone occurs on a word-final syllable. Peak delay is also observed in Mandarin (Xu, 2001). This work offers an in-depth understanding of the phonological aspects of the stress and tone systems of Punjabi.

5aSCb11. Manner-specific tongue shape differences in the production of Kannada coronal consonants. Alexei Kochetov (Linguist, Univ. of Toronto, 100 St. George St., Sidney Smith 4076, Toronto, ON M5S 3G3, Canada, al.kochetov@utoronto.ca) and N. Sreedevi (Clinical Services, All India Inst. of Speech and Hearing, Mysore, Karnataka, India)

The production of consonants of the same place but different manner of articulation can involve certain adjustments in the posture of the tongue shape. This can be due to requirements for specific gestures (e.g., lowering the tongue sides for laterals) or constraints on coordination of different gestures (e.g., the tongue-palate constriction and the velum lowering for nasals). This study used ultrasound imaging to examine sagittal tongue shape differences in the production of Kannada (Dravidian) laterals, nasals, and stops of two places of articulation—alveolar/dental and retroflex. Words with these consonants (as geminates) were produced multiple times by five female and five male native speakers of Kannada. The analysis of tongue shapes revealed a lower tongue body/blade for laterals than stops, but only in retroflexes. The opposite was observed for /l/ vs. /t/, likely reflecting the alveolar vs. dental constriction differences. The nasals were produced with a significantly more advanced tongue body than the corresponding laterals and stops. The tongue fronting for nasals can serve to accommodate the velum lowering as part of these consonants' gestural coordination. The magnitude of this effect is further modulated by the consonant's degree of articulatory resistance, which is greater for retroflexes than alveolars/dentals.

5aSCb12. A sociophonetic account of morphophonemic variation in Palestinian Arabic. William M. Cotter (Linguist & Anthropology, Univ. of Arizona, 1009 East South Campus Dr., Tucson, AZ 85721, williamcotter@email.arizona.edu)

This study presents findings from sociolinguistic fieldwork on Palestinian Arabic in the Gaza Strip and Jordan. The sample includes 30 speakers representing three age groups, both genders, and three Palestinian communities: indigenous Gaza City residents, refugees from Jaffa who live in Gaza City, and Gaza City refugees living in Jordan. Linear mixed effects analyses are presented on the vowel raising of the Arabic feminine ending /ah/. The traditional dialect of Gaza realizes this morpheme consistently as [a] (Bergsträsser 1915), with all other Levantine city dialects raising the feminine ending to [e] or [i] except after back consonants (Al-Wer 2007). Similarly, the traditional Jordanian dialect in the area in which Gaza City refugees live also raises this vowel to [e] (Herin 2014). Results indicate robust sociophonetic variation in the realization of this vowel across these communities. Jaffa refugees in Gaza, whose traditional dialect realizes this vowel as [e], are found to be lowering their realization of this vowel with each successive generation, with younger speakers showing a phonetic realization similar to young indigenous Gazans. Simultaneously, Gaza refugees in Jordan show higher phonetic realizations across generations, indicating convergence toward the local realization of this vowel in Jordanian Arabic.

5aSCb13. A computational classification of Thai lexical tones. Jamison E. Cooper-Leavitt (Laboratoire de Phonétique et Phonologie - Sorbonne Nouvelle, Paris 3, 19 rue des Bernardins, Paris 75005, France, jecooper@ucalgary.ca)

The leveling of contour tones in Thai, uttered in a continuous context, has served as a natural point of difficulty for tone recognition experiments. Two tone recognition experiments presented here both include five lexical Thai tones (high, mid, low, rising, and falling) as abstract Bayesian models incorporated into a multi-model Hidden Markov Model. The HMM was developed using Thai natural language utterances to test its performance in correctly identifying Thai lexical tone categories. All utterances used for testing and training were produced in a laboratory setting. Utterances for the first experiment were produced in a citation context, and utterances for the second experiment were produced in a continuous context. The results of the two experiments were compared to test if the context had a significant effect on correctly identifying tone category. Findings showed the context of the utterance had a significant effect on the HMM's ability to correctly identify tone category, $F(1,48) = 5.82$, $p = 0.020$. The identification of the correct tone category for citation utterances showed a significant increase in performance than for the correct identification of tone category for continuous utterances, $t(48) = 5.38$, $p < 0.001$.

5aSCb14. Long-term average spectrum of te reo Māori. Donal Sinex (New Zealand Inst. for Lang., Brain, and Behaviour, Univ. of Canterbury, Utah State Univ., Logan, Utah 84322, don.sinex@usu.edu), Margaret Maclagan, and Jeanette King (New Zealand Inst. for Lang., Brain, and Behaviour, Univ. of Canterbury, Christchurch, New Zealand)

Te reo Māori, the language of the indigenous people of New Zealand, lacks some consonants common in English, notably the fricative /s/ and voiceless plosives. However, Māori pronunciation has adapted over time and exposure to English. Thus, there is reason to expect that the spectrum of spoken Māori differs from that of New Zealand English, but that the magnitude of the difference may be decreasing. We measured the long-term average spectrum (LTAS) of the speech of Māori-English bilinguals, using recordings from a longitudinal study of Māori speech. The oldest talkers in the database were born in the late 19th century. Female and male, talkers, and both native and non-native speakers of Māori were included. The LTAS was determined for each talker and each language. For individual talkers, the average spectrum had consistently higher amplitude for English than Māori, for frequencies above approximately 2 kHz. Across talkers, the mean difference at 8 kHz was approximately 8 dB, for all but one group. The exception was younger female Māori L2 talkers, for whom the difference was only 3 dB. These observations are consistent with the consonant inventory of te reo Māori, and they also provide potentially useful information about ongoing changes in the language.

5aSCb15. Production and perception of the advanced tongue root vowel system in Ethiopian Komo. Paul Olejarczuk, Manuel A. Otero, and Melissa M. Baese-Berk (Univ. of Oregon, 1290 University of Oregon, Eugene, OR 97403, paulo@uoregon.edu)

Komo [xom] is an endangered and under-documented language spoken along the Ethio-Sudanese border. This paper presents the results of the first phonetic investigation of the Komo vowel system and reports on a related perception experiment carried out in the field. Our first aim is to provide an acoustic description of the Advanced Tongue Root (ATR) feature in Komo, which is contrastive in the high vowels and allophonic in the non-high vowels. To this end, we present acoustic measurements of $2,688$ vowel tokens produced by 16 speakers. Our second aim is to examine the influence of Komo's typologically unique vowel harmony system on listeners' perception of the [ATR] feature. Komo ATR harmony displays two competing processes triggered by the high vowels (Otero, 2015): [$+$ATR] spreads leftward to non-high vowels (e.g., /CaCi/ $\rightarrow$ [CəCi]), while [-ATR] spreads rightward to high vowels (e.g., /CɪCi/ $\rightarrow$ [CɪCɪ]). Sixteen Komo listeners and 16 native English-speaking controls performed an AX task with disyllabic (pseudo)-words featuring context vowels and [$\pm$ATR] vowel continua. Patterns in the results suggest that (a) Komo listeners (but not controls) were influenced by the [ATR] value of the context, and (b) targets of [$+$ATR] harmony were processed differently from targets of [-ATR] harmony.

5aSCb16. Preliminary acoustic descriptions of the pharyngeals and sosterior plosives of Northern Haida. Corey Telfer and Jordan Lachler (Linguist, Univ. of AB, 2-40 Assiniboia Hall, Edmonton, AB T6G 2E7, Canada, telfer@ualberta.ca)

Haida is a highly endangered language spoken on the archipelago of Haida Gwaii, off the coast of British Columbia, as well as in communities in Southeast Alaska. Like many languages of the Northwest Coast, Haida plosives include three manners of articulation: voiceless aspirated, voiceless unaspirated, and ejective. Analysis of recordings indicates that the Voice Onset Time of the posterior aspirated and ejective stops is nearly identical. It appears that these speech sounds differ primarily in their acoustic intensities, and a new measure called Burst Intensity Slope is proposed to quantify this difference. In addition, the Northern dialect has been described as including pharyngeal speech sounds (e.g., Krauss 1979, Enrico 1991); however, only one small acoustic study has been conducted to verify this (Bessell 1993). Using recordings of a small number of speakers, this paper aims to document the different types of pharyngeals using acoustic measurements. Of special interest is the pharyngeal plosive of Massett Haida, which often includes what appears to be concomitant aryepiglottal trilling ("growl voice"). This will be investigated by comparing the number of zero-count crossings with those of other types of plosives and vowels produced by the same speakers.

5aSCb17. Prevoicing differences in Southern English: Gender and ethnicity effects. Wendy Herd, Devan Torrence (MS State Univ., 100 Howell Hall, PO Box E, MS State, MS 39762, wherd@english.msstate.edu), and Joy Cariño (MS School of Math and Sci., Columbus, MS)

Differences in the way VOT is used across languages to maintain stop voicing contrasts have been well-documented, but less research has been focused on VOT variation within voicing categories. For example, native English speakers are generally reported to produce word-initial voiced stops with short positive VOTs, but within category gender and ethnicity differences have been reported in one preliminary study, with male speakers prevoicing stops more than female speakers and with African American speakers prevoicing stops more than Caucasian American speakers (Ryalls, Zipprer, and Baldauff, 1997). For the current study, native speakers of English from Mississippi were recorded reading three repetitions of a pseudo-randomized list of words designed to investigate the effects of gender and ethnicity on the prevoicing of word-initial voiced stops. Participants self-identified their gender and ethnicity in a language background survey completed after recordings. Significant ethnicity, but not gender, differences were found. Strikingly, African American speakers produced voiced stops with prevoicing approximately 90% of the time, while just 35% of the voiced stops produced by European American speakers were prevoiced. These findings strongly suggest that dialectal differences play a role in within category variation in the VOTs of word-initial voiced stops.

5aSCb18. Weight-sensitive stress and acoustic correlates of disyllabic words in Marathi. Esther S. Le Grezause (Linguist, Univ. of Washington, 4300 Woodland Park Ave. N APT 203, Seattle, WA 98103, elg1@uw.edu)

Little of the literature on Marathi phonology addresses stress, and none of it systematically. The tentative accounts that do exist are contradictory in their descriptions (Pandharipande 1997, Dhongde and Wali 2009). Moreover, there is no research on the acoustic correlates of Marathi stress. To address this gap, this study investigates word-level stress and the acoustic correlates of stress in disyllabic words in Modern Standard Marathi. Since weight criteria vary across languages (Hayes 1995, Gordon 2004), the main goals are to determine what constitutes heavy syllables and whether or not long vowels behave like heavy vowels with two moras (μμ). Three main hypotheses are tested: (H1) stress is weight-sensitive, (H2) Marathi follows the Latin weight system (Gordon 2002) with CVX heavy, and (H3) when syllables have the same weight the left-most syllable bears stress. The study tests these hypotheses using auditory and spectrographic analyses of recordings from two native Marathi speakers. Results confirm all three hypotheses: H1: Marathi has weight-sensitive stress, H2: it follows the Latin weight distinction and, H3: the first syllable is stressed when syllable weight is equal. Preliminary acoustic results indicate that pitch, F1, and F2 are slightly higher in stressed syllables than in unstressed syllables.

5aSCb19. A preliminary analysis of stop codas in Kandze Khams Tibetan. Vitor Leongue (Linguist, Indiana Univ., 1021 East 3rd St. – Memorial Hall 322 E, Bloomington, IN 47405, vleongue@indiana.edu)

Research on Tibetan linguistics has been scarce. There are very few systematic accounts of the sounds in the language, and acoustic-phonetic analyses, in particular, are very difficult to come by. Furthermore, much of the existing literature has focused on the Lhasa dialect; many other varieties are not as adequately described. The current study investigates the phonetic realization of coda consonants in the under-documented Kandze dialect of Khams Tibetan. Historically, Tibetan allows up to two consonants in coda position, as reflected in the written language and in some western varieties. In most modern Tibetan dialects, however, many of the codas have been lost or simplified in some way. Preliminary data from one male native speaker indicate that codas that were historically voiced oral stops are realized in Kandze as glottal stops word-finally. When occurring word-internally, these codas tend to undergo deletion, which may then trigger lengthening of the preceding vowel. In both final and internal positions, some of the codas may also trigger vowel fronting. Acoustic data are presented to illustrate the alternations exhibited by the codas and the vowels that precede them.

5aSCb20. A preliminary account of the Thangal sound system. Patricia McDonough, Erin Arnold, and Kelly Berkson (Linguist, Indiana Univ., 1021 East 3rd St. – Memorial Hall 322 E, Bloomington, IN 47405, pmcdonou@indiana.edu)

This work presents instrumental acoustic analysis of Thangal, a severely under-resourced Tibeto-Burman language spoken primarily in the Senapati District of Manipur by around 2500 people. While some research has been conducted in recent years on related languages such as Rongmei, and several Thangal wordlists compiled mainly from data collected in the 1800s also exist, linguistic investigation of Thangal in the past century has been limited at best, and empirical phonetic investigation is nonexistent. We use data from wordlists produced by native Thangal speakers to compile a preliminary phonetic inventory of the language and map the Thangal vowel space. Initial notable observations include the existence of prenasalized plosives and both aspirated and unaspirated consonants. This work was undertaken at the request of the community, whose ultimate hope is to develop a practical orthography in service to developing written materials in Thangal.

5aSCb21. Affricate contrasts in Déline Slavey. Maida Percival (Linguist, Univ. of Toronto, 100 St. George St., 4th Fl., Toronto, ON M5S 3G3, Canada, maida.percival@mail.utoronto.ca)

Dene or North Slavey is a Dene (Athabaskan) language with nine affricates: /ts, tsʰ, ts’, tʃ, tʃʰ, tʃ’, tɬ, tɬʰ, and tɬ’/. This paper provides a first investigation into the acoustics of these sounds using ~2000 tokens from eight speakers from Déline, NT, Canada. Within the lateral series, [tɬ] was found to be realized in a number of ways including [tɬ], [tɬl], [tl], and [tɬɮ], with phonetic voicing to varying degrees in the release. In contrast, [tɬʰ] was generally produced with a brief period of aspiration following the frication portion of the affricate, while ejectives were often followed by a brief period of silence between the frication and the vowel onset. COG measurements of [tɬ] (from the voiceless portion of the release) were lower than the alveolar and post-alveolar affricates, averaging 3500 Hz (s.d. = 2175 Hz). However, COG variation across and within speakers and words suggests varying place of articulation. Place of articulation may be less crucial to the identification of these sounds than manner cues such as the lateral component. Further work is underway to investigate differences in the distribution of energy in the frication and F0 perturbations between the different phonation types, and to investigate the alveolar and post-alveolar affricates in more depth.

5aSCb22. Acoustic characteristics of vowels in two Totonacan languages. Rebekka Puderbaugh (Dept. of Linguist, Univ. of AB, 2-40 Assiniboia Hall, Edmonton, AB T6G 2E7, Canada, puderbau@ualberta.ca)

This study examines the acoustics of vowels from two Totonacan languages, Upper Necaxa Totonac (UNT) and Huehuetla Tepehua (HT). Both languages have five-vowel systems consisting of the qualities /aeiou/ as well as phonemic quantity distinctions (short, long), and lexical stress. In addition, UNT makes use of contrastive phonation on vowels, while HT vowels may be produced with allophonically non-modal phonation when adjacent to glottalic segments and glottal stops. Data from four speakers (two female and two male) of each language are reported. Acoustic measures are based on multiple repetitions of each vowel in a variety of lexical items, elicited within a frame sentence. Measurements are taken from stressed vowels surrounded by obstruents wherever possible. A variety of analyses are undertaken. Traditional visualizations of the vowel spaces of male and female speakers using normalized F1-F2 plots are combined with comparisons of dynamic formant trajectories across the time course of vowel production. Vowel duration is compared across short and long vowel categories of all five qualities in both languages. The relationship of fundamental frequency to vowel identity and phonation type is investigated.

5aSCb23. Arabic as an under-documented language: Distinctions between neighboring Arabic dialects. Judith K. Rosenhouse (Linguist, SWANTECH, 9 Kidron St., Haifa 3446310, Israel, judith@swantech.co.il)

This paper focuses on Arabic dialects from Israel and Jordan, which belong to the Levant Arabic dialects group. Levant dialects, as part of Arabic, are rather well studied. However, studies still usually consider them following the language situation in the area before the end of World War I

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(1918). Then, the Ottoman Empire collapsed and the area of (current) Israel and Jordan came to be under the British mandate rule. Later, each of them won independence as separate states. These processes widened inter-dialect linguistic differences, but modern communication media and devices, and increased literacy, have drawn various dialects closer to one another. These processes have added new linguistic features, which have in various cases completely integrated in the borrowing dialects. As a result, these features cannot be distinctive dialect markers any more. Such developments are mostly under-documented. Phonetic features of vowels and consonants in some Israeli and Jordanian Arabic dialects will demonstrate this phenomenon of under-documentation. In this context, we also discuss the forensic linguistics field of dialect identification and verification of asylum seekers' mother tongue. Our conclusions suggest that these undocumented changes require new investigations.

5aSCb24. The status of voiceless nasals in Miyako Ryukyuan. Catherine Ford, Benjamin V. Tucker, and Tsuyoshi Ono (Univ. of AB, 2-40 Assiniboia Hall, Edmonton, AB T6G2E7, Canada, bvtucker@ualberta.ca)

The current study investigates voiceless nasals in Miyako, a language spoken on remote Japanese islands near Taiwan, focusing on the dialect Ikema. Voiceless nasals are rare cross-linguistically. Hayashi (2013) suggested that there is phonemic contrast between voiced /n, m/ and voiceless /n̥, m̥/. Pilot data from one speaker both confirmed and refuted Hayashi's claims, analysis indicating that these voiceless nasals occur both as voiceless and breathy voiced. Thus, the glottal state of this phoneme may depend on the phonetic environment. The current study further explores these findings based on data collected from four male and two female speakers. Participants were asked to produce target and minimal pair words in isolation and within novel sentences. Voiceless and voiced nasal segments were compared and analyzed for duration, cepstral peak prominence, and amplitude measures of H1 and H2 to determine whether breathy voicing state might differentiate target phonemes from /n/ and /m/. The results of this analysis are used to investigate the nature of this phonemic contrast and how Ikema is situated cross-linguistically in its use of voicing to distinguish nasal phonemes.

5aSCb25. Pitch specification of minor syllables in Sgaw Karen. Luke West (Linguist, Univ. of California, Los Angeles, 785 Weyburn Terrace, Apt. 421, Los Angeles, CA 90024, lukewest@g.ucla.edu)

Sgaw Karen is a Tibeto-Burman language with both lexical tone and a major-minor syllable divide. Unlike major syllables, minor syllables are structurally reduced and do not bear overt phonological tone. This acoustic study investigates the fundamental frequency of these minor syllables, summarizing surface pitch patterns and providing evidence for a phonetic target. The first acoustic look at minor syllables in Karenic languages, a phonetically understudied language family, 58 unique minor syllable sequences of each possible length (~500 tokens) are elicited from one native speaker from the Karen State, Myanmar (Burma). Analysis of minor syllable sequences adjacent to lexical tones shows (1) a low pitch target around 185 Hz to which minor syllables asymptotically approximate when given sufficient time, and (2) this target is not explicable by contextual interpolation. Results address the general problem of tonal underspecification and interpret findings in the context of a minor syllable typology. Acoustic surface patterns of the Sgaw Karen minor syllable point to phonetic pitch targets of gradient strengths. Findings provide a foundation for detailed tonal/intonational analysis of Sgaw Karen, and demonstrates a framework for acoustic documentation of underspecified tonal units in understudied languages of Southeast Asian in general.

5aSCb26. On the presence of voiceless nasalization in apparently effaced Somali Chizigula prenasalized stops. Michal Temkin Martinez and Haley K. Boone (English, Boise State Univ., 1910 University Dr., Boise, ID 83725-1525, haleyboone@u.boisestate.edu)

Somali Chizigula (G311; xma; also Mushungulu) is an endangered language spoken in Somalia and by Somali-Bantu refugees abroad. We report on an acoustic and aerodynamic study of N + stop sequences in Somali Chizigula. Our findings illustrate that while the only acoustic cue for the nasal in voiceless prenasalized stops is aspiration of the oral stop, appearing to have undergone total effacement, aerodynamic data show robust nasal airflow prior to the oral stop. Although nasal devoicing is not rare in Bantu languages (e.g., Bura and Bondei; Maddieson and Ladefoged 1993), the lack of acoustic cues during the nasal is. The current state of the devoiced nasal in Somali Chizigula could lead to a reanalysis of effacement as [nt → nt^h → (**n̥t^h**)→ t^h]. In addition to providing acoustic and aerodynamic data, we propose an articulatory phonology (Browman and Goldstein 1993) account for the voiceless prenasalized stop as it compares to other nasal + stop sequences in the language.

5aSCb27. Word length is (in part) predicted by phoneme inventory size and syllable structure. Ian Maddieson (Dept. of Linguist, UNM, Univ. of New Mexico, MSC03-2130, Albuquerque, NM 87131-0001, ianm@berkeley.edu)

Based on anecdotal data from a few languages, it has been proposed that typical word length in a language is correlated with the size of the phoneme inventory: more phonemes predicts shorter words. This hypothesis has now been examined in some detail using a substantial sample of languages. A standardized text widely employed for phonetic illustrations is used, namely, the fable of "The North Wind and the Sun." Illustrations published by the IPA and others have been collected for over 100 languages. These illustrations are arguably well-matched in style and vocabulary and reflect natural inflected forms embedded in text. Average word-length in these texts is calculated and correlated with the size of the language's consonant and vowel inventories, as well as an index reflecting the permitted complexity of syllable structure. Vowel inventory is treated in two ways; basic vowels (just the distinctions on the basic height, backness and rounding parameters), and total vowels (including other distinctions such as nasalization, length, and voice quality). Mean word length is indeed negatively correlated with size of consonant and vowel inventories, most strongly with the number of basic vowels. More complex syllable structures also correlate with a shorter mean word length.

Session 5aSCc

Speech Communication: Speech Production (Poster Session)

Maria V. Kondaurova, Chair
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All posters will be on display from 9:00 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 9:00 a.m. to 10:30 a.m. and contributors of even-numbered papers will be at their posters from 10:30 a.m. to 12:00 noon.

5aSCc1. Simulations of three-dimensional, self-oscillating vocal fold replicas with liquid-filled cavities. Eduardo J. Alvarez and Scott L. Thomson (Mech. Eng., Brigham Young Univ. - Idaho, BYU-Idaho, Rexburg, ID 83460, Alv14011@byui.edu)

Synthetic, self-oscillating vocal fold replicas are used in bioreactors to study the response of injected cells to phono-mimetic vibrations. In this type of replica, cells are contained within a cylindrically shaped cavity that runs anterioposteriorly and that is located near the replica surface. During vocal fold replica flow-induced vibration, the cavity deforms, thereby subjecting the cells to a periodically varying stress field. The characteristics of this stress field are unknown, and experimental determination is not presently feasible. Therefore, in this research, numerical simulations of these self-oscillating, cell-containing synthetic replicas are studied. A three-dimensional vocal fold model containing a liquid-filled cavity, developed using the commercial software package ADINA, is described. The effects of glottal airflow are modeled using the mechanical energy equation to simulate the pressure distribution along the replica airway surface. The liquid-filled cavity is modeled through a fully coupled fluid-structure interaction solver, with the liquid region governed by the viscous, unsteady, three-dimensional Navier-Stokes equations. Estimates of the stress field within the cavity as well as the predicted influence of the cavity on replica vibration will be reported.

5aSCc2. What the 'L?: An ultrasound study of the acoustic and articulatory characteristics of laterals in Brazilian Portuguese. Sherman D. Charles (Linguist, Indiana Univ., 1610 S Dorchester Dr. Apt. 49, Bloomington, IN 47401, sdcharle@indiana.edu)

Recent technological and computational advances have enabled researchers to investigate vocal tract articulation and acoustics with an ever-increasing degree of nuance and specificity. This is particularly enlightening in the case of articulations which are subject to tremendous cross-linguistic and interspeaker variability, like laterals (Ladefoged and Maddieson 1996). The current study uses simultaneous 3D ultrasonography and audio recordings to investigate vocal tract configurations and their acoustic consequences, and to provide a comprehensive descriptive analysis of alveolar and palatal lateral speech sounds produced by one Brazilian Portuguese speaker. The results confirm previous accounts of multiple articulatory constrictions and low F1 and F2 frequencies for alveolar laterals (Barbarena *et al*. 2014; Johnson 2012; Ladefoged 2003; Ladefoged and Maddieson 1996). Novel contributions include (1) identification of a dynamic tongue gesture that, based on a Perturbation Theory analysis (Stevens 1996), appears to directly reflect formant transitions from a preceding vowel to the alveolar lateral consonant, and (2) three-dimensional articulatory descriptions of both alveolar and palatal laterals in Brazilian Portuguese.

5aSCc3. Articulator movement contributions to formant trajectories in diphthongs. Christopher Dromey and Katherine McKell (Commun. Disord., Brigham Young Univ., 133 TLRB, BYU, Provo, UT 84602, dromey@byu.edu)

Our purpose was to learn about the relative contributions of lip aperture and protrusion as well as tongue height and advancement to the trajectories of F1 and F2 during diphthongs produced by 20 speakers. Audio and kinematic signals were recorded with an NDI Wave system. Audio segments were analyzed with PRAAT to extract the F1 and F2 histories during the diphthongs. These were time-aligned with the kinematic records in MATLAB. Because time-series data violate the assumptions of traditional correlation and regression analyses, novel methods were developed to represent the relative contributions of individual articulator movements to F1 and F2 changes. All movement and formant tracks during diphthong transitions were converted to z-scores and overlaid on a common time scale. Absolute difference scores between kinematic (predictor) variables and acoustic (dependent) variables were summed along each plot to reflect the strength of the contribution of each movement to the acoustic outcome. The lowest difference sums reflected the closest alignment between a movement and a formant track. Results revealed differences in the relative contributions of the tongue and lips for each diphthong, suggesting that formant patterns during diphthongs may be difficult to interpret in a straightforward way in terms of articulator movement.

5aSCc4. Rhotic articulation by first graders: A real-time three-dimensional ultrasound study. Olivia Foley, Sarah McNeil, Katherine Schlimm, Amy Piper, and Steven M. Lulich (Speech and Hearing Sci., Indiana Univ., 200 S. Jordan Ave., Bloomington, IN 47405, oliviarfoley@gmail.com)

The rhotic /r/ in American English is a complex sound that is produced differently by different speakers and in different phonetic contexts, but there is currently little knowledge about how children articulate this sound, which is typically acquired late in development. In this study, the Goldman-Fristoe Test of Articulation (GFTA-3) was administered to typically developing first graders (6 and 7 years old) as part of a 5-year longitudinal study. The GFTA-3 contains words in which /r/ occurs in a variety of syllable positions and phonetic contexts. Using 3D ultrasound imaging and palate impressions, it is possible to view the tongue shape in relation to the speaker's palate. Since first graders are younger than the age at which /r/ is usually completely acquired, this presents an opportunity to see how /r/ is articulated by children who may not yet have mastered it. Initial results from this study will be presented, with the aim of empirically characterizing tongue shape and motion in /r/ production by first graders.

5aSCc5. Velopharyngeal status of vowels produced with and without hard glottal attack in children with repaired cleft palate. Marziye Eshghi (Speech, Lang. and Hearing Sci., Craniofacial Ctr., Univ. of North Carolina at Chapel Hill, 002 Brauer Hall, Chapel Hill, NC 27599, marziye_eshghi@med.unc.edu) and David J. Zajac (Dental Ecology, Univ. of North Carolina at Chapel Hill, Chapel Hill, NC)

The objective of this study was to describe velopharyngeal (VP) closure patterns of isolated vowels produced by infants with repaired cleft palate (CP). Nasal ram pressure (NRP) and audio signals were obtained from six infants with repaired CP (four males and two females). Two infants were 12 months of age and four were 18 months of age. All infants were from American-English speaking families without known syndromes. A total number of 95 isolated vowels were analyzed perceptually and spectrographically. Fourteen vowels were identified as produced with hard glottal attack. Three

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infants produced all vowels with at least 96% complete VP closure. Three other infants produced all vowels with 59% complete VP closure and 39% partial VP closure. In addition, one of the infants produced an isolated vowel with a completely open velopharynx. Vowels with hard glottal attack, however, were produced with 100% complete VP closure across all subjects, regardless of VP status for vowels without hard glottal attack. It is hypothesized that early use of hard glottal attack by children with repaired CP might contribute to the often reported high prevalence of voice disorders and may even be associated with the development of glottal stop articulation. [Research reported in this publication was supported by the National Institute of Dental & Craniofacial Research of the National Institutes of Health under Award Number 1R01DE022566-01A1.]

5aSCc6. Event-related cortical potentials occurring prior to speech initiation. Silas Smith, Maira Ambris, Serena Haller, and Al Yonovitz (Dept. of CSD, Univ. of Montana, Missoula, MT 59812, silas.smith@umontana.edu)

Real time brain electrical potentials were obtained in subjects prior to the initiation of speech in a potential clinical paradigm using a single vertex electrode. The purpose of this research was to establish a real-time event related brain electrical potential system. Research in this area in the past has used a great number of facial and lip EMG electrodes. The marking point for determining the pre-event time epoch has been an EMG source. The data are typically acquired off-line and later averaged. This research uses a vocal signal as the marking point and displays in real time the event-related potential. The sample rate (25,600 samples/s) permitted an analysis of both slow negative waves and faster neurogenic signals. Results indicated reliable waveform morphology within and between subjects.

5aSCc7. Nasal coarticulation in normal infants: A physiological analysis. Marziye Eshghi (Speech, Lang. and Hearing Sci., Craniofacial Ctr., Univ. of North Carolina at Chapel Hill, 002 Brauer Hall, Chapel Hill, NC 27599, marziye_eshghi@med.unc.edu) and David J. Zajac (Dental Ecology, Univ. of North Carolina at Chapel Hill, Chapel Hill, NC)

There are conflicting views regarding coarticulation in speech of children. Kent (1983) suggested that children's speech is more "segmental" while Nittrouer *et al.* (1989) proposed that speech of young children has "syllabic organization" with syllables being units of speech production. According to Nittrouer *et al.* (1989), children may demonstrate more coarticulation than adults. The purpose of this study was to examine nasal coarticulation in the speech of infants over time. Nasal ram pressure (NRP) and audio signals were obtained from ten typically developing infants (five males and five females) at 12, 14, and 18 months of age. Six subjects produced 39 nasal syllables (either NV, VN, NVN, or VNV) at 12 months of age. Six subjects produced 62 nasal syllables at 14 months of age and eight subjects produced 95 nasal syllables at 18 months of age. Results revealed positive NRP at the midpoint of only 20% of vowels either preceding or following nasal consonants at 12 months of age. Positive NRP, however, was present during 62% and 77% of vowels produced at 14 and 18 months of age, respectively. Findings provide evidence for the development of nasal coarticulation over time from a more "segmental" to a "syllabic" level. {Research reported in this publication was supported by the National Institute of Dental & Craniofacial Research of the National Institutes of Health under Award Number 1R01DE022566-01A1.]

5aSCc8. Muscular structure of the human tongue from magnetic resonance image volumes. Maureen Stone (Univ. of Maryland Dental School, 650 W. Baltimore St., Rm. 8207, Baltimore, MD 21201, mstone@umaryland.edu), Jonghye Woo (Massachusetts General Hospital, Boston, MA), Junghoon Lee (Johns Hopkins School of Medicine, Baltimore, MD), Tera Poole, Amy Seagraves, Michael Chung, Eric Kim (Univ. of Maryland Dental School, Baltimore, MD), Emi Z. Murano (Hospital das Clinicas da Faculdade de Medicina, Sao Paulo, Brazil), Jerry L. Prince (Johns Hopkins Univ., Baltimore, MD), and Silvia S. Blemker (BME, Univ. of Virginia, Charlottesville, VA)

The human tongue has a complex architecture, consistent with its complex roles in eating, speaking, and breathing. Tongue muscle architecture has been depicted in drawings and photographs, but not quantified volumetrically. This study aims to fill that gap by measuring the muscle architecture of the tongue for 14 people captured in high-resolution 3D MRI volumes. The results show the structure, relationships, and variability among the muscles, as well as the effects of age, gender, and weight on muscle volume. Since the tongue consists of partially interdigitated muscles, we consider the muscle volumes in two ways. The functional muscle volume encompasses the region of the tongue served by the muscle. The structural volume halves the volume of the muscle in regions where it interdigitates with other muscles. Results show similarity of scaling across subjects, and speculates on functional effects of the anatomical structure.

5aSCc9. An electromagnetic articulometer study of tongue and lip troughs. Christine H. Shadle (Haskins Labs., 300 George St., New Haven, CT 06511, shadle@haskins.yale.edu), Hosung Nam (Haskins Labs., Seoul, South Korea), Argyro Katsika, Mark Tiede, and D H Whalen (Haskins Labs., New Haven, CT)

Troughs, which are a discontinuity in anticipatory coarticulation (Perkell 1968), have been shown to occur in the tongue body in /ipi/ and lips in /usu/ . An electromagnetic articulometer (EMA) study of a single subject reported previously showed that intraoral pressure during the consonant could not be solely responsible for the downwards tongue body movement observed during the labial consonants (/p, b, m, f, v/); results supported the hypothesis of a secondary tongue gesture during the consonant. Lip troughs during /s/ in rounded contexts were more ambiguous, possibly because of inconsistencies in corpus design. Six new subjects have been recorded using a WAVE EMA system and a revised corpus. Tongue trough magnitudes decreased in the order /f, v/ $>$ /p, b/ $>$ /m/, serving to disprove the aerodynamic hypothesis; they were deeper for long than short consonants (e.g., [iffi] $>$ [ifi]), and converged to the same point in asymmetric vowel contexts. The same type of convergence during the consonant was observed for lip aperture during $V_1C(C)V_2$ combinations, where $C = \{/s, ʃ/\}$. These recent results thus support both tongue and lip troughs being caused by secondary gestures of consonants. [Work supported by NIH NIDCD-DC-02717.]

5aSCc10. Underlying mechanism for nasal rustle noise during speech in children with cleft palate. Hedieh Hashemi Hosseinabad, Suzanne Boyce (Dept. of Commun. Sci. and Disord., Univ. of Cincinnati, 3202 Eden Ave., Cincinnati, OH 45267-0379, hashemhh@mail.uc.edu), and Ann W. Kummer (Div. of Speech-Lang. Pathol., Cincinnati Children's Hospital Medical Ctr., Cincinnati, OH)

Nasal rustle (AKA Nasal turbulence) occurs as an obligatory nasal air emission in children with cleft palate. It usually occurs as a result of air passing through a partially small velopharyngeal port. As the air moves through this small opening, the air pressure increases. When this pressurized air is released on the nasal side of the opening, it causes nasal secretions on top of the valve to bubble. The hypotheses is that nasal rustle is due to audible bubbling of secretions; so when secretions are removed, the extra noise disappears. In this study, we applied signal to noise ratio (SNR) before suctioning the secretions to explore the relationship between presence of noise and bubbling. Ten children with nasal rustle were recorded using the microphones of the Nasometer II during production of high pressure sentences pre-post suctioning. Results will be discussed. It is expected that the findings reveal that using SNR can be used clinically as an acoustic parameter to detect the effect of bubbling in nasal rustle.

5aSCc11. Complete tonal neutralization in Taiwan southern Min. Mao-Hsu Chen (Linguist, Univ. of Pennsylvania, 4200 Spruce St., Apt. 310, Philadelphia, PA 19104, chenmao@sas.upenn.edu)

This study reported the result of a production experiment on the tonal neutralization of context unchecked 21 tone and checked 21 tone with a glottal stop coda in Taiwan Southern Min, which are said to be realized the same as a high falling sandhi tone 51 on surface when occurring in context positions. Speakers of two age groups, younger and older, were recruited for the production experiments to examine whether the neutralization is complete. The comparison between the f0 contours was conducted with smoothing spline analysis of variance, and it showed a case of complete neutralization for both age groups, where the two f0 contours were simply

overlapped. As reported by linear mixed effects modeling, the durations of the target syllables with underlying checked and unchecked 21 tones were not significantly different from each other.

5aSCc12. Interaction between aspiration and tonal pitch in Weihai Chinese. Jie Deng and Sean A. Fulop (Linguist, California State Univ. Fresno, 5245 N Backer Ave., PB92, Fresno, CA 93740-8001, sfulop@csufresno.edu)

This paper considers the questions of whether, how, and by what mechanism the aspiration of voiceless stops in the syllable onset affects the pitch (F0) of lexical tone in a tone language. The literature on this topic is at odds with itself. Lai *et al.* ["The raising effect of aspirated prevocalic consonants on F0 in Taiwanese," *Proceedings of the 2nd International Conference on East Asian Linguistics* (2009)] review the history of conflicting results, with some studies reporting F0 lowering after aspirated stops, and some reporting F0 raising. Here is provided another battery of findings from another tonal dialect, Weihai Chinese. Using data from seven monodialectal female speakers, no effect of aspiration is found on tonal F0 following bilabial or alveolar stops, but a strong significant lowering effect was observed following the velar aspirated stops. A second issue addressed here has had almost nothing published about it, viz., what is the converse effect of the lexical tone pitch upon the aspiration time (VOT) of the onset stops. It was found that, in general, high-beginning tones correlated with shorter voice onset times in all stop onsets.

5aSCc13. The role of f0 on acquisition of a phonological contrast in Korean stop system. Gayeon Son (Linguist, Univ. of Pennsylvania, 619 Williams Hall, 255 S 36th St., Philadelphia, PA 19104, gson@ling.upenn.edu)

A number of studies have investigated the role of Voice Onset Time (VOT) on acquisition of stop voicing contrast. Korean stop contrasts (lax, tense, and aspirated), however, cannot be differentiated only by VOT since they are all pulmonic egressive voiceless stops. For this three-way distinction, another acoustic parameter, f0, critically operates. The present study explores how f0 is perceptually acquired and phonetically operates for Korean stop contrast over age. In order to reveal the relationship between f0 developmental patterns and age, a quantitative acoustic model dealt with word-initial stop productions by 60 Korean young children aged 20 months to 47 months. The results showed that f0 perceptual acquisition patterns are closely related to children's articulatory distinction especially between lax and aspirated stops. In the case of aspirated stops, phonetic accuracy depends on the perceptual thresholds in f0, and the significant phonetic differentiation between lax and aspirated stops was found at age over 32 months. However, f0 of tense stops is not significantly distinguished from the other two stop categories in production. These findings suggest that acquisition of f0 plays a crucial role in the formation of phonemic categories for lax and aspirated stops and this process significantly affects articulatory distinction.

5aSCc14. The physiological underpinnings of vowel height and voice quality. Laura Panfili (Linguist, Univ. of Washington, Box 352425, Seattle, WA 98195, lpanfili@uw.edu)

This study examines the distribution of creaky voice across vowel heights and discusses the physiological mechanisms that may influence this distribution and its methodological implications. The data for this study come from eight dyads from the ATAROS Corpus of audio-recorded conversations between Pacific Northwest English speakers (Freeman 2015). Stressed vowels in content words were tagged for phonation type based on auditory judgments. A chi square test of independence ($\alpha = 0.01$) found a significant relationship between vowel height and creak ($\chi^2(1, N = 2459) = 83.58, p < 0.001$), such that low vowels were more likely to be creaky than high vowels. This effect may be due to the same physiological mechanisms as Intrinsic Fundamental Frequency (IF0). Though the exact mechanism behind IF0 is unclear, various hypotheses have suggested that the tongue position required for high vowels pulls on the larynx, increasing tension, decreasing mass, and resulting in a higher F0 (Ladefoged 1964, Lehiste 1970). These laryngeal settings would also disfavor creaky voicing, perhaps explaining the distribution of creaky voice and vowel height. Future studies of phonation should consider vowel height in their methodologies.

5aSCc15. Representations of electromagnetic articulography data for tongue shaping and vocal tract configuration. Sungbok Lee (Elec. Eng., Univ. of Southern California, 3601 Watt Way, GFS-301, Los Angeles, CA 90089, sungbokl@usc.edu), Dani Byrd (Linguist, Univ. of Southern California, Los Angeles, CA), and Shrikanth Narayanan (Elec. Eng., Univ. of Southern California, Los Angeles, CA)

Electromagnetic articulography (EMA) measurements of the movements of articulator flesh points have predominantly been used to investigate kinematics of individual points and/or coordinative phasing between them. Such limited use of EMA data is in fact wasteful of information. In this preliminary report we introduce two representations of EMA data: one is a representation of lingual configuration derived from three individual flesh points on the tongue, and the other is a representation of the vocal tract configuration information based on a lower-dimensional representation of Euclidean distances between EMA articulatory points. The motivation is to maximize the use of information in the original EMA data and measurements in a way that allows for the representation of tongue shaping and vocal tract configuration, thereby maximizing the use of information available from EMA and basic EMA measurements. Implications of such representations and their patterning will be discussed in the light of speech production dynamics as functions of speech rate and categorical emotions expressed in speech. [Work supported by NIH DC03172 and NSF IIS-1116076.]

5aSCc16. Measuring contact area in synthetic vocal fold replicas using electrical resistance. Kyle L. Syndergaard, Stephen Warner, Shelby Dushku, and Scott L. Thomson (Mech. Eng., Brigham Young University-Idaho, BYU-Idaho, Rexburg, ID 83460, kyle.syndergaard@gmail.com)

Synthetic vocal fold replicas are a useful tool for studying human voice production. These replicas are often studied using high-speed imaging, but additional analysis procedures are desired that are lower cost, less data intensive, and more conducive to long-term vibration monitoring. Consequently, this work explores the use of electrical resistance to measure vocal fold contact area across the glottis of a synthetic vocal fold replica. The concept is based on electroglottography (EGG), a clinical tool for measuring vocal fold contact area *in vivo*. When a small current is passed through the vocal folds, the resistance across the folds varies inversely with the contact area between the folds. This change in resistance thus makes it possible to estimate the degree of glottal closure during vocal fold vibration. For this research, a method of enabling silicone vocal fold replicas to conduct electricity without compromising the sensitive material viscoelastic properties has been developed. The concept will be demonstrated and relationships between contact area and resistance in both static and vibrating (self-oscillating) replicas will be presented.

5aSCc17. Prosodic strengthening at the edges of prosodic domains in sighted and blind speakers. Lucie Menard, Pamela Trudeau-Fisette, and Melinda Maysounave (Linguist, Universite du PQ a Montreal, CP 8888, succ. Centre-Ville, Montreal, QC H3C 3P8, Canada, menard.lucie@uqam.ca)

Recent studies have shown that when producing isolated vowels, congenitally blind speakers produce smaller displacements of the lips (visible articulators), compared with their sighted peers. To investigate the role of visual experience on articulatory gestures used to produce salient speech contrasts, the production of vowels at the edges of low-level prosodic domains and high-level prosodic domains (in which gestures are reported to be strengthened) was studied in adult speakers of Quebec French. Ten sighted and ten congenitally blind participants were recorded during the production of the vowels /i/, /y/, /u/, /a/ in initial and final positions of two prosodic domains: word and intonational phrase. Synchronous acoustic and articulatory data were recorded using the Carstens AG500 Electromagnetic Articulograph system. Formant measures as well as displacements of the lips and tongue were analyzed. The results revealed that speakers produced larger ranges of lip and tongue movements at the edges of higher prosodic domains than at the edges of lower ones. Formant spaces and durational data are presented. The use of visible (lip and jaw) and invisible (tongue) articulators to implement the prosodic structure is discussed

5aSCc18. Three-dimensional tongue shapes of /r/ production in American English words. Steven M. Lulich (Speech and Hearing Sci., Indiana Univ., 200 South Jordan Ave., Bloomington, IN 47405, slulich@indiana.edu), Brandon Rhodes (Linguist, Univ. of Chicago, Chicago, IL), Max Nelson, Kelly Berkson, and Kenneth de Jong (Linguist, Indiana Univ., Bloomington, IN)

A number of studies have shown that /r/ production in American English involves complex tongue shapes. Previous 3D imaging studies, however, have been limited to sustained (static) /r/ sounds produced in supine position. Since supine versus upright posture and static versus dynamic speech production influences the shape of the tongue, it is unclear to what degree previous findings of three-dimensional tongue shapes are generalizable to /r/ sounds produced dynamically and with upright posture. This study presents upright-posture 3D tongue shapes of dynamically produced /r/ sounds from words embedded in a carrier phrase. Twenty young adult native speakers of American English participated in the study (10 males and 10 females), and analyses are ongoing.

5aSCc19. Articulatory targets for ultrasound biofeedback determined by tracking regional tongue displacements. Sarah M. Hamilton, T. Douglas Mast, Michael Riley, and Suzanne Boyce (Commun. Sci. and Disord., Univ. of Cincinnati, 3433 Clifton Ave., Cincinnati, OH 45220, hamilsm@mail.uc.edu)

Ultrasound biofeedback therapy (UBT) is a significant alternative to traditional therapy for speech disorders, but some users make little progress with the standard feedback display. For segments such as American English /r/, the required tongue movements are so complex that an ultrasound image of the tongue cannot adequately guide speakers to make the right movements. We propose to develop visual display regimes for ultrasound biofeedback that maximize speaker learning. As a preliminary step, we determine displacement ranges of tongue regions (blade, root, and dorsum) known to characterize accurate /r/ production, which will ultimately be used with robust image processing techniques to drive an improved UBT feedback display. Though blade, root, and dorsum movement are known to characterize /r/ production (Boyce *et al*., 2011; Espy-Wilson *et al*., 2000; Zhou *et al*., 2008), excursion of these tongue parts differs across individuals. Here, regional tongue displacements were tracked based on measured motion of local brightness maxima in low-pass-filtered midsagittal ultrasound images, normalized by reference anatomical landmark distances. Preliminary results show that the midsagittal dorsum, root, and blade show more extreme displacements relative to each other for the correct /r/ versus misarticulated /r/. This result is consistent with clinical observations of "humped" error /r/ tongue shapes.

5aSCc20. Models and methods for exploring anisotropy and inhomogeneity in vibrating vocal fold tissue. Ryan M. Oakey and Scott L. Thomson (Dept. of Mech. Eng., Brigham Young Univ. - Idaho, BYU-Idaho, Rexburg, ID 83460, rya07002@byui.edu)

Synthetic vocal fold models have long been used to study features of human voice production that cannot reasonably be studied *in vivo*. Numerous models have been developed to study different aspects of voice production. Some models have focused, for example, on mimicking geometry or tissue layering while others have been used to study the effect of vibration on live cells. Two aspects of vocal fold structure that have yet to be fully explored are the anisotropic and the inhomogeneous qualities of vocal fold tissue. These features significantly influence vocal fold flow-induced vibratory characteristics. The purpose of this research is to develop and explore models that incorporate anisotropy and inhomogeneity to study their effects on vocal fold vibration. In this presentation, methods and models for studying these features will be discussed; one example is the creation of a fiber matrix via rotary jet spinning that is embedded within silicone vocal fold models to study material anisotropy. Fabrication methodology and model response results will be presented, along with corresponding implications and possibilities for future use in the study of human voice production.

5aSCc21. Asymmetry in incomplete neutralization. Abby Kaplan (Linguist, Univ. of Utah, LNCO, Rm. 2300, 255 S Central Campus Dr., Salt Lake City, UT 84112, abby.kaplan@utah.edu)

Neutralizing patterns, such as final devoicing, are known to often be incomplete: "devoiced" final obstruents in languages such as Afrikaans may retain, e.g., longer preceding vowels than their voiceless counterparts. A common explanation for this phenomenon is that these partially devoiced obstruents are influenced by paradigmatically related obstruents that are fully voiced (e.g., Afrikaans *hoe[t]* ~ *hoe[d]e*, "hat(s)"). This proposal raises the question of whether such influence goes in the other direction too: is the [d] of *hoe[d]e* slightly devoiced, compared to a non-alternating [d] as in *roe[d]e* ("rod")? The experiment presented here tests this hypothesis in 28 Afrikaans nouns as produced by nine native speakers; vowel length, closure duration, release duration, and glottal pulses were measured as cues for voicing. The results provide no evidence for partial devoicing of voiced stops under the influence of devoiced counterparts elsewhere in the paradigm; that is, the [d] of *hoe[d]e* is no less voiced than the [d] of *roe[d]e*. I conclude that incomplete neutralization is an asymmetrical phenomenon: segments subject to neutralization, as in *hoe[t]*, may retain some contrasting cues; but these segments do not in turn encourage partial neutralization in morphologically related forms such as *hoe[d]e*.

5aSCc22. Efficiency of synthetic excitation obtained by interference of ultrasonic waveforms for reducing background noise for laryngeotomee. Romilla M. Bhat (Dept. of Electronics, Govt. Gandhi Memorial Sci. College, 9/C Om Nagar Udeywalla, Jammu, Jammu and Kashmir 180002, India, romillarosette@gmail.com) and Parveen K. Lehana (Dept. of Phys. and Electronics, Univ. of Jammu, Jammu, India)

Interference pattern of simulated ultrasonic waves using MATLAB(free version R2010a) using the laws of acoustics has been used to get excitation in the audio frequency range. Preliminary experiments have been done using different high frequencies and the recordings have been done in Goldwave 5.1 version and then subsequent analysis have been done in PRAAT to analyze the beat frequency obtained by interfering two high frequency (above 15 kHz) waves. The output thus obtained has been then subjected to Hilbert Transform for envelope detection using MATLAB(free version R2010a) to get the audio excitations. These audio excitations are further articulated by the glottal passage for speech synthesis for laryngeotomee or for producing alaryngeal speech. The results of simulation have been further utilised to fabricate a electronic circuit for generation of two continuous ultrasonic waves which are focussed to interfere inside the glottal tract, thereby producing a beat frequency in the audible range and reduce background noise. Index Terms: speech synthesis, ultrasonic waves, laryngeotomee Hilbert transform

5aSCc23. Testing a conceptual model of vocal tremor: Respiratory and laryngeal contributions to acoustic modulation. Jordon LeBaron and Julie Barkmeier-Kraemer (Univ. of Utah, 390 S. 1530 E., Rm. 1201 BEH SCI, Salt Lake City, UT 84112, jordon.lebaron@utah.edu)

Vocal tremor is a voice disorder characterized by rhythmic modulation of fundamental frequency (f_0) and sound pressure level (SPL) during sustained phonation. To date, contributions of oscillating speech structures to the acoustic modulations of vocal tremor are absent in the literature and are important for treatment purposes. The purpose of this study was to prospectively test the contributions of laryngeal and respiratory structure oscillations to acoustic modulation using singers to simulate vocal tremor. Laryngeal oscillation during production of natural vibrato was hypothesized to associate with f_0 modulation, whereas respiratory system oscillation during respiratory accented voicing was hypothesized to associate with SPL modulation. Ten female singers were recruited between 40 and 65 years of age without voice problems and meeting inclusion criteria for expertise necessary to produce natural vibrato and respiratory accented voicing during sustained phonation. Simultaneous endoscopic views of the larynx, respiratory kinematic, and acoustic signals were recorded during three trials of sustained /i/ during natural vibrato, or respiratory accented voicing. The rate and magnitude of kinematic movements of the chest wall and larynx are currently being compared to rate and magnitude of acoustic modulation. Preliminary qualitative evaluation of data supports hypothesized contributions of speech structures to acoustic modulation patterns.

5aSCc24. Normalizing nasality? Across-speaker variation in acoustical nasality measures. Will Styler (Dept. of Linguist, Univ. of Michigan, 611 Tappan St., 440 Lorch Hall, Ann Arbor, MI 48109, wstyler@umich.edu)

Although vowel nasality has traditionally been investigated using articulatory methods, acoustical methods have been increasingly used in the linguistic literature. However, while these acoustical measures have been shown to be correlated with nasality, their variability across-speakers is less-well investigated. To this end, we examined cross-speaker variation in coarticulatory vowel nasality in a large collection of multi-speaker English data, comparing measurements in CVC and NVN words at two points per vowel. Two known correlates were analyzed: A1-P0, where A1 is the amplitude of the harmonic under F1, and P0 is the amplitude of a low-frequency nasal peak (Chen 1997), and the bandwidth of F1 (Hawkins and Stevens 1985). Speakers varied both in terms of baseline measurements (e.g., the mean measurements in oral versus nasal vowels), and in the oral-nasal range (the measured difference between oral and nasal). This suggests that although analyzing centered within-speaker differences is unproblematic, the comparison of raw nasality measurements across speakers is unlikely to yield meaningful data. Moreover, this suggests that the perception of vowel nasality may require some process of speaker normalization, much like other aspects of vowel perception. Some suggestions for normalizing nasality measurements in research will be offered.

5aSCc25. A study on drinking judgment using phonation characteristics in speech signal. Won-Hee Lee and Myung-Jin Bae (SoongSil Univ., Hyungnam Eng. Bldg. 1212, Soongsil University 369 Sangdo-Ro, Dongjak-Gu, Seoul 156-743, South Korea, vbluelovev@ssu.ac.kr)

In this study, speech characteristics from before and after alcohol intoxication have been comparatively analyzed through speech analysis to obtain the degree of intoxication along with its parameters. The speech characteristics of an alcohol intoxicated person, such as pitch and formant and changing of sound levels, have been studied, but the speech characteristics before and after alcohol intoxication cannot be analyzed because of the sensitiveness of environmental changes. Therefore, we need to find more precise parameters, especially Phonation threshold pressure, the high levels of which result in dehydration of the laryngeal mucous membrane after alcohol intoxication. Speech after alcohol intoxication is with low lung pressure and decreasing lung capacity. Thus, we study the speech characteristics from before and after alcohol intoxication using speech-rate in speech signal and lung capacity.

FRIDAY MORNING, 27 MAY 2016 — SALON G, 8:00 A.M. TO 11:45 A.M.

Session 5aSP

Signal Processing in Acoustics: General Topics in Signal Processing

Edmund Sullivan, Cochair
Research, Prometheus, 46 Lawton Brook Lane, Portsmouth, RI 02871

Brian E. Anderson, Cochair
N145 Esc, Brigham Young Univ., MS D446, Provo, UT 84602

Contributed Papers

8:00

5aSP1. Characterizing nonlinear systems with memory while combating and reducing the curse of dimensionality using new Volterra Expansion Technique. Albert H. Nuttall (Sonar and Sensors Dept., NUWCDIVNPT (retired), Old Lyme, CT), Derke R. Hughes, Richard A. Katz, and Robert M. Koch (NUWCDIVNPT, 1176 Howell St., Newport, RI 02841, derke.hughes@verizon.net)

A generalized model for characterizing nonlinear systems was originally proposed by Italian mathematician and physicist Vito Volterra (1860-1940). A further developed by American mathematician and MIT Professor Norbert Wiener (1894–1964) and published in 1958. After direct involvement with Norbert Wiener publication, Albert H. Nuttall has recently made new inroads along with his coauthors in applying the Wiener-Volterra model. A general description of a nonlinear system to the second order is termed the Nuttall-Wiener-Volterra model (NWV) after its co-founders. In this formulation, two measurement waveforms on the system are required in order to characterize a specified nonlinear system under consideration: an excitation input, x(t) (the transmitted signal) and a response output, z(t) (the received signal). Given these two measurement waveforms for a given system, a kernel response, h= [h0,h1,h2,h3] between the two measurement points, is computed via a least squares approach that optimizes modeled kernel values by performing a best fit between a measured response z(t) and a modeled response y(t). New procedures developed by A. Nuttall are invoked to significantly diminish the exponential growth of the computed number of kernel coefficients with respect to third order and higher orders to combat and reasonably reduce the curse of dimensionality.

8:15

5aSP2. Wigner cross-products encode relative phases. James B. Lee (None, 6016 S. E. Mitchell, Portland, OR 97206, cadwal@macforcego.com)

Two distinct kinds of elements comprise the Wigner time-frequency distribution. First are elements occurring at the actual times and frequencies present in the signal: call these “actual” elements. Second are elements occurring at the arithmetic mean of the actual elements: call these “ephemeral” elements. The actual elements essentially are a power spectrum resolved in time and frequency; always they are positive. The ephemeral elements, sometimes called “cross-products,” can be positive, negative, both, or zero, depending on the relative phases of the actual elements; they encode phase information necessary to complete time-frequency representation of all information in the original signal.

8:30

5aSP3. The blind source separation and classification of diver's noise based on adaptive beamforming and frequency selection. Juan Yang (Inst. of Acoust., Chinese Acad. of Sci., No. 21 North 4th Ring Rd., Haidian District, Beijing 100190, China, yang_juanacoustics@163.com), Karim G. Sabra (School of Mech. Eng., Georgia Inst. of Technol., Atlanta, GA), Feng Xu, Xudong An, Hao Tang, Lei Jiang, and Peng Li (Inst. of Acoust., Chinese Acad. of Sci., Beijing, China)

This paper introduces an algorithm to separate and classify the diver's noise in the presence of other noise interferers. The passive detection and classification for divers has been discussed before by using the breathing rate as features. But the performance of this approach significantly diminishes when other noise sources existing simultaneously, especially the other divers or louder shipping noises. In this paper, the wideband MVDR is used as an adaptive beamformer for horizontal linear array to better discriminate the azimuth of diver source of interest and separate the noises from different sources. The frequency content of the diver noise is also estimated adaptively according to azimuth distribution with no prior knowledge to enhance the separation ability from noises of similar azimuth and used to improve the correct rate of the classification. The performance of the proposed algorithm is discussed using both diver noise recorded in tank and at-sea shipping noise recorded in ocean for varying signal to noise ratio.

8:45

5aSP4. Impulse suppression algorithm development of a compatible program for cochlear implant users. Juliana Saba (Bioengineering, Univ. of Texas at Dallas, 800 W Campbell Rd., Richardson, TX 75080, jns109020@utdallas.edu), Jaewook Lee, Hussnain Ali (School of Behavioral and Brain Sci. (Speech & Hearing), Univ. of Texas at Dallas, Richardson, TX), Son Ta, Tuan Nguyen (Bioengineering, Univ. of Texas at Dallas, Richardson, TX), and John H. Hansen (School of Behavorial and Brain Sci. (Speech & Hearing), Univ. of Texas at Dallas, Richardson, TX)

Cochlear implant patients are more likely to abstain from impulsive-like environments due to limitations of current sound processing strategies. Conventional sound processing algorithms in conjunction with automatic gain controls tend to suppress impulsive sounds achieved unfortunately at the cost of reducing speech intelligibility. In this work, we propose a signal processing algorithm to provide protection to cochlear implant users in impulsive/noisy environments. A 22-band Advanced Combination Encoder (ACE) sound processing strategy was used as the base algorithm for speech enhancement. We propose an adaptable mathematical relationship to define impulsive like sound conditions and use this relationship to reduce the intensity of electrical pulses without altering the frequency ranges associated with speech to maintain speech quality/intelligibility. By varying the floor level of the environment, the program reduced/increased suppression intensity to achieve an effective configuration for practical listening environments. The algorithm was implemented in MATLAB and evaluated in benchtop (offline) testing with one CI user. Qualitative and quantitative analysis were performed using a paired preference test, a quality test, and an intelligibility test. Results showed increased quality of sound by +18%, and an average quality rating of processed speech 6/10 compared against 2.67/10 of speech in an unprocessed impulsive environment.

9:00

5aSP5. Bearing estimation using a single moving acoustic vector sensor. Edmund Sullivan (EJS_Consultants, 46 Lawton Brook Ln., Portsmouth, RI 02871, bewegungslos@fastmail.fm)

It is well known that a single moving pressure sensor can be used to estimate the bearing to a pure tone source if the source frequency is known a priori. It can also be shown, from the analysis of the relevant Grammian, that even if the source frequency and bearing are jointly estimated, say with a Kalman filter, it cannot be done, i.e., at least two sensors are necessary. However, if a PV sensor is used, additional measurements are provided, thereby allowing the estimation to be done. Although additional measurements are provided, there is still only a single physical sensor. Thus, even for small moving undersea vehicle, with its small amount of real estate, bearing measurements can be made. An example using simulated data is given.

9:15

5aSP6. Estimating blood pressures based on directly measured heart sounds. Lingguang Chen, Sean F. Wu (Dept. of Mech. Eng., Wayne State Univ., 5050 Anthony Wayne Dr., Detroit, MI 48202, lingguang.chen@wayne.edu), Yong Xu (Dept. of Electric and Comput. Sci. Eng., Wayne State Univ., Detroit, MI), William D. Lyman, and Gaurav Kapur (Dept. of Pediatrics, Wayne State Univ., Detroit, MI)

The current standard technique for blood pressure determination is to use cuff/stethoscope, which is not suited for infants or children. Even for adults this approach yields 60% accuracy with respect to intra-arterial blood pressure measurements. Moreover, it does not allow for a continuous monitoring of blood pressures over the course of 24 h and even days. In this paper, a new methodology is developed that will enable one to estimate systolic and diastolic blood pressures based on the heart sounds measured directly from a human body. To this end, we must extract and separate the aortic, pulmonic, mitral, and tricuspid sounds that are involved in the first and second hear sounds, respectively, from the directly measured signals. Since many parameters such as the locations from which heart sounds are originated, and speed at which heart sounds travel inside a human body are unknown a priori, it is impossible to develop an analytic model to determine the blood pressure. Hence, an empirical model is developed to estimate blood pressures based on the data measured by two sensors. Preliminary clinical tests show that the methodology can separate heart sounds and estimated systolic and diastolic blood pressures correlate well with the benchmark data.

9:30

5aSP7. Marine mammal range and depth estimation using non-separated multipath propagation. Amelie Barazzutti (ASC/ENV, DGA Eng. and Integration, 60, bd du general Martial Valin, CS 21623, Paris Cedex 15 75509, France, amelie.barazzutti@gmail.com), Cédric GERVAISE (Chorus Chair, Fondation Grenoble-INP, Grenoble, France), Kerri D. SEGER, Aaron THODE (Scripps Inst. of Oceanogr., UCSD, La Jolla, CA, U.S. Minor Outlying Islands), Jorge Urbán R., Pamela Martínez-Loustalot, M. Esther Jiménez-López, Diana López-Arzate (Universidad Autónoma de Baja California Sur, La Paz, Mexico), and Jérôme I. MARS (Gipsa-lab, Univ. Grenoble Alpes, Grenoble, France)

In passive acoustic monitoring, source localization using multipath propagation can be challenging whenever the source-receiver configuration leads to unresolved paths. Here, we propose a single-hydrophone method to estimate the range of a source based on two steps. First, we define a time deformation operator (e.g., Bonnel *et al.* 2014) to estimate the signal's impulse response (IR). Each group of paths, when warped at its time of arrival, is transformed into a tone that corresponds to a peak in the IR, which can be isolated in the warped time-frequency domain, permitting a significant improvement of the time resolution of the multipath, or groups of multipath. Next, relative arrival times are used to estimate the source range and depth. When only groups of paths are resolved, the identity of the paths constituting the group remains unknown, preventing the use of classical localization methods. To overcome this critical limitation, we apply an approach based on the characterization of groups of paths to estimate source location. We evaluate the method's performance on synthesized data, and examine its potential use on humpback whale low-frequency modulated calls using data collected off Cabo San Lucas in 2013.

9:45–10:00 Break

10:00

5aSP8. Joint beamforming and spectral enhancement for robust automatic speech recognition in reverberant environments. Fanuel Melak Asmare (Elec. Eng., Jacobs Univ. Bremen, Am Wall, 80, Bremen 28195, Germany, melakeegzi@gmail.com), Feifei Xiong (Fraunhofer IDMT, Oldenburg, Germany), Mathias Bode (Elec. Eng., Jacobs Univ. Bremen, Bremen, Germany), Bernard Mayer (Oldenburg Univ., Oldenburg, Germany), and Stefan Goetze (Fraunhofer IDMT, Oldenburg, Germany)

This work evaluates multi-microphone beamforming techniques and single-microphone spectral enhancement strategies to alleviate the reverberation effect for robust automatic speech recognition (ASR) systems in different reverberant environments characterized by different reverberation times T_{60}

and direct-to- reverberation ratios (DRRs). The systems under test consist of minimum variance distortionless response (MVDR) beamformers in combination with minimum mean square error (MMSE) estimators. For the later, reliable late reverberation spectral variance (LRSV) estimation employing a generalized model of the room impulse response (RIR) is crucial. Based on the generalized RIR model which separates the direct path from the remaining RIR, two different frequency resolutions in the short time Fourier transform (STFT) domain are evaluated, referred to as short- and long-term, to effectively estimate the direct signal. Regarding to the fusion between the MVDR beamformer and the MMSE estimator, the LRSV estimator can operate either on the multi-channel observed speech signals or on the single-channel beamformer output. By this, in this contribution, four different combination system architectures are evaluated and analyzed with a focus on optimal ASR performance w.r.t. word error rate (WER).

10:15

5aSP9. Single hydrophone, passive ranging of tonal sources in shallow water using the waveguide invariant. Andrew Young and Andrew Harms (Elec. and Comput. Eng., Duke Univ., 2127 Campus Dr., Durham, NC 27708, ayoung.ece@duke.edu)

A statistical method is presented for estimating both the value of the waveguide invariant, β, and the range to a tonal source in shallow water environments. This method requires minimal *a priori* environmental knowledge. A maximum-likelihood estimate of β is obtained from spectral analysis of a received signal from a tonal source over a known track. The range to a separate tonal source transiting the same shallow-water area is then estimated through a similar maximum-likelihood method, with the additional requirement of a range rate estimate. A recently published technique for estimating range rate through analysis of interference patterns of cross-correlated received time series is compared to a grid-search technique, in the context of evaluating performance of the range estimator. Both simulated and experimental results are presented. Simulated data were obtained using Kraken normal mode code. Experimental results used data from the Swellex '96 (SW96) experiment. The negative impact of incorrect estimates of source range rate, varying bathymetry, and interfering sources are discussed.

10:30

5aSP10. A discourse on the effectiveness of digital filters at removing noise from audio. Nicole Kirch and Na Zhu (Eng. Technol., Austin Peay State Univ., 601 College St., Clarksville, TN 37044, nkirch@my.apsu.edu)

In real world applications, audio signals become degraded by irregular signals called noise. Cell phone, hearing aid, and audio recording industries rely on noise reduction in their signal processing to maintain clarity and intelligibility. This research project compares various digital filtering algorithms in their ability to remove noise without damaging the desired audio signal. The chosen filters operate either in the time or frequency domains: low pass, high pass, band pass, and windowing. Prerecorded and preprocessed samples of voice and music are saturated with computer generated noise or a physical noise source, and the original audio signals are then compared to the noisy and filtered signals. Effectiveness and unique characteristics of the filters are recorded and quantized. This data can be used in designing future acoustic devices.

10:45

5aSP11. Efficient modeling of room acoustic impulse response by segmentation into early and late components. Sahar Hashemgeloogerdi and Mark F. Bocko (Elec. and Comput. Eng., Univ. of Rochester, 405 Comput. Studies Bldg., Rochester, NY 14627, shashemg@UR.Rochester.edu)

A Room Impulse Response (RIR) has a complex time-frequency structure that can be divided into a number of discrete early reflections followed by a reverberant tail. Many acoustic signal enhancement applications, such as dereverberation and room equalization, require simple yet accurate models to represent an RIR. Parametric modeling of RIR typically approximate the overall response for a given position of a source and receiver inside a room, without considering the fundamental distinction between early reflection (modal region) and late reverberation (statistical region) of an RIR. In this paper, the RIR is estimated in two steps: first, the early reflection component is modeled utilizing fixed pole, parallel orthonormal basis functions to represent the modal behavior of the room. The late reverberation component of an RIR is more statistical in nature with exponentially decaying envelope, exhibiting different decay time constants for different frequencies. Wavelet decomposition using a multi-rate analysis filter bank is applied to model this part of the RIR. An iterative method is used to estimate the parameters of the model from a measured target RIR. The proposed hybrid method provides an accurate representation of an RIR while requiring a smaller number of parameters in comparison to existing methods.

11:00

5aSP12. Acoustic trajectory corrections to improve mapping of moving sources. Benoit Oudompheng, Lucille Lamotte (MicrodB, Ecully, France), and Barbara Nicolas (Lyon Univ., Créatis, CNRS, 7 Ave. Jean Capelle, Villeurbanne Cedex 69621, France, barbara.nicolas@creatis.insa-lyon.fr)

In the context of pass-by experiments, moving-source mapping is classically performed using the extension of conventional beamforming to moving sources: beamforming-MS. The main issue of moving-source mapping is the knowledge of the trajectory of the moving vehicle that contains the sources. This trajectory is mandatory to use the beamforming-MS. Indeed, trajectory errors induce localization artifacts in beamforming results, which can degrade beamforming-MS performance and bias physical interpretations of the localization results. This paper suggests an original method to acoustically correct trajectography mismatches, using a first beamforming-MS computation and spatial correlations, in order to improve moving-source mapping results. Experimental results of aerial pass-by experiments demonstrate the improvements to the mapping results using this trajectory correction.

11:15

5aSP13. Intensity filtering of acoustic sensor signals for multi-target tracking. Sihan Xiong (Penn State Univ., State College, PA), Thyagaraju Damarla (U.S. Army Res. Lab., 2800 Powder Mill Rd., Adelphi, MD, thyagaraju.damarla.civ@mail.mil), and Asok Ray (Penn State Univ., State College, PA)

This paper addresses tracking of closely spaced moving vehicles by using passive acoustic signals. A multiple signal classification algorithm has been applied to estimate the bearing of targets from acoustic sensor signals. Field data from several sensor arrays have been used to reduce estimation variance in intensity filtering, which is performed on the bearing data with a sequential Monte Carlo method. Performance of the algorithm in terms of estimation of number of targets and their tracks will be presented using actual field data.

11:30

5aSP14. Non-inertial low frequency vector sensor with torsional suspension. Dimitri Donskoy (Stevens Inst. of Technol., 711 Hudson St., Hoboken, NJ 07030, ddonskoy@stevens.edu)

Conventional low frequency (from 10 Hz) velocity vector sensors utilize inertial approach using neutrally buoyant body with inertial sensor (accelerometer or moving coil geophone) inside. In order to achieve high sensitivity at the low frequencies these sensors must have relatively high inertia (mass). High mass leads to increased size (such as soccer ball) of the sensor in order to satisfy the condition of neutral buoyancy. The developed alternative non-inertial sensor utilizes torsional frictionless and extremely compliant suspension enabling very light and small, yet sensitive with low resonance sensor.

Session 5aUW

Underwater Acoustics: Underwater Noise

Stan E. Dosso, Chair
School of Earth & Ocean Sci., Univ. of Victoria, PO Box 1700, Victoria, BC V8W 3P6, Canada

Chair's Introduction—8:30

Contributed Papers

8:35

5aUW1. Using ambient noise correlations for relocating drifting sensor networks. Brendan Nichols, James Martin (Mech. Eng., Georgia Inst. of Technol., 801 Ferst Dr. NW, Atlanta, GA 30309, bnichols8@gatech.edu), Chris Verlinden (Sci., U.S. Coast Guard Acad., New London, CT), and Karim G. Sabra (Mech. Eng., Georgia Inst. of Technol., Atlanta, GA)

A drifting sensor network, such as hydrophones mounted on autonomous underwater vehicles or drifting buoys, can be used as a random volumetric array for locating underwater acoustic sources. However, navigational uncertainties of mobile platforms underwater limit the positioning accuracy of their acoustic sensors. Precise positioning is required to perform coherent processing for sound source localization using this random volumetric array. It has been shown that ambient ocean noise correlations between fixed receivers can provide additional inter-element distance information to perform array element self-localization [Sabra *et al.*, IEEE J. Ocean Eng., **30** (2005)]. An extension of this approach for an array of drifting sensors will be proposed by optimizing the required averaging duration to account for sensor drift motion. Performance of this proposed approach will be demonstrated using at-sea data collected in the Long Island Sound.

8:50

5aUW2. Soundscape analysis in the Southern Ocean using elephant seals as acoustic glider of opportunity. Dorian Cazau, Julien Bonnel (Lab-STICC (UMR CNRS 6285), ENSTA Bretagne,Université Européenne de Bretagne, 2 rue François Verny, 29806 Brest Cedex 09, France, 2, rue François Verny, Brest 29200, France, julien.bonnel@ensta-bretagne.fr), Yves Le Bras, and Christophe Guinet (CEBC, UMR 7273 ULR-CNRS, Villiers en Bois, France)

The underwater ambient sound field contains quantifiable information about the physical and biological marine environment. Since 2011, we have been annually collecting underwater data over the migratory routes of bio-logged Southern Elephant Seal (SES). As done with classical underwater gliders, we extract from these data very high resolution (approximately 30 min/400 m) ocean ambient noise measurements. In this conference, we present an overall picture of the low-to-medium frequency (10–6000 Hz) ambient noise distribution and its variability in time and space at a regional scale within the Indian Ocean. We detail our methodology to extract robustly the measurements usually performed on ocean ambient noise, such as sound pressure level over different frequency bands and their statistical percentiles. Also, we present our first attempts of exploiting acoustic recordings from bio-logged SES to infer surface wind speed. Wind maps from the ASCAT satellite (IFREMER, France) were used to study correlation relations between surface wind speed and acoustic content (e.g., the ratio of sound pressure levels at 1 and 6 kHz). In complement, we test SVM and Neural Network methods to estimate the presence of different classes of winds (e.g., below and above 10 m/s) from underwater ocean noise.

9:05

5aUW3. Identifying frequency and temporal characteristics of ambient noise using correlation matrices. Stephen Nichols and David L. Bradley (Acoust., The Penn State Univ., 201 Appl. Sci. Bldg., State College, PA 16802, smn5198@psu.edu)

Long-term ambient noise recordings contain an immense amount of information which quickly becomes cumbersome to analyze fully and efficiently. Correlation matrices provide a promising approach to identifying and isolating the active source mechanisms in ambient noise datasets. By comparing the noise levels in different frequency bands, frequency regions which tend to change together are identified, pointing toward either one common source or multiple related sources for the identified frequency bands. An attempt at simplifying the frequency and temporal information provided by series of correlation matrices is presented, utilizing data recorded by the Comprehensive Nuclear-Test Ban Treaty Organization's hydroacoustic monitoring system.

9:20

5aUW4. Time-domain cross-correlation using a normal mode formulation in shallow water. Haiqiang Niu and Peter Gerstoft (Marine Physical Lab., Scripps Inst. of Oceanogr., 9500 Gilman Dr., San Diego, CA 92093-0238, hniu@ucsd.edu)

Cross-correlation of ambient noise fields can yield temporal peaks, from which the travel time of acoustic paths between the sensors can be estimated. A good approximation of the time arrival structure of the actual Green's function is obtained if the noise field is isotropic. However, this condition is rarely satisfied in the ocean environment. For instance, ship noise is a common discrete source. Typically, the existence of discrete sources can affect the ambient noise cross-correlation. This paper presents a time-domain normal mode formulation of cross-correlation function for a single point source in a shallow water environment. The relation between cross-correlation function and time-domain Green's function is investigated for the point source by using time-domain normal mode formulations. The cross-correlation of the same modes yields two-point Green's function provided that the point source is in the end-fire direction, while the cross-correlation of the modes with different indexes represents a range-dependent component.

9:35

5aUW5. Passive characterization of underwater sound channel using ray-based blind deconvolution algorithm. Sung-Hoon Byun (KRISO, 32 1312 Beon-gil, Yuseong-daero, Yuseong-gu, Daejeon 34103, South Korea, byunsh@kriso.re.kr) and Karim G. Sabra (Mech. Eng., Georgia Inst. of Technol., Atlanta, GA)

Ray-based blind deconvolution (RBD) algorithm is a blind channel estimation method, which can estimate both the channel impulse response and the original signal broadcast from an unknown source. It is based on high-

frequency ray assumption and requires only the elementary information of array geometry and sound speed at the array position. Its capability of estimating the channel in a passive manner can be used to infer the sound channel characteristics from a source of opportunity such as passing surface ship whose positioning information is available from geo-referencing systems like the Automatic Identification System. Recently, we applied the RBD algorithm to the shipping noise data recorded on vertical line arrays at multiple locations in shallow water and confirmed its applicability for channel estimation and performing environmental inversion. This presentation shows the channel impulse responses estimated from a variety of source-receiver geometry and discusses the characteristics of sound propagation as well as the environmental property such as bottom loss which can be observed using the RBD algorithm.

9:50–10:05 Break

10:05

5aUW6. Detecting ice noise in Arctic ambient noise recorded on a drifting vertical line array. Emma Reeves, Peter Gerstoft, Peter Worcester, and Matthew Dzieciuch (Scripps Inst. of Oceanogr., Univ. of California San Diego, 9500 Gilman Dr., La Jolla, CA 92093, ecreeves@ucsd.edu)

From April to September 2013, a 600-m Distributed Vertical Line Array (DVLA) with 22 hydrophones drifted from the North Pole toward the Fram Strait. The DVLA recorded low-frequency ambient noise (1953.125 Hz sampling) for 108 minutes six days per week. Spectral analysis of the ambient noise data indicates the presence of acoustic transients, which include airgun and drilling sources as well as ice shear-induced broadband events and FM tones. Similar ice shear signatures in the Beaufort Sea were highly correlated with environmental variables leading to rifting [Kinda *et al.*, J. Acoust. Soc. Am. **138**, 2034–2045 (2015)], demonstrating the potential of acoustics to monitor rapid sea ice processes and pack ice breakup. Motivated by this study, a wavelet-based audio fingerprinting method is applied for identifying ice noise in the drifting DVLA data. The algorithm blindly groups spectral signatures, allowing the categorization of previously unrecognized signatures. The size of each category at points along the drift path is presented as a local, seasonally dependent estimate of the presence of ice noise.

10:20

5aUW7. Very high frequency underwater noise from breaking waves. Grant B. Deane and M. D. Stokes (Marine Physical Lab., Scripps Inst. of Oceanogr., Code 0206, UCSD, La Jolla, CA 92093-0238, gdeane@ucsd.edu)

In the absence of anthropogenic and biological sources and under wind-driven seas, underwater sound generated by breaking waves is the dominant source of high frequency ambient noise in the ocean. At frequencies above O (1000 Hz), this noise is generated by pulses of sound radiated by air bubbles entrained in actively breaking whitecaps. A history of measurements extending back more than 50 years has shown that the sound radiated by breaking waves is well-correlated with wind speed and extends in frequency to at least 20 kHz. Here, we present measurements of high frequency noise radiated by breaking laboratory waves. Focused wave packets in the Glass Wave Channel at the Hydraulics Laboratory at SIO generated breaking waves ranging in type from spilling to plunging, which were representative of small to medium scale waves on seas driven by 8–12 m/s winds. Measurements made with high frequency hydrophones (ITC 1089D) show that the noise spectrum from these waves extends beyond 400 kHz, challenging the assumption that noise above 100 kHz is dominated by thermal effects. Moreover, the slope of the noise spectrum is roughly −17 dB/decade, consistent with oceanic noise spectra at lower frequencies.

10:35

5aUW8. Sound production in two species of eelgrass. Jeff Rice (Univ. of Washington, 18529 29th Ave. NE, Lake Forest Park, WA 98155, jrice1000@mac.com)

There have been numerous anecdotal reports that a species of the aquatic plant known as eelgrass (*Zostera marina*) makes an audible bubbling sound that can be heard on warm, sunny days in estuarine waters at low tide. This champagne-like fizzing sound results from the expiration of oxygen by the plant during photosynthesis and can be heard as airborne sound by listening closely to the surface of the water. The author recorded this sound in July of 2014 in the shallow mudflats of Puget Sound in Washington. Recordings were made of both airborne and underwater sounds. On the same day in the same general area, the author also recorded similar bubbling from a different, non-native species of eelgrass (*Zostera japonica*). The sounds of the two species were compared, revealing clear and distinct sound signatures for each. These recordings suggest that sound production in the two species of eelgrass is strongly present and may provide opportunities for further research.

10:50

5aUW9. An adaptive signal tools design inspired by marine mammals calls. Siyuan Cang and Xueli Sheng (College of Underwater Acoust. Eng., Harbin Eng. Univ., No. 145 Nantong Str., Nangang Dist., Harbin 150001, China, cangsiyuan@hrbeu.edu.cn)

As the main wisdom animal in the ocean, the marine mammals like dolphins and whales has got excellent capabilities for complex ocean environments through millions of years of evolution. They physically look simple, but are able to do lots of fantastic jobs, such as echolocation, preying, and even communication among populations of marine mammals. This paper proposes a method of designing an active sonar waveform inspired by the marine mammals calls. Two math models of marine mammal sound signal are adopted, one AR model, the other piecewise linear chirp model. The time-frequency structures, ambiguity function and Q function will be analyzed. As the simulation results shown, it has a good range and velocity resolution under environment with reverberation. In the end, a concept named Marine Mammal Signal Tools (MMST) is presented. The signal generating from MMST can be adaptive to varied ocean environments. In the sense, it can be extensively used and plays more significant roles in the modern sonar.