Session 4aAA

Architectural Acoustics, Engineering Acoustics, Signal Processing in Acoustics, and Noise: Microphone Array Applications in Room Acoustics

Gary W. Elko, Cochair
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Chair's Introduction—7:45

Invited Papers

7:50

4aAA1. Practical open-concentric spherical microphone array design. Mark R. Thomas (Sound Technol. Res., Dolby Labs., 1275 Market St., San Francisco, CA 94103, mark.r.thomas@ieee.org)

The problem of higher order sound field capture is considered. First-order microphone arrays, such as tetrahedral A-format designs, are commonplace, while interest remains in the increased spatial resolution delivered by higher order arrays. Such arrays typically consist of pressure microphones mounted on a solid spherical baffle, with which higher-order spatial components are estimated algorithmically. This produces a design trade-off, with small arrays being preferred for spatial aliasing performance and large arrays being preferred for reduced amplification of microphone capsule noise at low frequencies. A practical open sphere design is proposed that contains microphones mounted at multiple radii to fulfill both criteria. Coupled with the use of cardioid microphones, such an array captures higher order soundfields over a wider audio bandwidth than baffled spherical designs of fixed radius.

8:10

4aAA2. Self-calibration of dual, closed surface microphone arrays. Earl G. Williams (Acoust. Div., Code 7106, Naval Res. Lab., 4555 Overlook Ave., Washington, DC 20375, earl.williams@nrl.navy.mil) and William A. Kuperman (Scripps Inst. of Oceanogr., Univ. of California San Diego, La Jolla, CA)

With the recent availability of very inexpensive, miniature digital MEMS microphones, the construction of large count microphone arrays is financially within the reach of the basic researcher. Using various mathematical techniques such as the Helmholtz-Kirchhoff integral and the equivalent source method (ESM), dual surface microphone arrays provide the ability to separate incident and scattered fields instantaneously in time. When these arrays form a closed surface, this separation is extremely accurate. However, one of the concerns with any microphone array is the calibration of the individual elements, especially near the resonance frequency of the diaphragm. In this paper, a self-calibration technique is proposed, based on ESM, that uses external loudspeakers in a simple laboratory facility not required to be anechoic. The approach is tested numerically on a dual surface, 256 element pair spherical microphone array, and uses an algorithm for the minimization of a cost function based on the extinction of the scattered field when the unknown individual calibration coefficients of the array elements are varied. Use of a Burton-Miller approach to model the ESM sources of the scattered field improves dramatically the accuracy at specific frequencies. The problematic interior Dirichlet eigenfrequencies are addressed. [Work supported by the Office of Naval Research.]

8:30

4aAA3. Characterizing acoustic environments using spherical loudspeaker and microphone arrays. Hannes Gamper, Keith Godin, Nikunj Raghuvanshi, and Ivan J. Tashev (Res., Microsoft, One Microsoft Way, Redmond, WA 98052, hannes.gamper@microsoft.com)

Room acoustic parameters, including the reverberation time (T60) and the direct-to-reverberant ratio (DRR), quantitatively describe the behaviour of an acoustic environment. These parameters are typically derived from an acoustic impulse response (AIR) obtained using an omnidirectional sound source and an omnidirectional receiver. However, the acoustic interaction between a source with a complex radiation pattern, for example, a human talker, and a receiver with a complex directivity pattern, for example, a human listener, cannot be accurately described by an omnidirectional AIR or the acoustic parameters derived from it. Here, we propose characterizing acoustic environments while taking the source and receiver directionality into account. A spherical loudspeaker array allows modelling a source with an arbitrary radiation pattern. Analogously, a spherical microphone can be used to describe a receiver with an arbitrary directivity pattern. By combining spherical loudspeaker and microphone arrays, a multiple-input multiple-output (MIMO) AIR of an acoustic environment can be measured to simulate sources and receivers with arbitrary directivities and estimate acoustic parameters accordingly.

8:50

4aAA4. The room impulse response in time, frequency, and space: Mapping spatial energy using spherical array beamforming techniques. Matthew T. Neal and Michelle C. Vigeant (Graduate Program in Acoust., Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, mtn5048@psu.edu)

The auditory perception of rooms is a multi-dimensional problem. Our hearing system interprets time, frequency, and spatial information from arriving room reflections, but traditionally, only the time and frequency domains are considered in room acoustic metric and objective sound field analyses. This work aims to develop spatial visualizations of the energy in a room impulse response (RIR). With a spherical microphone array, a room's energy can be mapped in full three-dimensions. First, beamforming techniques are used to generate a set of directional RIRs from the spherical microphone array measurement. This set of directional RIRs is analogous to using a microphone with a directional beam pattern response, oriented individually at all points around a sphere. Then, these directional or beam RIRs are time windowed and band-pass filtered to create spatial energy maps of the room. Comparisons between a plane-wave beam pattern and a Dolph-Chebyshev beam pattern will be demonstrated in the context of RIR beamforming. As well, different strategies for normalizing peak energy amplitudes to either the direct sound or a spherical spreading condition will be compared. With these considerations, final results of these spatial energy visualizations and directional RIR animations will be demonstrated. [Work supported by NSF Award 1302741.]

9:10

4aAA5. Model-based direction of arrival estimations for sound sources using a spherical microphone array. Christopher R. Landschoot (Architectural Acoustics, Rensselaer Polytechnic Inst., 54 West Main St., Clifton Springs, NY 14432, crlandschoot@gmail.com), Jonathan Mathews, Jonas Braasch, and Ning Xiang (Architectural Acoustics, Rensselaer Polytechnic Inst., Troy, NY)

In many room acoustics and noise control applications, it is often challenging to identify the directions of arrivals (DoAs) of incoming sound sources. This work seeks to solve this problem reliably by beamforming, or spatially filtering, incoming sound data with a spherical microphone array via a probabilistic method. When estimating the DoA, the signal under consideration may contain one or multiple concurrent sound sources originating from different directions. This leads to a two-tiered challenge of first identifying the correct number of sources, followed by determining the directional information of each source. To this end, a probabilistic method of model-based Bayesian analysis is leveraged. This entails generating analytic models of the experimental data, individually defined by a specific number of sound sources and their locations in physical space, and evaluating each model to fit the measured data. Through this process, the number of sources is first estimated, and then the DoA information of those sources is extracted from the model being the most concise to fit the experimental data. This paper will present the analytic models, the Bayesian formulation, and preliminary results to demonstrate the potential usefulness of this model-based Bayesian analysis for complex noise environments with potentially multiple concurrent sources.

9:30

4aAA6. Iterative echo labeling technique for room geometry estimation. Soo Yeon Park and Jung-Woo Choi (Elec. Eng., Korea Adv. Inst. of Sci. and Technol., KAIST, 291 Daehak-ro, Yuseong-gu, Daejeon 34141, South Korea, parksean210@kaist.ac.kr)

Room geometry estimation using acoustic room impulse responses (RIRs) has been tackled in various ways. Most of them detect wall locations using the time of arrival (TOA) of echoes between multiple microphones and loudspeakers. To combine TOA information from multiple room impulse responses, echoes from the same walls should be collected and processed together. When multiple microphones are widely distributed in a room, however, it is hard to identify walls responsible for the generation of each echo. In this work, an iterative update algorithm is proposed for resolving the echo labeling problem with low computational complexity. Unlike conventional algorithms utilizing the rank property of Euclidean distance matrix, the proposed algorithm utilizes the property of convex hull formed by ellipses representing individual TOAs. The location of the reflectors are sequentially identified from common tangent lines constituting the convex hull, and the search result is iteratively updated by inspecting later echoes in time. In addition, a termination criterion is also developed in order to enable the geometry estimation without the prior knowledge of the number of the reflectors.

9:50

4aAA7. Nonsingular EB-ESPRIT for the localization of early reflections in a room. Byeongho Jo and Jung-Woo Choi (Elec. Eng., Korea Adv. Inst. of Sci. and Technol., #2103 N24 LG Innovation Hall KAIST, 291 Daehak-ro Yuseong-gu, Daejeon 34141, South Korea, byeongho@kaist.ac.kr)

The eigenbeam-ESPRIT (EB-ESPRIT) can directly identify the direction of arrivals (DoAs) of sound sources through the parametric estimation process. This parametric estimation is beneficial for identifying directions of early echoes in an enclosed space, because it does not need an extra peak searching step for finding dominant peak locations from a beamforming power map. Nevertheless, the use of EB-ESPRIT for echo localization has been hindered due to its ill-conditioning problem for sources positioned near the equator of the spherical coordinates. Room reflections in a reverberant room can impinge from any direction, and hence, the ill-conditioning is highly likely to occur. In this work, a nonsingular EB-ESPRIT technique is utilized for the echo localization without the ill-conditioning problem. The technique is based on new recurrence relations of spherical harmonics that express DoA parameters in form of sinusoidal functions. Therefore, the ill-conditioning due to the use of arctangent function can be avoided. The performance is demonstrated using the room impulse responses simulated and measured for a rigid spherical microphone array. Frequency smoothing is applied to localize echoes that are highly coherent to each other, and results demonstrate that accurate localization of echoes is possible irrespective of their incidence angles.

10:10–10:25 Break

10:25

4aAA8. Analysis of sound field isotropy based on directional energy decay curves. Marco Berzborn and Michael Vorlaender (Inst. of Tech. Acoust., RWTH Aachen Univ., Kopernikusstr. 5, Aachen 52074, Germany, marco.berzborn@akustik.rwth-aachen.de)

The analysis of the diffuseness of sound fields is of great interest in room acoustic applications ranging from the analysis of concert venues to reverberation room design and calibration. However, a standardized robust diffuseness estimation method is currently lacking. The fundamental definition of the diffuse sound field is that it is isotropic—requiring the sound field to be composed of infinitely many sound waves from uncorrelated sources with directions of arrival uniformly distributed over a sphere. Due to their symmetry, spherical microphone arrays are especially favorable for the analysis of the isotropy requirement. In this work, we propose the directional energy decay curve which we calculate from a directional room impulse response captured with a spherical microphone array. Further, we show how the analysis of the spatial variations of the directional energy decay curve enables examining the isotropy of sound fields in rooms. Finally, we present a simulation study of multiple room configurations with varying degrees of sound field diffuseness.

10:45

4aAA9. Beamforming loudspeaker array in a multi-layered configuration. Jeong-Guon Ih (Mech. Eng., KAIST, 373-1 Guseong-Dong, Yuseong-Gu, Daejeon 305-701, South Korea, J.G.Ih@kaist.ac.kr) and Wan-Ho Cho (KRISS, Daejeon, South Korea)

A planar, broadside acoustic array with uniform loudspeaker spacing is often adopted for generating a highly directional acoustic beam, but the performance in radiated power and spatial bandwidth should always compromise with the transverse size limit. To achieve a high-power, directional sound in a compact size, a multi-layered loudspeaker array configured in both broadside and end-fire types is devised. This design concept is tested with an array of 3 layers with 0.2 m spacing. Each layer contains 7 loudspeakers in a centrally symmetric arrangement on a plane with the longest dimension of 0.6 m. In each layer, the separation distance between loudspeakers is uniformly set as 0.2 m. The optimal beamforming method is employed for calculating the control filter. The compensation for the performance degradation due to self-scattering effect is also included in the control filter, for which the additional phase delay due to scattering is estimated from the simulated response of the array system. The maximum sound level at 1 m distance from the array center is achieved as 145 dB at 1.5 kHz, and, at 30 m position, the level decrease of 26–30 dB with the directivity index of 15 dB is observed for 1–5 kHz.

11:05

4aAA10. Modal reconstruction of the sound field in a room at low frequencies. Efren Fernandez-Grande (Acoust. Technol., DTU - Tech. Univ. of Denmark, Ørsteds Plads, B. 352, DTU, Kgs. Lyngby DK-2800, Denmark, efg@elektro.dtu.dk)

Capturing the spatio-temporal (or spatio-spectral) properties of the sound field in a room is valuable for its characterization. To measure the full spatio-spectral properties within the volume of the room, i.e., the frequency response functions at all points, a very large number of measurements is required. Even at low frequencies, sampling the three-dimensional space enclosed by a room is challenging. It is therefore of interest to employ sound field reconstruction methods to predict (interpolate/extrapolate) the sound field over the volume of the room, based on a limited number of measurements. The aim of this study is to develop a sound field reconstruction method that exploits the modal structure of the sound field in a room, to reconstruct the sound field at low frequencies over the entire volume of the space. The proposed methodology makes use of the fact that the spatial structure of the acoustic field in a room is inherently block-sparse and can be extracted from spatially distributed measurements. The modeshapes are represented as the superposition of a small number of plane waves, and the modal basis is used to reconstruct the sound field.

11:25

4aAA11. Postfiltering based on the coherent-to-diffuse power ratio. Michael Günther, Andreas Brendel, and Walter Kellermann (Chair of Multimedia Communications and Signal Processing, Friedrich-Alexander-Universität Erlangen-Nürnberg, Cauerstr. 7, Erlangen 91058, Germany, michael.guenther@fau.de)

In microphone array processing, the availability of multiple sensors observing an acoustic scene allows the extraction of additional spatial sound field properties. The coherent-to-diffuse power ratio (CDR) is a signal-dependent quantity which relates the coherent, typically desired, component and the diffuse, typically undesired, component of a recorded signal and can be employed in different signal enhancement tasks such as dereverberation and noise suppression. Given suitable models for the coherent and diffuse signal components, the complex spatial coherence function of the observed sound field facilitates an efficient estimation of the CDR. Several unbiased CDR estimators can be derived which differ in the sensitivity to estimation errors and the amount of incorporated prior information, e.g., the direction of arrival (DOA) of a desired point-like source. We review popular DOA-dependent and DOA-independent CDR estimators and investigate their efficacy in single-channel postfiltering when multiple point-like acoustic sources, i.e., one target source and at least one coherent interferer, are present.

4a THU. AM

Contributed Papers

11:45

4aAA12. Measurement of room spatial correlation functions using spherical microphone arrays. Gary W. Elko (mh Acoust., 25A Summit Ave., Summit, NJ 07901, gwe@mhacoustics.com) and Jens Meyer (mh Acoust., Fairfax, Vermont)

The study of acoustic spatial correlation functions has been ongoing since early analytical development and experiments were done by Farran and Hills in 1952. The measurement of sound diffuseness using spatial correlation functions has been discussed previously by the authors in 2009. It has been conjectured that audio rendering of the spatial impression of a soundfield is related to the degree of correlation between the microphones used to record the soundfield. The correlation between the recorded channels is related to the diffuseness of the sound field. Many different models of diffuse fields have been described and are based on a definition that involves incoherent sound coming uniformly from all directions. Schroeder surmised that as the modal density exceeds three overlapping modes in one modal bandwidth that the random uniform model can be utilized in practice.

In order to obtain measured room spatial correlation functions one has to ensemble average either multiple source or receiver positions. With eigenbeamforming spherical microphone arrays these functions can be ensemble averaged by utilizing multiple steered beampatterns with at least two spaced arrays. We will show some theoretical spatial correlation functions for different beampatterns and orientations and compare these to measurements made in a real room.

12:00

4aAA13. Transient solution for the directional response at the focus of a paraboloidal reflector. John M. Cormack and Mark F. Hamilton (Appl. Res. Labs., Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, jcormack@utexas.edu)

A transient solution is presented for the directional response at the focus of a paraboloidal reflector due to an incident plane wave. The validity of the solution is associated with the Kirchhoff approximation used to determine the boundary condition on the surface of the reflector, which requires the wavelengths to be short compared with the minimum radius of curvature of the reflector. The solution at the focus due to an incident plane wave is obtained by applying the principle of reciprocity to the solution in the far field due to a point source at the focus. Both solutions are expressed as the convolution of the unit step response with the time derivative of the waveform incident on the reflector. For a point source far removed from a shallow paraboloidal reflector, it is shown that the pressure in the far field reduces to the expression obtained by Morse (*Vibration and Sound*) for transient radiation from a baffled circular piston. Results are presented illustrating the angular dependence of the reflected pressure waveforms at the focus for incident plane waves that include a rectangular pulse, an N wave, and a tone burst. [J.M.C. was supported by the ARL:UT McKinney Fellowship in Acoustics.]

THURSDAY MORNING, 8 NOVEMBER 2018 — SHAUGHNESSY (FE), 8:00 A.M. TO 12:00 NOON

Session 4aAB

Animal Bioacoustics: Soundscapes, Noise, and Methods in Animal Bioacoustics

Carrie C. Wall, Chair
Cooperative Institute for Research in Environmental Sciences, University of Colorado at Boulder, 216 UCB, Boulder, CO 80309

Contributed Papers

8:00

4aAB1. An experiment of sound effects on shrimps for ecoacoustics' research. Zhengliang Cao (Shanghai Ocean Univ., 999 Hucheng Huan Rd., Shanghai 201306, China, caozhengliang@yahoo.com)

It is important to understand relationships of underwater soundscapes, marine ecology, and acoustic behaviors for refine research questions on ecoacoustics. Benthic crustaceans live in a local range and are normally used to an indication of environmental impacts, especially for the ecosystem in ocean. To get much information of interaction impacts on benthic crustaceans, this paper introduces an experiment of sound effects on shrimps in an aquatic pond. In the experiment, there is only a kind of benthic crustaceans, Penaeus prawns. Environmental and artificial noises are recorded by a passive sound monitor. The preliminary results are obtained through preprocessing data acquisition, analyzing effective phenomena, and comparing sound characteristics. It will be valuable to consider much simple cases for main questions. So data of natural rainfall noise, aerator noise and artificial noise may help to know about the acoustic ecology of shrimps in an aquaculture environment. Since the water quality is very poor in real shrimp pond, it cannot accurately obtain the relationship between the sound of shrimp and its behavioral characteristics. Further research considerations are also be prospected in this paper. [The project is funded by The National Natural Science Foundation of China (41374147) and (14111103900).]

8:15

4aAB2. A new model for underwater noise research in larval fishes: Biomedical and ecological implications. Allison Coffin (Integrative Physiol. and Neurosci., Washington State Univ., 14204 NE Salmon Creek Ave., Vancouver, WA 98686, allison.coffin@wsu.edu), Jie Xu, Dmitry Gritsenko (Mech. and Industrial Eng., Univ. of Illinois at Chicago, Chicago, IL), Kristy Lawton (Integrative Physiol. and Neurosci., Washington State Univ., Vancouver, WA), Beija Villalpando (College of Arts and Sci., Washington State Univ., Vancouver, WA), Joseph A. Sisneros (Psych., Univ. of Washington, Seattle, WA), Ashwin Bhandiwad (Psych., Univ. of Washington, Bethesda, MD), and Phillip Uribe (Integrative Physiol. and Neurosci., Washington State Univ., Vancouver, WA)

In humans, excessive noise exposure from occupational or recreational sources causes permanent hearing loss. Similarly, exposure to underwater anthropogenic noise can cause hearing loss in aquatic organisms, including fish. While fish can recover from noise-induced hearing loss, underwater noise exposure can cause behavioral changes that reduce organismal fitness. In all vertebrates, acoustic trauma can cause damage to sensory hair cells.

To better study the effects of noise on hair cells, we have developed a noise exposure system that uses broadband sound to damage hair cells of the inner ear and lateral line of larval zebrafish. Acoustic over-exposure kills hair cells in an intensity- and time-dependent manner, with maximum hair cell damage observed 72 hours after noise exposure. This time course is consistent with mammalian studies, where hair cell death occurs days to weeks after noise exposure. Other features of acoustic trauma are also conserved between zebrafish and mammals, including activation of apoptotic signaling cascades and changes in hair cell-afferent synapses. These studies demonstrate that larval zebrafish are a tractable new model for studies of noise-induced hair cell death. However, our acoustic trauma system could also be used in other species, allowing for new studies of underwater noise in larval fishes.

8:30

4aAB3. Potential changes in the communication space of humpback whale social sounds in increasing wind and vessel-dominated noise. Rebecca Dunlop (School of Veterinary Sci., Univ. of Queensland, Cetacean Ecology and Acoust. Lab, Gatton, QLD 4343, Australia, r.dunlop@uq.edu.au)

An animal communication network involves complex acoustic interactions between multiple senders, receivers, and eavesdroppers. Any reduction in communication space, due to signal masking, may have detrimental effects on their ability to obtain social information. Humpback whales use social sounds (vocal and surface-generated percussive sounds) for within-, and between-, group communication. To generate masking models, and infer communication space, changes in signal-above-noise and frequency content of received humpback whale social sounds were modelled with the combined effect of increasing background noise (wind or vessel-dominated) and distance from the source (signalling whale). Results suggest that the signaler's communication space in increasing wind-dominated noise (shallow water) was maintained out to approximately 3 km by using a Lombard response. In high wind noise (over 105 dB re 1 μPa; 12–15 knots), the vocal communication space was significantly reduced, though the increased use of surface-generated sounds likely aided in maintaining this space. In vessel-dominated noise, communication space was significantly reduced in levels exceeding 110 dB re 1 μPa (vessel within approximately 2 km), with no evidence that the signaler switched to using surface-generated sounds. These models can be updated as more information on humpback auditory capabilities becomes available.

8:45

4aAB4. Assessing risk of underwater noise impact on marine mammals throughout a new methodology. Marie Mauran, Florent Le Courtois, Marie Cachera, Yann Stéphan (HOM, Shom, 13 Rue de Châtellier, Brest 29200, France), Jérôme Spitz (Observatoire PELAGIS - UMS 3462, La Rochelle, France), and G. Bazile Kinda (HOM, Shom, Brest, France, bazile.kinda@shom.fr)

Shipping noise has been identified as a threat on underwater ecosystems. In particular, several impacts have been documented on cetaceans, e.g., communication's masking and stress increasing. To assess the risk related to such anthropogenic noise requires both quantifying the noise levels and estimating the distribution of the cetaceans' population. However, current methods evaluating the risks related to anthropogenic pressures generally rely on strong expert priors, which may be difficult to define. This presentation aims at introducing a new framework for the comparison of anthropogenic pressure levels maps and cetaceans' distribution in order to infer the risk of impact and provide management solutions. The methodology, combining simple statistical analyses and a theoretical representation tool was applied to the Bay of Biscay using shipping noise model and fin whale observation from regular surveys for the years 2012 and 2016. Relationships between cetaceans' distribution and noise levels were investigated and linked to mammals theoretical responses to pressure. A trend analysis between 2012 and 2016 was also proposed to identify the noise hotspots. The results were interpreted in both terms of ecological meaning for fin whale and conservation measures for shipping noise, according to marine policies requirements.

9:00

4aAB5. Monitoring noise levels and delphinid presence through autonomous acoustic in Rio de Janeiro coastal area. Lis Bittencourt (Programa de Pós-Graduação em Oceanografia, Rio de Janeiro State Univ., São Francisco Xavier, 540, sala 4002, bloco E, Rio de Janeiro, RJ 20550013, Brazil, lis.bitt@gmail.com), André Sousa, Tatiana Bisi, José Lailson-Brito, and Alexandre d. Azevedo (MAQUA - Laboratório de Mamíferos Aquáticos e Bioindicadores, Rio de Janeiro State Univ., Rio de Janeiro, Brazil)

In order to investigate the soundscape in a Rio de Janeiro coast site, passive acoustic monitoring was conducted during two non-consecutive weeks in the summer of 2015/2016 by deploying one SM2M+ device. The equipment recorded at a 66% duty cycle with sample rate of 96 kHz and 36 dB gain. Third octave levels (TOLs) were calculated for all recordings through PAMGuide software. To search for delphinid presence, a band limited energy detector was employed using Raven 1.5 in a 512 Hann window, 50% overlap. TOLs varied across frequencies and day hours. Light hours were noisier than dark hours in 25 frequency bands (MW, $p<0.01$), with highest mean level being measured at 794 Hz at 06am (105.4 $\pm$ 5.4 dB re 1μPa), and lowest mean level measured at 39.8 kHz at 01am (80.3 $\pm$ 3.7 dB re 1μPa). A total of 281 delphinid sound emissions were detected in nine occasions, seven during week 1 and two during week 2. Seven detection events occurred during dark hours. Although more than one species of delphinid is known to occur in Rio de Janeiro coast, these results indicate a frequent nocturnal use of the area, which was only possible to observe through autonomous monitoring.

9:15

4aAB6. Do nearby seismic airguns reduce singing effort in humpback whales? Michael J. Noad and Rebecca Dunlop (School of Veterinary Sci., The Univ. of Queensland, Gatton, QLD 4343, Australia, mnoad@uq.edu.au)

The high-level percussive sounds generated for seismic sea floor exploration have the potential to disrupt normal behaviors of whales. This study assesses if there is any reduction in the singing behaviour of migrating humpback whales in response to nearby airguns. Singing whales were acoustically tracked as they migrated along the coastline of south-eastern Queensland. A 20 cubic inch airgun or 140 cubic inch array of airguns was towed through the study area for 1 hour with the airguns firing every 11 sec (active treatments) or without the airguns operating (controls). Singing activity across each day was measured by counting the number of singing whales within the 10km-radius study area every 10 min, from 0700 to 1700, including during experiments. Singing activity during active periods and controls were compared with each other as well as with pseudo-randomly selected 1 hour periods when experiments were not underway (baseline). Changes in singing effort were also recorded by noting the number of instances of a singer stopping or starting during these periods. Preliminary analyses show no significant differences in singing activity, nor changes in singing effort, between active, control, or baseline periods.

9:30

4aAB7. Estimating received levels for acoustically tracked whales from Navy mid-frequency active sonar. Cameron R. Martin, Stephen W. Martin (National Marine Mammal Foundation, 2240 Shelter Island Dr. Ste. 200, San Diego, CA 92106, cameron.martin@nmmpfoundation.org), and E. E. Henderson (Navy Marine Mammal Program, SSC Pacific, San Diego, CA)

Multi-channel passive acoustic data were collected from bottom-mounted hydrophones off the coast of Kauai, HI at the U.S. Navy Pacific Missile Range Facility's instrumented range. In February 2017, data were recorded during the Submarine Command Course (SCC) training event that included mid-frequency active sonar (MFAS). Data were post-processed for minke whale (*Balaenoptera acutorostrata*) vocalizations and MFAS with automated detection, classification, and localization algorithms. Semi-automated processes were used to perform spatio-temporal associations of whale localizations to create individual whale tracks and to associate localized

MFAS transmissions with positional data from multiple ships. For each whale track, the received level from each MFAS transmission was estimated using a parabolic equation propagation model that statistically represented the uncertainty of whale location and depth. Exposures were accumulated over the duration of each whale track to estimate the cumulative sound exposure level (cSEL; dB re: $1\mu Pa^2 s$). Twenty-three minke whales were exposed to MFAS during the SCC; the maximum exposure was 169.8 dB cSEL, and the minimum distance to a ship transmitting MFAS was 4.8 km. Currently, statistical analyses of track kinematics and call characteristics are being conducted to quantify any significant changes between animal track kinematics with and without MFAS training presence.

9:45–10:00 Break

10:00

4aAB8. Effects of vessels and noise on the subsurface behavior of endangered killer whales (*Orcinus orca*). Marla M. Holt, Brad Hanson, Candice Emmons (NOAA NMFS NWFSC, 2725 Montlake Blvd. East, Seattle, WA 98112, Marla.Holt@noaa.gov), Jennifer B. Tennessen (Lynker Technologies, University Park, PA), Deborah Giles (Univ. of Washington Friday Harbor Labs, Friday Harbor, WA), and Jeffery Hogan (Cascadia Res. Collective, Olympia, WA)

Prey availability and disturbance from vessels and noise are identified threats to endangered Southern Resident killer whales. Vessel noise can mask echolocation signals used for hunting and/or disrupt foraging with implications for energy acquisition in a likely prey-limited population. We utilized suction cup-attached digital acoustic recording tags (DTAGs), to measure received noise levels, understand killer whales' use of sound, and determine effects of vessels/noise on subsurface behavior. During the 28 tag deployments, we collected vessel data concurrently along with opportunistic predation observations to validate feeding. Broadband received levels (dB re $1\mu Pa$) were significantly different across years. Of the vessel factors considered, both vessel count and speed were significant explanatory variables of received levels. Vessels emitting echosounder signals were commonly received by the DTAGs and overlapped with the echolocation frequencies use to hunt fish. Additionally, different phases of foraging were differentiated from the acoustic record, including the detection of crunching sounds after fish kills. Together with movement data analysis, these results allow the identification of different whale activities, including prey capture dives, to test hypotheses of vessel/noise effects on behavior. This work, along with a comparative investigation involving Northern Resident DTAG data, inform killer whale conservation and management measures.

10:15

4aAB9. Localizing bioacoustic signals with long-baseline hydrophone arrays. Benjamin T. Hendricks, T. Aaron Gulliver (Elec. and Comput. Eng., Univ. of Victoria, 3800 Finnerty Rd., Eng. Office Wing, Rm. 448, Victoria, BC V8P 5C2, Canada, hendrick@uvic.ca), Janie L. Wray (North Coast Cetacean Society, Hartley Bay, BC, Canada), Eric M. Keen (Marine Ecology and Telemetry Res., Seabeck, WA), Hussein M. Alidina (Oceans, World Wildlife Fund - Canada, Victoria, BC, Canada), Chris R. Picard (Gitga'at Oceans and Lands Dept., Hartley Bay, BC, Canada), and Hermann Meuter (none, Hartley Bay, BC, Canada)

Real-time localization and tracking of vocalizing marine mammals is a powerful tool to a) mitigate the risk of ship strikes and disturbance of the animals as well as b) to understand the impact of anthropogenic noise on habitat use of cetaceans. A long baseline hydrophone array has been installed in Squally Channel, a culturally, ecologically, and economically important marine environment in northern British Columbia, Canada. The array consists of four synchronized bottom-mounted hydrophones that permanently record and radio-transmit data to a land-based laboratory. Automated analysis tools have been developed to detect and localize transient bioacoustic signals from three or more hydrophones in real-time. They comprise the correlation of hydrophone signals, the construction of a replica surface using acoustic propagation models, and signal localization from a spatial likelihood surface. An overview of the localization method is presented. We assess spatial localization precision and discuss random and systematic sources of error stemming from the signal type and quality, bathymetry, sound speed profile, as well as the spatial distribution of hydrophones and acoustic sources. Simulation results are compared to field measurements. Based on our findings, challenges and opportunities associated with localization using long baseline arrays in coastal environments are discussed.

10:30

4aAB10. Multi-channel cross-correlation used to estimate time delays. Eva-Marie Nosal (Ocean and Resources Eng., Univ. of Hawaii at Manoa, 2540 Dole St., Holmes Hall 405, Honolulu, HI 96822, nosal@hawaii.edu)

Many marine mammal localization techniques rely on estimating the time of arrival delays of a call between the hydrophones of an array. This is commonly achieved by cross-correlating (some version of) received signals between hydrophone pairs. This process is complicated in cases with noise, multiple animals, and multipath arrivals. In this talk, I explore and demonstrate the potential of a multi-channel cross-correlation technique to help establish time delay estimates. Possible extensions to signal enhancement for detection and classification purposes are also discussed. [This work was supported by ONR Award No. N00014-16-1-2598.]

10:45

4aAB11. Turning birds into bats—Multi-modal tracking to study collective behaviour. Jens C. Koblitz, Oren Frokosh, Mate Nagy, Nora Carlson, Hemal Naik, and Couzin Iain (Max Planck Inst. for Ornithology, Univ. of Constance, Constance 78464, Germany, jkoblitz@orn.mpg.de)

Acoustic localization has been used to track numerous vocalizing animals, including whales, bats birds by measuring the time of arrival differences of a sound recorded by multiple receivers. This method, however, requires the species of interest to vocalize regularly to achieve decent temporal resolution. In order to track the movements of birds that do not vocalize regularly, a miniature, radio controlled ultrasonic speaker is attached to the birds and constantly emits ultrasound chirps. The chirps from various tagged animals can be discriminated based on information encoded in the signal. A 30 microphone array covering the ceiling of a large aviary ($14.8 \times 6.6 \times 3.9$ m) is used to record the chirps and provides the basis for accurate localization of the sources. As the chirps do not overlap with the frequency of the animals vocalizations, these can be localized and assigned individually in a flock of moving birds. In addition, a VICON motion capture system provides a simple and accurate method to ground truth the acoustic localizations as it records very precise movement information of the individuals while line of sight is between the marker on the animal and a number of cameras is established. The acoustic and visual tracking systems working together provides a unique and novel answer to consistent and precise localizations of non-vocalizing individuals (and groups of individuals) as they move through space.

11:00

4aAB12. Increasing access to big bioacoustic data through cloud-based systems. Carrie C. Wall, Charles Anderson (Cooperative Inst. for Res. in Environ. Sci., Univ. of Colorado at Boulder, 216 UCB, Boulder, CO 80309, carrie.bell@colorado.edu), Allyson Spring (Cloud Platform, Google, Alexandria, VA), and Jason Gedamke (Sci. & Technol., NOAA NMFS, Silver Spring, MD)

The National Oceanic and Atmospheric Administration's (NOAA) National Centers for Environmental Information (NCEI) has recently developed an archive for the long-term stewardship of passive acoustic data. Protocols for archiving passive acoustic data are currently being established in support of the NOAA Ocean Noise Reference Station Network project, and monitoring marine mammals and fish. Archives maintain data, but access to these data is a core mission of NCEI that allows users to discover, query, and analyze the data in new and innovative ways. To facilitate global access to what will be 100s of TB of bioacoustic data, NCEI has partnered with Google Cloud through the NOAA Big Data Project. Cloud-based access to large volumes of data creates the ability to listen to sound files, download more easily, and bring processing routines to the data instead of bringing the data to the processing.

11:15

4aAB13. Acoustic signature and footprint of artificial nurseries in harbors. Julie Lossent (Res. Institut CHORUS, 22 rue du Pont Noir, Saint Egrève 38120, France, julie.lossent@chorusacoustics.com), Anais Gudefin (Ecocean, Montpellier, France), Cédric Gervaise (Res. Institut CHORUS, Grenoble, France), Amélie Foncuberta, Lecaillon Gilles (Ecocean, Montpellier, France), and Lucia D. Iorio (Res. Institut CHORUS, Grenoble, France)

Underwater soundscapes are used as a cue by vertebrate and invertebrate larvae and post-larvae to locate suitable settlement habitats. In the project SEANAPS, we investigated if soundscapes can play a role in ecological restoration actions. We assessed whether Biohuts® (artificial nurseries installed in harbors) have a distinct acoustic signature from the harbor and the natural environments. We conducted four acoustic measurement sessions in Cap d'Agde (France) and a simultaneous monitoring of vagile fauna present in Biohuts®. Directional listening networks allow us to locate the broadband transient sounds from benthic invertebrates and fish, build maps of these emissions and isolate the biophony emitted by the vagile fauna of Biohuts®. The Biohuts® produced 78 emissions/min/m^2, which represent 200 and 130 times more than a pseudo-natural habitat (seawall) and a natural rocky habitat, respectively. The acoustic footprint of the Biohuts® signatures is 120 m radius in the harbor and 500 m in the harbor road. The harbor road is a naturally silent corridor, which could slow down the larval recruitment by its lack of attractive acoustic signature. This information can be used to resize the network of Biohuts®. Moreover, we demonstrated that the acoustic signature of a Biohut® informs about the vagile fauna it hosts.

11:30

4aAB14. Riverscape meets underwater soundscape: Acoustic habitat selection by brook char in a small stream. Zaccaria Kacem, Marco A. Rodríguez, and Raphaël Proulx (Environ. Sci., Univ. of Quebec in Trois-Rivières, 575 rue des forges, app2, Trois-Rivières, QC G9A 5W4, Canada, zaccaria.kacem@uqtr.ca)

Stream habitats are characterised by geophysical descriptors such as water temperature, depth, substrate type, and flow speed. So far, few studies have focused on underwater sounds as an important feature of habitat selection by fish. In this study, we described stream habitats at high resolution to evaluate the relative importance of the underwater soundscape and other geophysical descriptors for understanding the distribution of brook char densities. Our results showed the high acoustical heterogeneity of stream habitats (ranging from 40 dB up to 150 dB re 1uPa), which was related to differences in water velocity and depth as expected from theory. Brooks char densities were nevertheless positively related to sound intensities, irrespective of water velocity, depth, or facies type. Our findings showed that underwater sounds integrate the many environmental dimensions of stream and may be used as cues for habitat selection. The positive relationship between brook char densities and sound intensities could be related to the high auditory threshold of Salmonidae.

11:45

4aAB15. Long-term characterization of the marine soundscape in Kona, Hawaii. Karlina Merkens (CRP, NOAA/PIFCS (Lynker Tech.), 3710 SW Caldew St., Portland, OR 97219, kmerkens@gmail.com), Erin M. Oleson (NOAA/NMFS/PIFSC, Honolulu, HI), and Simone Baumann-Pickering (UCSD/SIO, La Jolla, CA)

Soundscape analysis of marine environments is still a new field of inquiry, with the majority of studies focusing on individual species or acoustic functional groups. Additionally, the challenges of making long term acoustic recordings in marine habitats, particularly in deep water, have limited the duration of many analyses. The Pacific Islands Fishery Science Center has been collecting passive acoustic data at a depth of 630 m off the Kona coast of the Island of Hawaii since 2007, and has recently begun a soundscape analysis effort to characterize the contributors to the local acoustic environment. This project seeks to examine not only low frequency noise patterns, below 1 kHz, but also higher frequency sounds that may overlap in frequency with the signals of many cetacean species. We have begun by identifying the relative contribution of odontocete echolocation, and also close approaches of boats and the presence of echosounders. The temporal patterns in these signals reveal daily and seasonal cycles in biological and anthropogenic sounds, which provide both detailed insight into the interactions between humans, animals and their environment, and long term trends in activity of those sound sources.

THURSDAY MORNING, 8 NOVEMBER 2018 SIDNEY (VCC), 9:15 A.M. TO 12:00 NOON

Session 4aBA

Biomedical Acoustics: Biomedical Acoustics I

Tyrone M. Porter, Chair
Boston University, 110 Cummington Mall, Boston, MA 02215

Contributed Papers

9:15

4aBA1. Preclinical assessment and enhancement of passive acoustic mapping in clinically realistic propagation environments. Michael Gray and Constantin Coussios (Inst. of Biomedical Eng., Univ. of Oxford, Oxford OX37DQ, United Kingdom, michael.gray@eng.ox.ac.uk)

Passive acoustic mapping (PAM) techniques have been developed to detect, localize, and quantify cavitation activity during therapeutic ultrasound treatments. Building on a decade of *in vitro* and small animal studies, this paper presents a series of pre-clinical experiments and simulations conducted in preparation for an upcoming clinical trial of ultrasound-mediated targeted drug delivery. The effects of tissue attenuation and refraction, array shape and element diffraction, soundspeed uncertainty, and imaging algorithm will be discussed. Critically, techniques will presented for removing factors that would otherwise bias PAM results both within and between patients. Together, the results indicate the potential to significantly enhance both the qualitative and quantitative capabilities of PAM for ensuring clinical therapeutic safety and efficacy. [Work supported by the National Institute for Health Research Oxford Biomedical Research Centre.]

9:30

4aBA2. Attenuation compensation of ultrasound backscatter from human carotid plaque. Sheronica L. James, Russell Fedewa (Biomedical Eng., Cleveland Clinic, 9500 Euclid Ave., ND-20, Cleveland, OH 44195, jamess4@ccf.org), Heather Gornik (Cardiovascular Medicine, Cleveland Clinic, Cleveland, OH), Sean Lyden (Vascular Surgery, Cleveland Clinic, Cleveland, OH), and D. Geoffrey Vince (Biomedical Eng., Cleveland Clinic, Cleveland, OH)

Carotid atherosclerotic plaque composition may be a valuable predictor of stroke risk. Here, a patient adaptive attenuation compensation technique is introduced. A Siemens S3000, 9L4 probe and Axius Direct software were used to acquire radiofrequency (RF) data from 19 subjects prior to carotid endarterectomy. Histology slices of the excised plaque were prepared and matched to ultrasound frames. Regions of interest (ROI) were selected from a homogeneous area within the slide stack and matched to corresponding ROI's in the RF data. ROI's were categorized as fibrous (n = 32), hemorrhagic and/or necrotic core (n = 74), or calcium (n = 32). Additionally, 209 adventitia ROI's were obtained from six normal subjects. ROI power spectra were computed and normalized to a uniform phantom. Five attenuation compensation methods were applied to the spectra: (1) reference phantom with 0.5 dB/cm-MHz attenuation; (2) nominal power spectral shift estimator; (3) optimum power spectral shift estimator; (4) normalized backscatter from adventitia; and (5) patient adaptive two-step attenuation compensation. A linear fit of the resulting estimated backscatter transfer functions (*eBTF*) was performed over the fundamental bandwidth of 3–7 MHz. Only the patient adaptive two-step attenuation compensation distinguished among the means of all four tissue types for the linear mid-band fit parameter.

9:45

4aBA3. Beam coding with orthogonal complementary Golay codes for signal-to-noise ratio improvement in ultrasound mammography. Yasin Kumru and Hayrettin KÖymen (Elec. and Electronics Eng., Bilkent Univ., Ankara 06800, Turkey, yasin.kumru@bilkent.edu.tr)

In this paper, the experimental results of using Golay coded signals at 7.5 MHz to detect breast microcalcifications of 50 μm size is presented. For improving the success on detecting the microcalcifications, orthogonal Golay complementary sequences up to 40 chips having cross correlation for minimum interference are used as coded signals and they are compared to tone burst pulse of equal energy in terms of resolution under weak signal conditions. The measurements are conducted using an experimental ultrasound research scanner, DiPhAS having 256 channels, a phased array transducer with 7.5 MHz center frequency, and the results obtained through experiments are validated by Field-II simulation software. In addition, to investigate the superiority of coded signals in terms of resolution, multipurpose tissue equivalent phantom (Model 84-317) containing series of monofilament nylon targets, 240 μm in diameter, and cyst-like objects with attenuation of 0.5 dB/(MHz*cm) is used in the experiments. We obtained ultrasound images of monofilament nylon targets and cyst-like objects for the evaluation of resolution. Experimental results show that it is possible to differentiate closely positioned small targets with increased success by using coded excitation in very weak signal conditions.

10:00

4aBA4. Experimental reconstruction of sound speed and attenuation profiles. Pedram Mojabi and Joe LoVetri (Elec. and Comput. Eng., Univ. of MB, 75 Chancellor's Circle, Winnipeg, MB R3T5V6, Canada, pedram.mojabi@gmail.com)

Quantitative reconstruction of sound speed and attenuation profiles using ray-based approaches are considered. To this end, experimental datasets from two different ultrasound imaging systems are utilized. The first is our in-house system that is composed of a relatively small number of co-resident fixed piezo-electric transducers. The second, referred to as the MUBI system [Camacho, Archives of Acoustics, 2012], is equipped with many more transducers that can be rotated around the object-of-interest, allowing higher resolution reconstructions. The raw data collected from our system cannot be directly used in the reconstruction algorithms because various sources of noise and systematic errors corrupt the data. We compare the image reconstruction performance when different preprocessing techniques are applied to the raw data, as well as when various techniques of determining the time-of-arrival, such as the Akaike information criterion (AIC), are employed. The quantitative ray-based reconstruction algorithms are tested on simplistic tissue-mimicking phantoms [Mojabi & LoVetri, Submitted to IEEE UFFC, 2018].

10:15–10:30 Break

10:30

4aBA5. Estimation of the thermo-physical property of biological tissues from temperature rise due to ultrasound exposure. Yukako Tsujimoto and Iwaki Akiyama (Doshisha Univ., 1-3 Tatara-Miyakodani, Kyotanabe, Kyoto 610-0394, Japan, ctub1049@mail4.doshisha.ac.jp)

This study discusses the feasibility of a method for ultrasonic tissue characterization according to the thermo-physical properties of biological tissues. Since this method measures the ratio of sound velocity variation due to ultrasonic heating, it is significant to clarify the relationship between the ratio and thermo-physical properties. The ratio of sound velocity variation of a tissue sample was measured from phase shifts of echoes before and after the temperature rise by transmitting an ultrasonic pulsed wave. In the experiments the transducer used for heating was a resonance frequency of 5.0 MHz, concave ring shape, inner diameter of 5 mm, and outer diameter of 12 mm. It was arranged coaxially on the outer perimeter of the transducer used for the measurement. This transducer was a resonance frequency of 10 MHz, a focal distance of 15 mm, and a diameter of 4 mm. The exposure time of heating ultrasound is 100 ms. As a result of heating sound pressure of 1.0 MPa, 1.5 MPa and 2.0 MPa, the estimated values of temperature rise were 0.56 °C, 1.1 °C and 2.0 °C, respectively. [This study was supported by MEXT-Supported Program for the Strategic Research Foundation at Private Universities, 2013-2017.]

10:45

4aBA6. Experimental and numerical investigation of backscattered signal strength from different concentrations of nanobubble and microbubble clusters. Hossein Haghi, Amin Jafari Sojahrood (Phys., Ryerson Univ., 350 Victoria St., Toronto, ON M5B 2k3, Canada, hossein.haghi@ryerson.ca), Al C. De Leon, Agata Exner (Biomedical Eng., Case Western Reserve Univ., Cleveland, OH), and Michael C. Kolios (Phys., Ryerson Univ., Toronto, ON, Canada)

Optimizing the clinical performance of Nanobubbles and Microbubbles (MBs and NBs) requires not only a good understanding of their individual bubble dynamics, but how bubble interactions change these dynamics. Here we report our experimental and simulation results showing the changes in backscattered signal strength from different concentrations of NBs or MBs. NBs and MBs are sorted by size to isolate two populations with mean diameters of 200 nm and 1 micron, respectively. Our results show that increasing number density of NBs or MBs is accompanied with an increase in the strength of their backscattered signals up to a certain concentration, above which the strength of the signal decreases. This decrease is not associated with the medium attenuation which was not accounted for. Our numerical simulations that include inter-bubble interactions are in general agreement with our experimental results. Moreover, acquired data show the that concentration at which there is signal saturation for NBs is orders of magnitude higher than that of MBs (1e9 vs 1e6 bubbles per mL), suggesting a potential explanation behind the very strong backscatter observed in NB clusters compared to MBs. Numerical simulations show that increasing the number density of bubbles decreases the individual signal strength of each individual bubbles.

11:00

4aBA7. Characterization of perfluorocarbon nanodrops with 18-MHz plane waves. Jeffrey A. Ketterling (Lizzi Ctr. for Biomedical Eng., Riverside Res., 156 William St., New York, NY 10006, jkettermeister@gmail.com), Tiffany-Trang Nguyen, and Mario L. Fabiilli (Dept. of Biomedical Eng., Univ. of Michigan, Ann Arbor, MI)

Perfluorocarbon (PF) nanodrops (NDs) undergo a phase change from liquid to vapor with an acoustic stimulation. Activation thresholds are established with relatively high-concentrations of NDs. Activating single NDs with high-frequency plane waves without the need to switch between transmit waveforms would allow uninterrupted high-speed imaging studies to be undertaken in microvasculature. Lipid-coated perfluoropentane (PFP) or perfluorohexane (PFH) NDs were made via sonication (S) or sonication and extrusion (SE). A control sample consisted of lipid blend (LB) without PF. A Verasonics Vantage with an 18-MHz array was used to acquire backscatter data from activated NDs. A test tank with a 200-μm diameter input flow channel was used to pass a diluted concentration of NDs into the image plane of the array. The injection flow rate was 55 mL/hr. Batches of 5 plane waves were transmitted over $\pm$ 5 degrees at an effective PRF of 1000 Hz. Data were collected over 10 s intervals and then processed to count activation events. An event was defined as a backscatter signal 25 dB above background noise. PFPSE showed the highest counts at a given pressure. Agents appeared to be activated prior to entering the image plane. PFPS cases showed a reduced rate of activation but with a similar linear rise. PFH and LB had minimal to no events. The data indicate a small fraction of the injected agents were activated but do not indicate a distinct activation threshold was reached.

11:15

4aBA8. Influence of surfactant encapsulation on the acoustic droplet vaporization. Thomas Lacour, Tony Valier-Brasier, and François Coulouvrat (Institut Jean Le Rond d'Alembert, Sorbonne Université CNRS UMR 7190, 4 Pl. Jussieu, Paris 75005, France, thomas.lacour@upmc.fr)

Nanodroplets have great, promising medical applications such as contrast imaging, embolotherapy or targeted drug delivery. Their functions can be mechanically activated by means of focused ultrasound inducing a phase change of the inner liquid known as acoustic droplet vaporization (ADV) process. In this context, a four-phases (vapor + liquid + surfactant layer + surrounding environment) model of ADV is proposed. Attention is especially devoted to the nonlinear mechanical properties of the surfactant layer encapsulating the metastable bubble/droplet system and described as a model of nonlinear surface tension with two parameters. Various responses to ultrasound excitation are illustrated, depending on surfactant properties and acoustical excitation parameters. Different classes of ADV outcomes are exhibited, and a relevant threshold ensuring complete vaporization is defined. The dependence of this threshold with acoustical, geometrical, and mechanical parameters is also discussed.

11:30

4aBA9. Phase shift nanoemulsions facilitated focused ultrasound nonthermal ablation in mice brain. Chenguang Peng (Biomedical Eng., Boston Univ., Boston, MA), Tao Sun, Natalia Vykhodtseva, Yongzhi Zhang, Chanikarn Power, Nathan McDannold (Radiology, Brigham Women Hospital, Boston, MA), and Tyrone M. Porter (Biomedical Eng., Boston Univ., 110 Cummington Mall, Boston, MA 02215, tmp@bu.edu)

High intensity focused ultrasound (HIFU) thermal ablation is an emerging technique for noninvasive transcranial surgery. One of the major issues of transcranial HIFU thermal ablation is that the pressure and power required to generate lesions is relatively high and could induce skull heating and pain in some cases. Artificial nuclei like phase shift nanoemulsions (PSNE) have been shown to significantly lower the acoustic pressure and power needed to create lesions through inertial cavitation. However, traditional PSNE made from dodecaperfluoropentane (DDFP) were not able to activate below a peak negative pressure (PNP) of 3.5 MPa, which may be difficult to achieve inside the brain. In this study, we proposed to use a more volatile perfluorocarbon—perfluorobutane (PFB)—to synthesize PSNE so that they could be activated at much lower pressures. A transcranial focused ultrasound transducer operating at 740 kHz and 12 CD-1 mice were used to test the vaporization of PFB PSNE and the corresponding damage. The PNP was increased stepwise up to 1.8 MPa and the broadband emission from the inertial cavitation nucleated by vaporized PFB-based PSNE were recorded to quantify the inertial cavitation level. A significant elevation of broadband emission was noticed first at PNP = 1.25 MPa and increased dramatically, indicating strong inertial cavitation. The corresponding biological damage was analyzed with H&E staining and exhibited confined ischemic and hemorrhagic lesions.

11:45

4aBA10. Novel sonification and haptic displays of physiologic variables to improve patient safety. Joseph Schlesinger, Taylor Combs, Megan Holmberg, Seiver Jorgensen, and Samantha Kultgen (Vanderbilt Univ. Medical Ctr., 1211 21st Ave. South, Medical Arts Bldg., #526, Nashville, TN 37212, joseph.j.schlesinger@vanderbilt.edu)

Auditory medical alarms in healthcare are uninformative, poorly localizable, have a low positive predictive value, and are set at thresholds forcing practitioners to be reactive instead of proactive. Sonifying patient physiology as it transitions from normal to abnormal will allow practitioners to respond before the patient status devolves into an emergency. Multimodal interactions have the potential to enhance future sonification tools by reducing alarm fatigue—the addition of haptic cuing to novel physiologic sonification may improve patient safety. To increase the overall efficiency of alarm design in healthcare, we propose reducing alarm fatigue by integrating multisensory streams. We present an experiment that characterizes how the integration of tactile and auditory signals affects the speed and accuracy of alarm responses. Participants received multisensory auditory and haptic input while performing a task designed to tax attentional resources mimicking working in the ICU. Our results indicate there is a trend towards increased perception of change of physiologic variables with concordant haptic stimuli and participants are significantly better at determining the direction of change versus the physiologic variable of change. Future directions include simplification of the sonification schemata and increasing information complexity in the haptic modality to utilize multisensory integration.

Session 4aMU

Musical Acoustics: General Topics in Musical Acoustics

Eric Rokni, Chair
Department of Physics, Rollins College, Winter Park, FL 32789

Contributed Papers

8:55

4aMU1. Are phantom partials produced in piano strings? Thomas Moore, Lauren Neldner, and Eric Rokni (Dept. of Phys., Rollins College, 1000 Holt Ave., Winter Park, FL 32789, tmoore@rollins.edu)

The anomalous frequency components in the sound from a piano, often referred to as phantom partials, were first noted almost 75 years ago. They occur at the sums and differences of the frequencies generated by the transverse motion of the string, and it is generally accepted that they are produced by longitudinal motion of the string that is associated with the normal transverse motion. Experimental evidence will be presented demonstrating that while some of the power in a phantom partial may originate in the string, most of it does not. A simple theory explaining how phantom partials may originate in the wooden components of the piano will be presented.

9:10

4aMU2. Overtone series in variable density strings. Greg S. Elliott (Phys., Univ. of Puget Sound, 1500 N Warner CMB 1031, Tacoma, WA 98416, gelliott@ups.edu), Martin Jackson, and Ethan Russell (Mathematics and Comput. Sci., Univ. of Puget Soud, Tacoma, WA)

The vibrational mode frequencies of a taut string depend on the form of its mass distribution. To lowest order, deviations away from a uniform mass density that are even about the center of the string cause shifts in the overtones away from harmonic, while deviations that are odd do not. The problem of determining the overtone series for a given mass distribution is formulated and a full solution is derived for variations in the inverse mass density that are expressed as a Fourier cosine series. Variations arising from a single Fourier component predominantly shift the relative frequency of a single mode, with the $2n^{th}$ Fourier component shifting the frequency of the n^{th} mode. For a physically reasonable range of mass density variations, the frequency of a mode can typically be shifted by two semitones. Wrapped metal strings with variations in inverse mass density with different Fourier components have been fabricated, characterized, and tested. Light plucking produces overtones in agreement with the model; ordinary plucking produces incommensurate tones at sum and difference frequencies. Overtone series with several different major and minor tones are possible for variations involving more than one Fourier component.

9:25

4aMU3. How different strings affect violin qualities. Lei Fu, Gary Scavone (Computational Acoust. Modeling Lab., Ctr. for Interdisciplinary Res. in Music Media and Technol. (CIRMMT), Schulich School of Music, McGill Univ., 555 Rue Sherbrooke Ouest, Montreal, QC H3A 1E3, Canada, lei.fu2@mail.mcgill.ca), and Claudia Fritz (Lutheries-Acoustique-Musique, Institut Jean Le Rond d'Alembert, Unité Mixte de Recherche 7190, Sorbonne Université / Ctr. National de la Recherche Scientifique, Paris, France)

The brand and model of strings used on violins are considered to play a significant role in their playability and sound quality. An experiment was designed to test the perceptual quality of different violin strings. A professional violinist selected two violins, from a set of the same make/model, that had similar sound and playing qualities. Three different types of strings were chosen for this study: Dominant, Kaplan, and Pro-Arté strings. Professional and advanced student violinists were invited to play and evaluate the violins. The experiment involved three phases: in the first phase, the two violins were strung with the same types of strings; in the last two phases, the strings of one of the violins was changed to the other two different types of strings (in a random manner). Violinists were asked to freely describe the differences of the two violins in each phase and rate them on eight specified criteria. Preliminary results indicate that the violin with Dominant strings was perceived as being more responsive and having a brighter sound, while that with the Pro-Arté strings was considered to have a richer sound and better overall quality.

9:40

4aMU4. Perceptual thresholds of violin soundpost length variation. Lei Fu (Computational Acoust. Modeling Lab., Ctr. for Interdisciplinary Res. in Music Media and Technol. (CIRMMT), Schulich School of Music, McGill Univ., 555 Sherbrooke St. West, Montreal, QC H3A 1E3, Canada, lei.fu2@mail.mcgill.ca), Claudia Fritz (Lutheries-Acoustique-Musique, Institut Jean Le Rond d'Alembert, Unité Mixte de Recherche 7190, Sorbonne Université / Ctr. National de la Recherche Scientifique, Paris, France), and Gary Scavone (Computational Acoust. Modeling Lab., Ctr. for Interdisciplinary Res. in Music Media and Technol. (CIRMMT), Schulich School of Music, McGill Univ., Montreal, QC, Canada)

The soundpost of a violin is an essential component of the instrument, providing structural support and a mechanical coupling between top and back plates. Subtle changes made by a luthier to the soundpost dimensions or position can result in significant variations in sound and playing qualities. A study was designed to investigate the perception of sound quality associated with changes to the soundpost length, making use of an adjustable carbon fiber soundpost. Violin excerpts were recorded with different soundpost lengths. Violinists and luthiers were invited to listen to the recordings and provide perceptual feedback using a computer interface. Tasks included pairwise comparisons and free sorting. Perceptual thresholds in detecting soundpost length changes were characterized, as well as possible groupings according to soundpost lengths that might be considered too short, too long and optimal. Bridge admittances were also measured and analyzed for different soundpost lengths.

9:55

4aMU5. Spherical harmonic expansions of high-resolution directivity data. Samuel D. Bellows and Timothy W. Leishman (Brigham Young Univ., 910 N 900 E Apt. 111, Provo, UT 84604, sbellows@byu.edu)

Quantifying the directivity patterns of sound sources provides not only deeper insights into understanding the source and its acoustical properties but also the potential to simulate the propagation of its sound in various settings. Being able to describe the detailed 3-dimensional directivity of highly complex and dynamic sound sources such as musical instruments and human speech in a simple way while at the same time preserving the fine details of the directivity patterns has broad applications. By using seventeenth-order spherical harmonic expansions of data collected on a 2,664-

point spherical array, large amounts of the measured directivity of musical instruments and human speech was condensed in a simple, accurate, and powerful tool for understanding these complex systems.

10:10

4aMU6. Studying the clarinet with an artificial mouth: Comparison of playing frequencies between model and measurement. Jack D. Gabriel and Whitney L. Coyle (Dept. of Phys., Rollins College, 1000 Holt Ave., Winter Park, FL 32789, jgabriel@rollins.edu)

In the field of musical acoustics, instruments, such as the clarinet, are often played with the use of an artificial mouth or playing machine in order to objectively measure the playing characteristics such as sound levels, playing frequency, regime changes, etc. The purpose of this research was to study the tuning tendencies of the clarinet experimentally and compare this to the models found in literature. A clarinet was artificially blown to determine the playing frequencies for varying levels of blowing pressure and reed opening. These measurements could then be compared to computational models of the clarinet. The experimental results will be presented and compared with the theoretical values from past studies that predicted playing frequencies both analytically and computationally. Finally, suggestions for improving the models will be presented and discussed.

10:25–10:45 Break

10:45

4aMU7. Construction and preliminary studies of the 3-6-(7) tones Oja. Stephen G. Onwubiko (Music, Univ. of Nigeria, Nsukka Enugu, UEnugu, Nsukka 234042, Nigeria, stephen.onwubiko@gmail.com) and Tracianne B. Neilsen (Brigham Young Univ., Provo, UT)

Common in the eastern parts of Nigeria among Igbos, the oja is a Nigerian wooden flute. More like a whistle with its rich sharp tone, the oja is an end blown flute about 14–16 cm long with a narrowed, hollowed cavity running down its entire length and another across its width. These form three holes that can be covered in various combinations with the fingers of both hands, giving the flute a range of about a sixth or a seventh depending on the master player. The mouthpiece is "U" or "V" shaped. With the three holes, it is expected that there would be three distinct pitches produced from the "oja" but more than three tones can be heard or sounded. This paper showcases the construction of oja flute, and presents preliminary studies on how the sixth can be realized. Pictorials, data, and videos of practical experience are presented to support the theoretical discourse.

11:00

4aMU8. Construction and validation of a high resolution finite element model of the Japanese *koto*. Angela K. Coaldrake (Elder Conservatorium of Music, Univ. of Adelaide, Adelaide, SA 5005, Australia, kimi.coaldrake@adelaide.edu.au)

The Japanese *koto* is recognized for its complex resonances, but few studies have been conducted to understand its acoustic properties. This paper presents a COMSOL Multiphysics finite element model of a hand-crafted, professional-grade Japanese *koto*. To do this, the complex internal and external geometry was captured on 2400 CAT scan cross-sections and imported into Comsol. The model was validated by reference to literature data, Chladni patterns, frequency response experiments, acoustic camera and laser scanning vibrometer studies. These results were then compared to the sound as played on the actual instrument. Challenges in the model's development arising from the materials and construction of the instrument are discussed, most notably the anisotropic nature of the *paulownia* wood including multiple grain orientations used in its construction and the added complexity of modeling the organic shape of the real, hand-crafted 1.83 m long instrument with its major internal variations. These developments were guided by parallel studies in less intractable geometry such as simpler box models or models with idealized geometrical shapes lofted along a spline. The CAT scan derived model is used as a quasi-experimental tool to investigate the effect of internal components of the *koto* on its resonances and results presented.

11:15

4aMU9. Kuramoto and Sommerfeld: Toward comprehension of symphony orchestras in concert halls. James B. Lee (None, 6016 S. E. Mitchell, Portland, OR 97206, cadwal@macforcego.com)

Given comprehension of physical acoustics in concert halls, an accurate replacement to the inadequate concatenation of reverberation theory, geometric acoustics, and Beranek's "perceptual categories" that long has held the field in architectural acoustics, it remains to explain how the symphony orchestra couples to the space. Individual instruments do not couple to the space directly, but first to others in their respective sections: first violins to first violins, second violins to second violins, violas to violas, cellos to cellos, basses to basses, and so on through the winds, and others. Each section is synchronized in phase by a Kuramoto mean-field. Then the coherently radiating sections couple to each other asynchronously and create a unique propagating field according to the Sommerfield radiation condition. Kuramoto mean-field coupling is key, for absent that a section's individual instruments would tend to randomize in phase and sum to zero power. The Sommerfeld radiation condition does not guarantee such a uniform field, only a unique one, the power and information of which are conveyed to the audience as constrained by the usual Gabor-Huygens principle: proximity effects and resonant scattering are essential.

11:30

4aMU10. Human-technology interfaces with the tactile metronome. Nathan Hesselink (Univ. of British Columbia, UBC School of Music, 6361 Memorial Dr., Vancouver, BC V6T 1Z2, Canada, n.hesselink@ubc.ca)

The story of the metronome is the story of humankind coming to terms with evolving conceptions of time and coordination as mediated through technologies of the modern age. What began as a tool for regulating and documenting tempo soon became the temporal yardstick by which aspiring musicians strove to emulate in practice and performance. In the early twentieth century the metronome took on the new role of synchronizing live musicians with moving images on a screen (the so-called "click track"), and as the century progressed the metronome would come to dictate the manner in which musicians related to each other in the recording studio and in live events. This paper focuses on the latest manifestation of this phenomenon, the tactile metronome, looking at how vibrotactile stimulation is being used for temporal synchronization as well as an enhanced sense of embodiment for the performer, including haptic feedback. Four modern products tailored to performing musicians will be introduced and analyzed in the overlapping contexts of synchronization and embodiment. It is my argument that the metronome has now come full circle, returning a sense of human feel to the experience of making music.

Session 4aNS

Noise, Speech Communication, and Psychological and Physiological Acoustics: Effects of Noise on Human Performance I

Joonhee Lee, Cochair
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Z. Ellen Peng, Cochair
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Chair's Introduction—8:00

Invited Papers

8:05

4aNS1. Energetic and informational masking effects of hospital noise. Frederick J. Gallun (VA RR&D National Ctr. for Rehabilitative Auditory Res., VA Portland Health Care System, 3710 SW US Veterans Hospital Rd., Portland, OR 97239, Frederick.Gallun@va.gov)

The ability to understand speech in a hospital setting is limited by the overlap of speech and noise energy at the level of the cochlea, sometimes called "energetic masking." However, there are also additional "informational masking" effects that can occur when the noise is comprised of informational sounds such as speech, alarms, or other meaningful auditory events (e.g., Kidd and Colburn, 2017). While energetic overlap has been well-studied and is quantified in a number of international standards such as the Speech Intelligibility Index (ANSI, 2004), there is no similar metric for informational masking. Two of the main difficulties with predicting informational masking, and thus the intelligibility of speech in real world environments, are 1) the lack of tools that defines the meaningfulness of an interfering sound and 2) the much greater variability in susceptibility of individual listeners to informational as opposed to energetic masking. These ideas will be illustrated with data from several recent studies of speech intelligibility with various types of maskers, including hospital noise, and several current and emerging techniques for assessing speech intelligibility for individual listeners in simulated environments will be described.

8:25

4aNS2. Do orienting responses predict annoyance from irrelevant sounds? Jordan N. Oliver (Speech, Lang. and Hearing Sci., Purdue Univ., West Lafayette, IN), Lea Sollmann, Annika Niehl (Systems Neurosci. & NeuroTechnol. Unit, Univ. of Appl. Sci., Saarbruecken, Germany), and Alexander L. Francis (Speech, Lang. and Hearing Sci., Purdue Univ., Speech, Lang. and Hearing Sci., West Lafayette, IN 47907, francisa@purdue.edu)

Unexpected sounds are distracting and can be annoying but individuals may differ in susceptibility to them. Irrelevant sounds occurring at sparse temporal intervals induce a psychophysiological orienting response reflecting involuntary capture of attention away from the primary task. We hypothesize that the frequency and/or magnitude of individual listeners' orienting responses to irrelevant sounds will predict annoyance ratings and task performance in distracting noise. Participants read essays while seated in a comfortable chair in a sound-shielded booth facing a semicircular array of 6 speakers located 1.5 m away at 30°, 60°, and 90° to the left and right. Unintelligible background speech (ISTS) played at 60 dB(A) SPL from each loudspeaker (unsynchronized). At 50–70 s intervals one of 12 non-speech sounds (IADS) played for 6 s from one loudspeaker at approximately 70 dB(A) SPL. Order and location of sounds were randomized but each sound played from each speaker exactly once over the experiment (72 trials, ~80 min). Cardiovascular, electrodermal, electro-ocular, and bilateral posterior auricular muscle activity were recorded from participants to quantify orienting response. Behavioral measures of reading comprehension, noise sensitivity, personality traits, and subjective effort, frustration and annoyance were also collected and will be related to physiological measures.

8:45

4aNS3. Spatial release from masking in reverberation for children and adults with normal hearing. Z. Ellen Peng (Waisman Ctr., Univ. of Wisconsin-Madison, 1500 Highland Ave., Madison, WI 53705, z.ellen.peng@wisc.edu), Florian Pausch, and Janina Fels (Medical Acoust. Group, Inst. of Tech. Acoust., RWTH Aachen Univ., Aachen, NRW, Germany)

The ability to perform spatial release from masking (SRM), i.e., benefiting from a spatial separation between target and distracting talkers, is an important developmental skill for children to navigate noisy environments such as classrooms. While studies on children's SRM were mostly conducted in anechoic rooms or sound booths to date, little is yet known how realistic room acoustics including reverberation in everyday listening affect this ability. In this study, we measured SRM from children and adults with normal hearing in virtual acoustic environments that mimic typical classrooms with different acoustic conditions. Two virtual classrooms were simulated with mean mid-frequency reverberation times (RTs) of 0.4 and 1.1 s, one within and the other poorer than the recommendation of classroom acoustics standard (ANSI S12.60). Overall, children performed more poorly than adults on both speech intelligibility and SRM. Children's speech intelligibility of the target speech masked by distractor voices was better under the less reverberant condition. Interestingly, while adults showed better SRM under RT = 0.4 s, children did not perform any differently between the two RT conditions. The findings will be discussed for further investigation and implications for classroom acoustic designs. [Work supported by the EU 7th Framework Programme, ITN FP7-607139.]

9:05

4aNS4. Children's real-time spoken language processing in the presence of background speech. Tina M. Grieco-Calub, Katherine M. Simeon, and Shana Birger (Northwestern Univ., 2240 Campus Dr., FS, 2-231, Evanston, IL 60208, tinagc@northwestern.edu)

Children's naturalistic environments often contain noise, including background speech and environmental sounds, that disrupts their speed and accuracy of spoken language processing. Integrating congruent auditory and visual speech cues, a skill that improves throughout childhood, facilitates language processing in background noise. The present study implemented an integrated eye-tracking and touch-screen paradigm to explore how children utilize visual speech cues in the presence of background speech and how these behaviors change with age. Typically-developing children (ages 3–12 years), either heard (auditory-only) or heard and viewed (audiovisual) a female speaking sentences (e.g., *Find the dog*) in quiet or in the presence of a male two-talker speech masker at +2 dB SPL. Children were then instructed to select an image, among a set of three, that matched the sentence-final word. During each trial, children's eye gaze was recorded, which allowed us to quantify children's fixations to the target versus the distractor images. These data enable us to make fine-grained measurements of children's language processing in real time. Performance was also quantified by accuracy of target image selection. Discussion will focus on children's spoken language processing in the presence of background speech and in the presence and absence of congruent audiovisual speech cues.

9:25

4aNS5. Effects of background noise in vowel productions of children with cochlear implants. Areti Okalidou (Educational and Social Policy, Univ. of Macedonia, 156 Egnatia St., P.O. Box 1591, Salonika 540 06, Greece, okalidou@uom.edu.gr), Z. Ellen Peng (Waisman Ctr., Univ. of Wisconsin, Madison, WI), Polina Pantazidou (Educational and Social Policy, Univ. of Macedonia, Salonika, Greece), Janina Fels (Inst. for Tech. Acoust., RWTH, Aachen, Germany), Michalis Nistikakis (Educational and Social Policy, Univ. of Macedonia, Salonika, Greece), and George Kyriafinis (Dept. of Medicine, 1st University ENT Clinic of Ahepa Hospital, Aristotle Univ. of Thessaloniki, Salonika, Greece)

Early cochlear implantation led to significant improvements in the speech of deaf children, yet, reduced accuracy and greater token-to-token variability are noted. Vowel spaces are either normal or reduced and vowels are produced with less consistency (e.g., Baudonck *et al*. 2011; Neumeyer *et al*. 2010). So far, studies investigated speech in quiet. In the real world, children with cochlear implants (CI) function in noisy environments with auditory feedback containing speech and noise which is known to alter speech production even in typically developing adults (e.g., Bottalico *et al*., 2015). Since children with CI produce less robust vowels, this study aimed to examine the effects of everyday noise on their productions. Eight children with CI, 6;10–12;0 years old, produced words with Greek vowels while listening to auditory stimuli provided directly via auxiliary input port to the speech processor. The conditions were: a) quiet, b) speech-shaped noise, and c) speech-shaped noise with reverberation. Acoustic measurements of F1 and F2, pitch, duration and intensity were made. Apart from inter-subject variability, results indicated changes in pitch, F1, and intensity with added noise along with occasional changes in duration and F2. Individual analysis focused on age and vowel space changes as the acoustic conditions changed.

9:45

4aNS6. Relationship between perception of speech in noise and disordered speech: Influence of sensorineural hearing loss. Sarah E. Yoho and Stephanie A. Borrie (Communicative Disord. and Deaf Education, Utah State Univ., 1000 Old Main Hill, Logan, UT 84321-6746, sarah.leopold@usu.edu)

It is well known that there exists substantial individual variability in the ability to understand speech in adverse listening conditions. Despite this variability, a recent study with normal-hearing listeners has revealed a strong relationship between the ability to perceive speech in noise (environmental degradation) and dysarthric speech (source degradation) [Borrie *et al*., *Journal of Acoustical Society of America,*141, 4660–4667 (2017)]. While a large body of literature on the difficulty faced by hearing-impaired listeners in understanding speech in noise exists, the difficulties faced by this population in understanding dysarthric speech has received much less attention. Further, investigations into the relationship between processing speech in noise and dysarthric speech for listeners with hearing loss do not exist. This current study extends on previous findings, investigating the relationship between processing speech in noise and dysarthric speech for listeners with sensorineural hearing loss. Preliminary results replicate previous findings of a relationship between the ability to perceive speech in noise and dysarthric speech for normal hearing listeners. This relationship is also observed for hearing-impaired listeners; however, these listeners perform substantially better with dysarthric speech relative to speech in noise. The complex interplay between hearing loss and type of degradation will be discussed.

10:05–10:20 Break

10:20

4aNS7. Investigating individual susceptibility to the detrimental effects of background noise and reverberation in simulated complex acoustic environments. Kristi M. Ward (Northwestern Univ., 2240 N. Campus Dr., Rm. 2-381, Evanston, IL 60208, kmward@u.northwestern.edu), Z. Ellen Peng (Waisman Ctr., Univ. of Wisconsin-Madison, Madison, WI), Maryam Landi (National Ctr. for Physical Acoust. (NCPA), Univ. of MS, Oxford, MS), Andrew Burleson, Pamela Souza, and Tina M. Grieco-Calub (Northwestern Univ., Evanston, IL)

On a daily basis, individuals perform a variety of tasks in complex acoustic environments that contain background noise and reverberation. Previous research has demonstrated that, when tested in a laboratory setting, background noise and reverberation impair speech recognition and disrupt cognitive processing. These detrimental effects are especially pronounced for young children, older adults, and individuals with hearing loss. However, the specific environmental and individual factors that account for performance declines in the presence of background noise and reverberation, and whether these relations are generalizable to real-world complex acoustic environments, remains poorly understood. In the present study, children and adults performed speech recognition and speech comprehension tasks amidst background noise and reverberation in a state-of-the-art virtual sound room (ViSoR). ViSoR contains a Variable Room Acoustics System, which simulates the acoustic properties of real-world environments in a free-field test setting. Participants also completed standardized measures of attention, auditory working memory, and receptive vocabulary, which will be used to quantify the extent to which individual factors contribute to the observed changes in speech recognition and comprehension. Together, the findings from this study will provide additional insight as to the factors that underlie individual susceptibility to the detrimental effects of background noise and reverberation.

10:40

4aNS8. Effects of measured acoustic and other indoor environment factors on student achievement in K-12 classrooms. Kieren H. Smith (Durham School of Architectural Eng. and Construction, Univ. of Nebraska-Lincoln, Brigham Young Univ., Provo, Utah 84602, kierenhs@gmail.com), Ann Arthur, James Bovaird (Univ. of Nebraska-Lincoln, Lincoln, NE), and Lily M. Wang (Durham School of Architectural Eng. and Construction, Univ. of Nebraska-Lincoln, Omaha, NE)

Students can be affected by what they hear, see, and feel within the indoor classroom environment, whether or not it is cognizantly perceived by them. To understand how the indoor environment affects students' performance, measurements were logged in 220 K-12 classrooms in two Midwestern states during three seasons with two occupied days per season. Measurements of acoustics, indoor air quality, thermal comfort, and lighting were taken and a variety of metrics were calculated. For example, acoustic measurements provided reverberation time, clarity index, and equivalent sound levels during both active and inactive portions of the occupied school day. Achievement data in the form of percentile ranks on math and reading tests were also collected for the students who received instruction in each classroom. Structural equation models and multivariate linear regression models were utilized to analyze the effect of the indoor environment factors on student math and reading achievement. Significant findings, particularly in the area of acoustics, are presented and related to previous work in this area. [Work supported by the United States Environmental Protection Agency Grant Number R835633.]

11:00

4aNS9. Acoustic conditions for students' engagement in active learning classrooms. Shiva Hadavi and Joonhee Lee (Dept. of Bldg., Civil and Environ. Eng., Concordia Univ., 1515 Saint-Catherine St. W, Montreal, Montreal, QC H3G 2W1, Canada, shiva.hadavi1988@gmail.com)

Educators have developed innovative teaching strategies in order to maximize learning outcomes in classrooms. Active learning classrooms are new learning spaces that facilitate the teaching strategies with enhanced students' engagement and collaborative discussions. Previous studies showed that the design of the space impacts on students' achievement. However, acoustic requirements of the active learning classrooms have not been investigated yet. This paper presents, thus, the acoustic conditions of the active learning classrooms located in Montreal. The acoustical parameters such as background noise, reverberation time, and speech transmission index in unoccupied conditions and occupied noise levels during learning activities are examined. The results are compared with conventional classrooms. Since active learning classrooms are more prone to acoustic deficiencies, the results of this paper can provide a better understanding of the acoustic design requirements for these spaces.

11:20

4aNS10. VR-based realistic assessment of annoyance for the noise of children running in apartment buildings. Hyun In Jo, Shahzad Ahmed, Hyun Wook Kim, Chun Ki Seo, and Jin Yong Jeon (Dept. of Architectural Eng., Hanyang Univ., Hanyang University, Seoul, Seongdong-gu 133-791, South Korea, best2012@naver.com)

This study assessed annoyance response to temporal variations of sound energy of heavy floor impact in apartment buildings by using a virtual reality environment. As for a real moving sound source, the noise of a child running diagonally on the floor was reproduced by applying HRTF to the sound source of a single impact ball. Annoyance was assessed in the acoustic environment where visual information was provided by using HMD and directional information was given by using HRTF. A comparison scale using the equal-appearing interval (EQI) was adopted to reflect the sense of daily life. As a result, when visual information was provided in the VR environment, the annoyance of subjects decreased. On the other hand, directional information was given, the subjects became more sensitive to annoyance. Consequently, annoyance was more affected by the direction of a heavy moving sound source than the visual information. The VR-based environment of this study turned out to be realistic in a pyschoacoustic experiment.

11:40

4aNS11. Lombard effect in restaurant setting: How much would you spend to eat at this restaurant? Pasquale Bottalico (Speech and Hearing Sci., Univ. of Illinois Urbana-Champaign, 901 S. 6th St., Champaign, IL 61820, pb81@illinois.edu)

The objective of this study is to determine the minimum level of noise in a restaurant that starts the Lombard effect, and how it relates to the perceived communication disturbance and the willingness to spend time and money for a meal. Twenty-eight participants were instructed to read a passage in the presence of restaurant noise from 35 dB(A) to 85 dB(A). As the noise level increased, participants began to be disturbed by the noise at 52 dB(A) and began to raise their voice at 57 dB(A). The willingness to spend time and money decreased starting at 52 dB(A).

THURSDAY MORNING, 8 NOVEMBER 2018 COLWOOD 1/2 (VCC), 8:30 A.M. TO 11:35 A.M.

Session 4aPA

Physical Acoustics and Biomedical Acoustics: Interactions of Sound Beams with Objects I

Likun Zhang, Cochair
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Grant Eastland, Cochair
System Acceptance & Operational Readiness, Naval Undersea Warfare Center Division Keyport, 610 Dowell St., Keyport, WA 98345

Invited Papers

8:30

4aPA1. Particle assembly and object propulsion using acoustic holograms. Kai Melde and Peer Fischer (Micro, Nano and Molecular Systems, Max Planck Inst. IS, Heisenbergstr. 3, Stuttgart, BW 70569, Germany, melde@is.mpg.de)

The contact-free manipulation of particles using ultrasound fields is an active field of research promising a number of applications. Conventional acoustic tweezers use strongly focused beams or higher order Bessel beams to provide a trap for single particles to be manipulated with. Other more mature methods use resonators to create elongated potential wells for collective particle trapping or separation. The resulting assemblies have a limited complexity, because the fields are highly symmetric. We recently introduced the acoustic hologram as an alternative way to create arbitrary ultrasound fields. In this talk, I will present two concepts, one for particle trapping and one for propulsion of objects, that have been enabled by this new method. The first is parallel assembly of microparticles at a surface in the shape of a projected acoustic image. Using a global trigger, these particles can be fused together to form a mechanically stable object. The second demonstration is a seemingly dynamic effect resulting from our static hologram. By defining the phase gradient (essentially the wave vector) along the water-air interface, it is possible to continuously propel objects along predefined trajectories. The necessary complexity to create such ultrasound fields with defined amplitude and phase distribution is easily managed using acoustic holograms.

8:50

4aPA2. Acoustic tweezer and its biomedical applications. K. K. Shung (Biomed Engr, Univ of S. Calif., 136 DRB, 1042 Downey Way, Los Angeles, CA 90089, kkshung@usc.edu)

Acoustic tweezer, the counterpart of optical tweezer in acoustics, was developed more than 10 years later than optical tweezer. It has been found to be capable of performing many tasks similar to optical tweezer but substantial differences also exist. Because of its larger wavelength, acoustic tweezer manipulates larger particles and yields larger forces in the nanonewton range. A number of biomedical applications have been studied including measuring intercellular forces and cellular mechanical properties. These results along with its physical principles will reviewed in this paper.

9:10

4aPA3. The development of dynamic holographic acoustic tweezers. Asier Marzo and Bruce W. Drinkwater (Mech. Eng., Univ. of Bristol, University Walk, Bristol BS8 1TR, United Kingdom, b.drinkwater@bristol.ac.uk)

Radiation forces due to sound waves can be used to create acoustic tweezers that can trap and manipulate matter without contact. They provide unique advantages when compared to the more established optical tweezers, such as higher trapping forces per unit input power and the ability to trap a wide range of sample materials. This paper describes the development of dynamically reconfigurable holographic acoustic tweezers that can independently manipulate multiple millimetre-scale particles. We present hardware and algorithms that create converging acoustic radiation forces at multiple locations in space with an array of phase controlled emitters. We experimentally demonstrate a 40 kHz airborne ultrasonic system and manipulate up to 25 particles simultaneously. As the acoustic field is dynamically up-dated in real-time and manipulation speeds of up 40 mm/s are shown. When considered on the scale of a wavelength, this system has similar manipulation capabilities to optical tweezers. We show experimental results that demonstrate potential applications, e.g., for assembly processes both in the micro-metre and millimetric scale, as well as positioning and orientation of multiple object which could led to biomedical applications.

9:30

4aPA4. Applications of acoustic radiation force for material characterization and imaging. Mostafa Fatemi (Physiol. & Biomedical Eng., Mayo Clinic College of Medicine & Sci., 200 First St. SW, Rochester, MN 55905, fatemi.mostafa@mayo.edu) and Azra Alizad (Radiology, Mayo Clinic College of Medicine & Sci., Rochester, MN)

Acoustic radiation force has been used in many interesting medical and industrial applications in the past two decades. The fact that radiation stress can be confined to a small focal area and exerted remotely inside or on an object allows one to use this phenomenon in a variety of ways to explore properties of the object. Vibro-acoustography is a material characterization and imaging method that uses harmonic or impulsive acoustic radiation force to stimulate the targeted material and receives the resulting acoustic response. The output acoustic signal is then used to extract important information about the object. Biomedical applications of vibro-acoustography and its closely related methods include imaging, flow detection, and tissue characterization. Other applications include non-destructive evaluation/testing of materials, modal analysis of small structures, and microscopy. This talk presents an overview of vibro-acoustography and its closely related techniques over the past 20 years. Various biomedical and industrial applications will be presented. The talk will conclude with a discussion on potential future applications of this technology.

9:50

4aPA5. Compact selective tweezers based on focalized acoustical vortices and spiraling interdigitated transducers. Michael Baudoin, Jean-Claude Gerbedoen, Antoine Riaud, Olivier Bou Matar (IEMN, Univ. Lille, CNRS, Centrale Lille, ISEN, Univ. Valenciennes, UMR 8520, IEMN, Ave. Poincaré, Villeneuve d'Ascq 59652, France, michael.baudoin@univ-lille1.fr), Nikolay Smagin (IEMN, Univ. Lille, CNRS, Centrale Lille, ISEN, Univ. Valenciennes, UMR 8520, Valenciennes, France), and Jean-Louis Thomas (INSP, Sorbonne Universités, CNRS UMR 7588, Paris, France)

With the emergence of regenerative medicine, cell printers, and labs on chips, the contactless selective manipulation of microscopic objects such as particles, cells, or drops has become a key feature. To complete this task, acoustic tweezers appear as a tremendous alternative to their magnetic and optical counterpart. Indeed, they do not require pre-tagging of the manipulated object and they enable particles trapping with forces several orders of magnitude larger than optical tweezers at same input power. Recently, Baresh *et al.* [*Phys. Rev. Lett.*, **116**, 024301 (2016)] demonstrated the selective 3D manipulation of particles with a specific class of waves called acoustical vortices. Nevertheless, such manipulation was achieved with a complex transducer array coupled with a high end programmable electronics. This system is cumbersome, not compatible with microscopes and hardly miniaturizable. To overcome these difficulties, our team developed new tweezers [Riaud *et al.*, *Phys. Rev. Appl.* **7**, 024007 (2017)] based on spiraling interdigitated transducers (IDTs), some electrodes sputtered at the surface of piezoelectric substrates patterned by photolithography. The shape of the electrodes encodes the phase of the field like a hologram. For applications, these tweezers have many attractive features: they are selective, flat, easily integrable, and compatible with disposable substrates.

10:10–10:25 Break

10:25

4aPA6. Phase-shift based expansions for material and frequency dependence of scattering and of radiation forces. Philip L. Marston (Phys. and Astronomy Dept., Washington State Univ., Pullman, WA 99164-2814, marston@wsu.edu)

When considering the scattering of sound and radiation forces for spheres, it has historically been helpful to gain a good understanding of situations lacking dissipation. In that case the linear acoustic scattering by spheres is characterized by real partial-wave phase shifts. At low frequencies it is possible to obtain simple expansions showing the dependence of each phase shift on material properties and on frequency [P. L. Marston and L. Zhang, *J. Acoust. Soc. Am.* **141**, 3042–3049 (2017); P. L. Marston, J. Acoust. Soc. Am. 142, 1167–1170 (2017); P. L. Marston, J. Acoust. Soc. Am. 142, 3358–3361 (2017)]. Those expansions can then be used to describe the frequency and material dependence of scattering and radiation forces for standing and traveling waves at low frequencies beyond the usual Rayleigh-scattering approximation. While spherical Bessel and Hankel functions are used in the derivation, the final expansion coefficients are expressed using algebraic functions. The derivation of the simplified series expansion for the radiation force in the traveling wave case is subtler than the standing wave case. Well-known limiting expressions are recovered in both cases. Some applications to spheres in acoustic beams are also known. [Work supported by ONR.]

10:45

4aPA7. Dipolar and quadrupolar mode dissipation of spherical probes spinning in vortex beam acoustical tweezers. Diego Baresch (Dept. of Chemical Eng., Imperial College London, London SW7 2AZ, United Kingdom, diego.baresch@upmc.fr), Régis Marchiano (Institut Jean Le Rond D'Alembert, UPMC, Paris, France), and Jean-Louis Thomas (Institut des NanoSci. de Paris, UPMC, Paris, France)

In this work, the possibility of simultaneously trapping and rotating single polystyrene beads, or clusters, against gravity with an ultrasonic vortex beam is demonstrated. The induced rotation of a single particle is compared to a torque balance model accounting for the acoustic response of the particle. Two dominating dissipation mechanisms of the acoustic orbital angular momentum responsible for the observed rotation are examined. The first takes place in the bulk of the absorbing particle, while the second arises as the angular momentum flux is dissipated in the viscous boundary layer surrounding the particle. Importantly, even in the long-wavelength (Rayleigh) regime, it is crucial to fully model the dissipation of the dipolar and quadrupolar vibrational modes of the particle in a slightly viscous fluid such as water. Further results suggest that, while the induced outer (Eckart) rotational flow can be neglected in water, it can play an important role in other viscous and complex fluids.

Contributed Papers

11:05

4aPA8. Measurement and calculation of lateral trapping strength of focused beams generated by a two-dimensional ultrasound array. Mohamed A. Ghanem (Aeronautics and Astronautics Eng., Univ. of Washington, Box 352400, Seattle, WA 98195-2400, mghanem@uw.edu), Adam D. Maxwell (Urology, Univ. of Washington, Seattle, WA), Vera A. Khokhlova (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab, Univ. of Washington, Seattle, WA and Phys. Faculty, Moscow State Univ., Seattle, Washington), Oleg A. Sapozhnikov (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab, Univ. of Washington, Seattle, WA and Phys. Faculty, Moscow State Univ., Moscow, Russian Federation), and Michael R. Bailey (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab, Univ. of Washington, Seattle, WA)

This study is aimed at developing a method to controllably manipulate kidney stones in the human body using a single ultrasound source. Calculation and measurement of the lateral force on a sphere using various acoustic beams (vortex and radially-varying phased beams) were compared. A 256-element, 1.5-MHz array was used to synthesize the beams. Spheres of 1–6 mm ($6 < ka < 20$) diameter made of glass, ceramic, or brass were positioned on an acoustically matched platform at the focus rigidly attached to the transducer by a frame. The transducer and platform were rotated to the angle at which the trapped sphere fell. The acoustic power was <9 W and was adjusted by the duty cycle (10–60%) to control the range of the trapping angle. Maximum temporal average intensity I_{SPTA} was 46.5 W/cm^2. The acoustic force was calculated numerically as in Sapozhnikov and Bailey [JASA, 133, 616 (2013)] and the angle calculated with static force equilibrium equation. Good agreement between calculation and measurement was observed, with an average error in angle measured of 11.3%. The maximum lateral forces were 80% of the axial radiation force and 52% the gravitational force. [Work supported by NIH P01-DK043881, K01-DK104854, R01EB7643, and RBBR 17-02-00261.]

11:20

4aPA9. Acoustic non-diffracting tractor beams with lateral trapping. Xudong Fan (Univ. of MS, University, MS) and Likun Zhang (Univ. of MS, 145 Hill Dr., Oxford, MS 38677, zhang@olemiss.edu)

In spite of the recent exploration of non-diffracting tractor beams that pull a particle by nonconservative forces, the pulling over a long distance is not realistic without the simultaneous trapping in the lateral direction by gradient forces. Trapping of a small particle by both ordinary and vortex Bessel beams is examined in the parameter space of mass density and compressibility. The results reveal beam parameters for stable trapping. It is found that under certain conditions the trapping is independent on paraxiality parameter of the beam. The findings are validated by calculations for a variety of objects. Material parameters for the simultaneous trapping and pulling of a small particle by an ordinary Bessel beam is identified. These findings pave the way for experimental realization of stable acoustic tractor beams for a small particle.

Session 4aPP

Psychological and Physiological Acoustics: Acoustics Outreach: Linking Physiology and Behavior for Future Collaborations I

Amanda Lauer, Cochair
Otolaryngology-HNS, Johns Hopkins University School of Medicine, 515 Traylor, 720 Rutland Ave., Baltimore, MD 21205

Anna C. Diedesch, Cochair
Communication Sciences & Disorders, Western Washington University, 516 High St., MS 9171, Bellingham, WA 98225

Chair's Introduction—9:15

Invited Papers

9:20

4aPP1. Challenges and opportunities in bridging behavior, physiology, and anatomy in translational hearing research. Amanda Lauer, Ye-hyun Kim, and Katrina Schrode (Otolaryngology-HNS, Johns Hopkins Univ. School of Medicine, 515 Traylor, 720 Rutland Ave., Baltimore, MD 21205, alauer2@jhmi.edu)

The Acoustical Society of America has a long, rich history of physiology research aimed at understanding the neural mechanisms underlying hearing and hearing dysfunction. In particular, this research has included pivotal studies in animal models that reveal the neurobiology of hearing and hearing disorders. More recently, research in animal models and research in human subjects have been considered in completely separate conferences or sessions. This situation presents a barrier to communicating important scientific information and fostering collaborations that bridge the gap between research in animal models and human listeners. It is crucial that the two types of research interface and inform one another to maximize translational potential and converge on fundamental knowledge. Experimental manipulations can be performed in non-human animals that cannot be performed in humans, but human and non-human animals can also be tested using similar techniques to increase translatability. Examples of mouse models tested using detailed analysis of synapse-level changes, clinically relevant evoked potentials, and behavioral tests will be presented. Areas for mutually beneficial interdisciplinary investigation will be discussed with the intention of inspiring new lines of communication between researchers focused on non-human animal and human subjects research.

9:40

4aPP2. Importance of translational research in audiology. Anna C. Diedesch (Commun. Sci. & Disord., Western Washington Univ., 516 High St., MS 9171, Bellingham, WA 98225, anna.diedesch@wwu.edu)

Students, postdocs, and junior research-faculty in audiology all face an interesting challenge: which conferences should I attend, and can I afford the time and likely financial burden associated with attending several meetings in one year? As research shifts towards translational topics, it becomes difficult to remain siloed within our respective fields. It is increasingly important for clinical researchers in audiology to communicate with specialized medical doctors, physiologists, engineers, psychoacousticians, etc. An example of such technological advancements that may contribute to the field of audiology are neuroimaging techniques and signal processing capabilities. For instance, enhanced imaging techniques allow for mapping the auditory pathway for different spatial cue information and auditory prostheses are now capable of retaining or supplementing binaural information through signal processing. But how many patients visit an audiologist with complaints of sound localization ability? In this example, it is critical for clinical researchers to be aware of physiological and technological advancements in order to appropriately translate spatial hearing research to the clinic. Here, I will discuss the benefit of clinicians attending translational auditory conferences by highlighting binaural hearing research and why attendance at translational conferences such as the Acoustical Society of America is essential for clinical research in audiology.

10:00

4aPP3. From behavior to physiology and back again: The role of auditory cortex in vocal production and control. Steven Eliades and Joji Tsunada (Otorhinolaryngology: Head and Neck Surgery, Univ. of Pennsylvania, 3400 Spruce St., 5 Ravdin, Philadelphia, PA 19104, seliades@pennmedicine.upenn.edu)

Vocal communication plays an important role in the lives of both humans and many animal species. Ensuring accurate communication, however, requires auditory self-monitoring to control vocal production and rapidly compensate for errors in vocal output. Despite the importance of this process, the underlying neural mechanisms are relatively unknown. Previous work has demonstrated that neurons

in the auditory cortex are suppressed during vocal production, while simultaneously maintaining their sensitivity to vocal feedback, suggesting a role in auditory self-monitoring. The behavioral role of auditory cortex in vocal control, however, remains unclear. We investigated the function of auditory cortical activity during vocal self-monitoring and feedback-dependant vocal control in marmoset monkeys. Using real-time frequency-shifted feedback during vocalization, we demonstrate that marmosets exhibit rapid compensatory changes in vocal production, a feedback-dependent behavior that is predicted by the activities of neurons in auditory cortex. We further establish the role of auditory cortex in vocal control using electrical microstimulation to evoke rapid changes in produced vocalizations. These findings suggest a causal role for the auditory cortex in vocal self-monitoring and feedback-dependent vocal control, linking mechanisms of production and perception, and have important implications for understanding human speech motor control.

10:20

4aPP4. Blast exposure impairs sensory gating: Evidence from measures of acoustic startle and auditory event-related potentials. Melissa A. Papesh and Frederick J. Gallun (National Ctr. for Rehabilitative Auditory Res., Portland VA Medical Ctr., 4000 SW 91st Court, Portland, OR 97225, Melissa.Papesh@va.gov)

Many military Veterans who have been exposed to high-intensity blast waves suffer from traumatic brain injury (TBI), resulting in chronic auditory deficits despite normal hearing sensitivity. The current study examined the neurological cause of this chronic dysfunction by testing the hypothesis that blast exposure leads to impaired filtering of sensory information at brainstem and early cortical levels. Groups of blast-exposed and non-blast-exposed participants completed self-report measures of auditory status, auditory perceptual tasks, and physiological measures of sensory gating, including prepulse-inhibition and habituation of the acoustic startle reflex and electrophysiological assessment of a paired-click sensory gating paradigm. Blast-exposed participants showed significantly reduced habituation to acoustic startle stimuli and impaired filtering of redundant sensory information at the level the auditory cortex. Linear regression analyses revealed that poorer sensory gating at the cortical level was primarily influenced by a diagnosis of TBI, while reduced habituation was primarily influenced by a diagnosis of posttraumatic stress disorder. A statistical model was created including cortical sensory gating and habituation to acoustic startle, which strongly predicted performance on a degraded speech task. These results support the hypothesis that blast exposure impairs central auditory processing via impairment of neural mechanisms underlying habituation and sensory gating.

10:40–10:55 Break

10:55

4aPP5. Subcortical neural correlates of behavior in the auditory system. Ramnarayan Ramachandran (Dept. of Hearing and Speech Sci., Vanderbilt Univ. Medical Ctr., 111 21st Ave. South, Wilson Hall 065, Nashville, TN 37240, ramnarayan.ramachandran@vanderbilt.edu)

One of the biggest open questions in neuroscience is the relationship between neuronal activity and behavior. Little is known in the auditory system about the nature of the relationship between stimulus encoding and perception. To shed light on this topic, we recorded responses of single neurons in the central nucleus of the inferior colliculus (CNIC) and cochlear nucleus (CN) of macaques that were trained to detect tones or vowels during behavioral performance. Using signal detection theoretic methods, we determined that neurometric tone and vowel detection thresholds based on rate measures in CNIC matched behavioral thresholds, while thresholds of CN neurons were higher than behavior. Additionally, the response magnitudes of a majority of the CNIC neurons varied significantly depending on the behavioral accuracy, compared to a much smaller proportion of CN neurons. Analysis of temporal patterns of response (e.g., phase locking) suggests that neurometric thresholds of all CN and most CNIC neurons matched behavioral thresholds during vowel detection. Additionally, every CNIC and CN neuron showed significant response pattern differences based on behavioral accuracy. These results suggest that behaviorally relevant coding early in the pathway is temporal in nature and is transformed to more rate based measures at higher levels of processing.

11:15

4aPP6. Suprathreshold hearing in middle age and relationship to cochlear synaptopathy. Hari M. Bharadwaj, Brooke Flesher, Alexandra Mai, Kelsey Dougherty, Jennifer M. Simpson, and Michael G. Heinz (Purdue Univ., 715 Clinic Dr., Lyles-Porter Hall, West Lafayette, IN 47907, hbharadwaj@purdue.edu)

Animal models have demonstrated that the afferent synapses and nerve terminals innervating the cochlea are vulnerable to damage from acoustic overexposure and aging. This synaptopathy can occur without hair-cell loss, or more severely when accompanied by permanent audiometric shifts. In humans, postmortem temporal bone studies have shown that cochlear synaptopathy occurs throughout adulthood, decades before audiometric loss nominally occurs. However, effective non-invasive assays of synaptopathy have yet to be established in humans (or genetically heterogeneous animal cohorts). Moreover, whether synaptopathy contributes to age-related temporal perception deficits is debated. We are currently studying young and middle-aged humans with clinically “near-normal” audiograms using physiological and perceptual measures. Preliminary results suggest that although high-frequency audiometric shifts occur with age as previously known, middle age per se is associated with physiological effects consistent with cochlear synaptopathy. Further, we find that the effects of age and threshold elevation can oppose and obscure one another on certain perceptual measures. Finally, with speech-in-noise measures designed to degrade high-frequency envelope cues, perceptual deficits associated with middle age per se are apparent and consistent with synaptopathy. These results will be discussed in light of the ongoing debate about the prevalence and consequences of synaptopathy in humans.

11:35

4aPP7. The role of central processing in modulation masking release. Nima Alamatsaz and Antje Ihlefeld (Biomedical Eng., New Jersey Inst. of Technol., 323 Martin Luther King Blvd, Fenster Hall, Rm. 645, Newark, NJ 07102, ihlefeld@njit.edu)

When background sound is present, hearing impaired (HI) individuals and cochlear-implant (CI) listeners typically are worse at hearing out target sound as compared to normal-hearing (NH) listeners. This perceptual deficit occurs both when the background consists of noise that fluctuates over time ("modulated") and for stationary background noise ("unmodulated"). In addition, the difference in thresholds between tone detection in modulated and unmodulated noise, referred to as modulation masking release (MMR), is much reduced or absent in HI and CI as compared to NH. Both peripheral and central processing mechanisms contribute to MMR. We previously showed that central MMR is reduced in human CI listeners, and that sound deprivation reduces central MMR in Mongolian gerbils. Here, we began to explore the neurobiological basis of central MMR. NH gerbils were trained to hear out target tones (1 kHz) in modulated (10-Hz rectangularly gated) versus unmodulated bandlimited background noise, and chronically implanted with recording electrodes in core auditory cortex. Neural discharge was analyzed as a function of the broadband energy ratio between target and background sound to determine how different types of background sound affect neural information transmission in awake behaving gerbil. Preliminary results will be discussed in the context of how hearing loss may affect central MMR.

THURSDAY MORNING, 8 NOVEMBER 2018 UPPER PAVILION (VCC), 9:00 A.M. TO 12:00 NOON

Session 4aSC

Speech Communication: Speech Production (Poster Session)

Laura L. Koenig, Chair

Haskins Labs and Long Island University, 300 George Street, New Haven, CT 06511

All posters will be on display from 9:00 a.m. to 12:00 noon. To allow authors an opportunity to view other posters in their session, authors of odd-numbered papers will be at their posters from 9:00 a.m. to 10:30 a.m. and authors of even-numbered papers will be at their posters from 10:30 a.m. to 12:00 noon.

Contributed Papers

4aSC1. Prominence-marking mechanisms in emotional speech. Marcus Vinicius Moreira Martins (Faculdade de Letras, Universidade Federal de Minas Gerais, Avenida Antônio Carlos, 6627, Belo Horizonte, Minas Gerais 31270901, Brazil, marcusvmmartins@gmail.com) and Waldemar Ferreira-Netto (Departamento de Letras Clássicas e Vernáculas, Universidade de São Paulo, São Paulo, São Paulo, Brazil)

The aim of this work is to investigate the prominence-marking mechanisms in emotional speech, particularly in Brazilian Portuguese intonational system. Three professional actresses interpreted a text in three conditions: anger, sadness and neutral, in a total of 130 sentences for each condition (n = 390). Others tests established the minimum value for a prominence mark is +3 semitones from the mean fundamental frequency (F0). In the first analysis, it is observed the distribution of prominences frequencies (F0) is normal for neutral, but non-normal in the other two cases (skewness; anger = -1.45 and sadness = -0.5) and the difference between distributions is significant in all cases. In the rage and neutral conditions, the prominence is marked consistently 5 semitones above fundamental frequency mean, non-random, however in sadness the prominence-marking mechanism is almost random. A second frequency distribution test revealed the difference between distributions is significant between anger and other conditions ($p < 0.05$), but it is not significant between sadness and neutral speech ($p > 0.05$). These data suggest that the distribution of frequencies in the pitch range combined with an analysis of the prominences may be a relevant criterion to distinguish the sadness from other emotions, especially in automatic recognition process.

4aSC2. Neighborhood-conditioned coarticulation effects in French listener-directed speech. Rebecca Scarborough (Linguist, Univ. of Colorado, 295 UCB, Boulder, CO 80309, rebecca.scarborough@colorado.edu), Cécile Fougeron (LPP, CNRS/Université Sorbonne Nouvelle, Paris, France), and Luciana Marques (Linguist, Univ. of Colorado, Boulder, CO)

Words from dense phonological neighborhoods (Hi-ND words) are realized with greater vowel hyperarticulation and increased coarticulation in English, relative to words from sparser neighborhoods (Lo-ND words) [e.g., Wright 2004]. Here, the relation between coarticulation and neighborhood density is investigated for French by looking at patterns for anticipatory and carryover nasal coarticulation, coronal and uvular CV coarticulation, and V-to-C rounding coarticulation. Twelve native French speakers (northern France) produced 82 disyllabic words of French containing a context for one of these 5 types of coarticulation in phonetically similar high-low ND pairs. Test words were produced to a real listener who had to transcribe them in the instructed positions on a grid. F1, F2, and A1-P0 (a spectral measure of nasality) were measured at 5 timepoints across each test vowel. In addition to having more hyperarticulated (peripheral) vowel midpoints, Hi-ND words showed greater anticipatory nasality and stronger coronal transitions in appropriate contexts than Lo-ND words. Carryover nasality and uvular and rounding coarticulation did not differ across NDs. These results can be interpreted in light of the relation between coarticulation and contrast and the role these effects may serve in making Hi-ND words, which are subject to greater lexical competition, easier to perceive.

4aSC3. Speech-like movements emerge from simulated peri-oral muscle activation space without neural control. Jonathan M. de Vries (Interdisciplinary Studies Graduate Program, Univ. of Br. Columbia, 270, 2357 Main Mall, Vancouver, BC V6T 1Z4, Canada, devriesj@alumni.ubc.ca), Ian Stavness (Computational Sci., Univ. of SK, Saskatoon, SK, Canada), Sid Fels (Elec. And Comput. Eng., Univ. of Br. Columbia, Vancouver, BC, Canada), and Bryan Gick (Linguist, Univ. of Br. Columbia, Vancouver, BC, Canada)

Muscle synergies have been proposed as a solution to the degrees of freedom problem inherent in the biomechanics of speech production [Gick & Stavness, 2013, *Front. Psych, 4*, 977]. However, the majority of experiments involving the extraction of muscle synergies prove theoretically insufficient to determine whether such synergies are of neural origin or simply reflect the lower dimensionality of an under-sampled biomechanical/neural task space [Kutch & Valero-Cuevas, 2012, *PLoS computational biology*, *8*(5), e1002434]. Therefore, to what extent biomechanics of the human vocal tract may constrain the "ecological state space" of a speaker during locution remains uninvestigated. As a proof of concept, we created a simplified version of the peri-oral region using FEM modeling in a physics-based simulator (ArtiSynth [Lloyd *et al*., 2012, *Soft tissue biomech…surgery,* pp. 355]). Systematic simulations enabled us to model the full kinematic/biomechanical space. Visualization of the resulting biomechanical state space using t-SNE [van der Maaten & Hinton, 2008, *J.MLR*., *9,* 2579] illustrates that speech-like movements (e.g., lip rounding or spreading) emerge as self-organizing structures (muscle synergies) critically without a direct neural controller. These results are discussed in the context of current and future explorations of motor control of speech. [Funded by NSERC.]

4aSC4. Frequency effects in palatalization: Articulatory evidence from English. Jae-Hyun Sung (Yonsei Univ., 50 Yonsei-ro, Seodaemun-gu, Seoul 03722, South Korea, jsung@yonsei.ac.kr)

Lexical frequency and its influence on speech production have been widely acknowledged across languages and varieties, and frequency effects are so prevalent in various phonological and coarticulatory phenomena. Previous experimental studies have reported that high lexical frequency leads a greater chance of overt coarticulation (for example, more frequent overt palatalization of /d/ in *would you* (high-frequency) than *bad you* (low-frequency)). This study tests for articulatory evidence of frequency effects in palatalization by examining palatalized consonants resulted by lexical and post-lexical palatalization in English using ultrasound imaging. Comparisons of tongue contours of palatalized consonants produced by 12 native speakers of American English show that differences in lexical frequency are not directly linked to articulatory gestures. That is, speakers do not necessarily produce a greater degree of palatalization (i.e., tongue contour closer to the palate) in high-frequency words or phrases. Moreover, while most speakers maintain some degree of articulatory contrast in high- vs. low-frequency contexts, the way speakers differentiate palatalized consonants from high- vs. low-frequency items is highly individualized. The findings from this study are in line with individual variation in coarticulation, and merit further exploration in the role of lexical frequency in speech production.

4aSC5. Lexically conditioned phonetic variation: A test with the voicing contrast in Japanese. Keiichi Tajima (Dept. of Psych., Hosei Univ., 2-17-1 Fujimi, Chiyoda-ku, Tokyo 102-8160, Japan, tajima@hosei.ac.jp), Kiyoko Yoneyama (Dept. of English, Daito Bunka Univ., Tokyo, Tokyo, Japan), and Mafuyu Kitahara (Dept. of English Studies, Sophia Univ., Tokyo, Japan)

Many lexical factors have been shown to influence phonetic realization of words. For example, studies have shown that voice onset time (VOT) of word-initial stops is shorter in high-frequency words than in low-frequency words, and is longer in words that form a voicing minimal pair, e.g., *cod-god*, than in words that do not, e.g., *cop-*gop*. The present study begins to ask whether such lexically conditioned phonetic variations are language-general, by examining productions of words in Japanese. The stimuli were two-mora Japanese minimal pairs contrasting in word-initial /k/ vs. /g/, half of which were real words, e.g., /kara/, while the other half were similar-sounding nonwords, e.g., */kapa/. Furthermore, half of the items had a lexical competitor contrasting in voicing, e.g., /kara/-/gara/, while the other half did not, e.g., /kana/-*/gana/. The stimuli were split so that each participant read only one member of each minimal pair. Twenty-four native Japanese speakers read the target items interspersed with filler items. Results showed opposite trends from those previous reported. Specifically, VOT for /k/-initial words was *longer* for (high-frequency) real words than for (low-frequency) nonwords, and was *shorter* for words that had a lexical competitor than for words that did not. [Work supported by JSPS.]

4aSC6. Vowel formant trajectories in naturalistically produced loud speech. Susanne Fuchs (Leibniz-Zentrum Allgemeine Sprachwissenschaft, Berlin, Germany) and Laura L. Koenig (Haskins Labs and Adelphi Univ., 300 George St., New Haven, CT 06511, koenig@haskins.yale.edu)

Louder speech, relative to typical speech, may show changes in addition to the expected increased SPL, such as higher values of the first formant (F1), longer durations, and higher articulatory velocities. Most past work has assessed vowel midpoints only. To what extent can any formant differences be observed across the full time course of the vowel? Here we evaluate formant trajectories in loud and conversational speech. Adult female speakers of German produced acronyms containing /a: i: u:/, and words that put various vowels in a bilabial$_{\text{alveolar}}$ context. Greater loudness was elicited by increasing speaker-experimenter distance. Preliminary analyses extracting values at 25%, 50%, and 75% of the vowel duration show that formant trajectories in normal and loud speech are usually quite parallel throughout the vowel, particularly for high and tense vowels (where loudness differences are minimal overall). In some cases normal-loud differences may increase from the 50% to 75% time points. In general, it does not appear that articulatory/acoustic differences between loud and normal speech are restricted to vowel midpoints. We will also carry out analyses over shorter time intervals to evaluate very early and late regions in the vowels.

4aSC7. Differences in clear speech strategies across read and spontaneous speech produced in interactive tasks for young and older adults. Valerie Hazan and Outi Tuomainen (Speech, Hearing and Phonetic Sci., UCL, Chandler House, 2, Wakefield St., London WC1N 1PF, United Kingdom, v.hazan@ucl.ac.uk)

This study investigates whether clear speech strategies vary according to the demands of the task used for eliciting clear speech. Spontaneous speech involves greater planning than sentence reading and this might particularly affect talkers who find communication more effortful. 77 talkers (24 young and 53 older adults) carried out two collaborative tasks with a conversational partner in two listening conditions. The "read" task involved reading a sentence to the partner who repeated it back; the "spontaneous" task involved completing the diapix problem-solving task. Partners either heard each other easily (NORM) or communicated with the partner having a simulated profound loss, naturally eliciting clear speech (CLEAR). The percentage relative change between NORM and CLEAR was calculated for articulation rate, long-term average spectrum, fundamental frequency, vowel formant ranges. Analyses focus on task and age effects. There were significant task [$F(1,75) = 10.27$;$p = 0.002$, $\eta2 = 0.886$] and task by measure effects [$F(5,375) = 17.97$;$p<0.001$, $\eta2 = 0.193$], with a greater reduction in articulation rate for read (M = -27.3%) than spontaneous (M=-9.2%) speech, $p<0.001$. Task effects did not vary significantly with talker age. In summary, the use of read speech may over-emphasise the role of articulation rate as a clear speaking style strategy but this was the case for young and older talkers.

4aSC8. Do speakers maintain clear speech throughout natural conversation? Dae-yong Lee and Melissa M. Baese-Berk (Dept. of Linguist, Univ. of Oregon, 161 Straub Hall, 1290 University of Oregon, Eugene, OR 97403, daeyongl@uoregon.edu)

Previous studies have shown that speakers tend to speak more clearly when they speak with a listener in adverse listening conditions in naturalistic settings. Most previous studies assumed that when speakers speak more clearly, they maintain the style throughout the conversation. However, a clear speech style deviates from speakers' conversational speaking style, and extra effort is required in order to produce clear speech. Therefore, it is possible that speakers begin a conversation in a clear speaking style and

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revert back to a conversational style at some point during the conversation. This study aims to examine whether native English speakers use and maintain a clear speaking style while participating in a conversation with non-native speakers in naturalistic settings. Native English listeners will respond to sentences from naturalistic conversations. The sentences are from either a conversation between two native English speakers or a conversation between a native English speaker and a non-native English speaker. Intelligibility (typing what the listeners heard) and comprehensibility (rating how easy it is to understand the sentences) are measured to examine the usage of clear speech throughout the conversations. The results of this study will provide a better understanding of the circumstances under which speakers use clear speech.

4aSC9. Geminated liquids in Japanese. Maho Morimoto (Linguist, Univ. of California, Santa Cruz, 1156 High St., Santa Cruz, CA 95064, mamorimo@ucsc.edu) and Tatsuya Kitamura (Faculty of Intelligence and Informatics, Konan Univ., Kobe, Hyogo, Japan)

Liquid geminates are uncommon and disfavored in Japanese native phonology. However, instances of emphatic expressions and loanwords from languages with liquid geminates such as Italian or Arabic suggest that they are not impossible. The current study examines the durational, acoustic, and articulatory properties of geminated liquids in Japanese to obtain further insights into the nature of /r/ in Japanese and the process of gemination in general. We report on a production experiment whereby eight native speakers of Japanese pronounced mimetics of the form CVCVCVCV with and without emphatic gemination (e.g., kirakira > kir:akira "shiny"). In addition to the audio, tongue movements were recorded using the EMA technology. Preliminary durational analysis suggests that singleton-geminate ratio is about 1:3, which is slightly larger than the ratios previously reported for geminated obstruents. We explore what articulatory strategies speakers employ to lengthen the liquid consonant whose prototypical singleton production is [ɾ].

4aSC10. Final devoicing in Singapore English. Daryl Chow (English Lang. & Lit., National Inst. of Education, Singapore, Singapore, Singapore) and Viktor Kharlamov (Lang., Linguist & Comparative Lit., Florida Atlantic Univ., 777 Glades Rd, CU-97, Ste 280, Boca Raton, FL 33431, vkharlamov@fau.edu)

This study investigates the voicing contrast in word-final obstruents in Singapore English. Previous auditory accounts have suggested that word-final obstruents undergo complete devoicing in Singapore English and become identical to their voiceless counterparts, including neutralization of differences in adjacent vowel duration (among others, Bao 1998, Wee 2008). On the basis of acoustic data from the NIESCEA corpus (Low 2015), we examine the extent to which voicing-related differences are neutralized when reading wordlists, sentences and passages as well as in unscripted conversations. Results suggest that neutralization is phonetically incomplete in Singapore English, with underlying voicing being recoverable from such parameters as closure voicing ratio and durations of consonants and adjacent vowels. These findings are in line with the results for other devoicing languages, such as German and Russian, that also show incomplete neutralization of the voicing contrast in final obstruents (e.g., Roettger *et al.* 2014, Kharlamov 2014).

4aSC11. Positional allophony in ejective stops: A case study of Georgian. Chloé Gfeller (UFR Linguistique, Université Paris Diderot - Paris 7, 5 rue Thomas Mann, UFR Linguistique - à l'attention de Mme Chitoran, Paris cedex 13 75205, France, chloe.gfeller@gmail.com)

Many phonetic studies have dealt with ejective stops, but allophonic differences between initial and intervocalic positions have not been investigated. Ejectives have been typologically classified as either "stiff" or "slack," depending on their glottal tension (Kingston 1985). Since stops in general tend to show positional acoustic differences superficially similar to those induced by differences in glottal tension (notably differences in VOT duration), we ask whether positional allophony of ejectives manipulates glottal tension. In an acoustic study of Georgian ejectives, we observed longer mean VOTs in initial position for labial (28 ms vs 23 ms) and coronal ejectives (33 ms vs 25 ms), consistent with increased glottal tension. Additionally, in a subset of cases, the vowel following an intervocalic ejective is completely glottalized, consistent with decreased glottal tension in intervocalic position. However, we found no significant difference overall in the mean duration of glottalized vowel phonation after ejectives between initial and intervocalic positions, notably because, in the other intervocalic tokens, the duration of vowel glottalization is comparable to that in initial position. This suggests that the observed positional differences in VOT in Georgian ejective stops are not reducible to differences in glottal tension.

4aSC12. The role of vertical larynx movement in the tense-lax contrast of Seoul Korean stops across phrasal position. Yoonjeong Lee and Louis Goldstein (Linguist, Univ. of Southern California, 3601 Watt Way, GFS 301, Los Angeles, CA 90089, yoonjeol@usc.edu)

In contemporary Seoul Korean (SK), phrasal tone patterns are co-active with tone patterns that distinguish tense and lax stops. Utilizing real-time MRI, the current investigation of the SK stops elucidates the articulatory synergies that distinguish tense versus lax and how they function within SK's Accentual Phrase (AP) prosodic system. The f0 and corresponding larynx height values were measured from sequences containing the three oral stops of SK placed in AP-initial and AP-internal prosodic positions. Larynx height was obtained by finding the intensity centroid within a locally selected region in the MR image around the larynx. We find a strong *positive* correlation between f0 and larynx height, indicating that vertical larynx movement is engaged in some tonal manipulation. We confirm the *categorical* consonant effect on f0 AP-initially (tense >> lax), compared to a *gradient* effect AP-internally (tense >= lax). In AP-initial position, the larynx height results mirror the f0 results, suggesting that the f0 goals associated with tenseness result in large part from a vertical larynx position manipulation. In AP-*internal* position, however, larynx height does not differentiate tense versus lax consonants, suggesting that some other articulatory activity, such as vocal fold stretching, is responsible for its f0 variations. [Work supported by NIH.]

4aSC13. The diphthongs of the acrolectal Kenyan English spoken by the Black Indigenous Kenyans. RUTH W. IRUMBI (Commun., Lang. and Linguist, Pan Africa Christian Univ., Nairobi, Nairobi 01000, Kenya, wrndungu@gmail.com), Mathew K. Karia (Special Needs, Kenyatta Univ., Nairobi, Nairobi, Kenya), and Joshua M. Itumo (English Lang. & Linguist, Kenyatta Univ., Nairobi, Nairobi, Kenya)

Kenya has three main varieties of English: the variety spoken by white Kenyans (WhKE), the acrolectal Kenyan English spoken by the Black Indigenous Kenyans (BIKE), and the mesolectal varieties, which are ethnically marked (Kioko & Muthwii, 2004; Hoffmann, 2010; Njoroge, 2011).There have been calls to describe the acrolectal variety of English .This paper documents the BIKE diphthongs based on data from a research carried out in 2016. The participants, seven males and seven females, were university lecturers who were purposively selected from the three major language groups in Kenya (Bantu, Cushitic and Nilotic). These respondents read "*The Boy Who Cried Wolf*", a passage adapted for IPA English phonemic analyses. Four tokens for each of the eight RP diphthongs were sampled giving us 448 tokens. The recorded audio files were converted into *.wav* files for acoustic analysis using *Praat* (Version 6.0.05). The findings presented herein, show that KenE has six diphthongs: / ai/, /oi/, /ua/, /ia/, /ea/, /ua/ and /au/, which are acoustically different from the eight RP diphthongs. In this paper, we explain how the BIKE diphthongs differ from the RP ones; describe how they cluster based on the gliding; and also map them in the vowel trapezium.

4aSC14. Canadian French rhotics and their continued change. Justin T. Craft and Tamarae Hildebrandt (Linguist, Univ. of Michigan, 440 Lorch Hall, 611 Tappan St., Ann Arbor, MI 48109, juscraft@umich.edu)

In previous decades, Canadian French has undergone a sound change throughout Western Quebec where the apical trill [r] has moved to a production using the tongue dorsum, often being produced by speakers as a velar fricative [x, ɣ], uvular trill [ʀ], or uvular fricative [χ, ʁ] (Sankoff and Blondeau 2007, 2010). This change is argued to have been primarily driven by younger female speakers adopting the dorsal production, but change is also reported across the lifespan of older speakers. Discrepancies about whether variation in rhotic productions is the result of allophony from syllable position (Sankoff and Blondeau 2010) or whether variation is socially constrained (Milne 2012) also exist in the literature. This study presents an analysis of rhotics in harmonic words as produced by 9 speakers from Montreal, Quebec. We report preliminary results highlighting the acoustic differences in production (measuring center of gravity, F1, & F0) between speakers of different genders. These results support the hypothesis that variation is socially constrained, in addition to lending support to early reported hypotheses about the role of women as a driving force of this sound change (Sankoff *et al.* 2001).

4aSC15. Regional variation in West Yorkshire filled pauses: Implications for forensic speaker comparisons. Erica Gold, Sula Ross, and Kate Earnshaw (Dept. of Linguist and Modern Lang., Univ. of Huddersfield, Queensgate, Huddersfield HD1 3DH, United Kingdom, e.gold@hud.ac.uk)

In the current linguistic literature, West Yorkshire (a county in Northern England) has received relatively little commentary, as it is often overshadowed by other bigger regions and cities like Manchester or Newcastle-upon-Tyne. However, in forensic phonetics, literature on regional variation is often vital to forensic casework. Sociophonetic studies aid forensic phoneticians in making judgments regarding whether speaker characteristics are typical of a region or not. For both forensic and sociophonetic motivations, this paper begins to look at the variation present in West Yorkshire by analyzing variation in filled pauses across speaking styles from three boroughs within West Yorkshire (Bradford, Kirklees, and Wakefield) from the West Yorkshire Regional English Database (WYRED; Gold *et al.*, 2016). This paper analyses 60 speakers from WYRED. All speakers are male, aged 18-30, and English is their first and only language. This study measures the vocalic portion of all "uh" /V/ and "um" /V + N/ productions in over 8,000 tokens across four different tasks. Filled pauses were manually segmented in Praat and F1, F2, and F3 midpoints were extracted. Results suggest that filled pauses may be influenced by speaking style, but more importantly the three regions exhibit some significant differences in their filled pause realizations.

4aSC16. Distributional patterning of reflex cough during telephone conversations. Mairym Llorens (Linguist, Univ. of Southern California, 3601 Watt Way, Grace Ford Salvatori 301, Los Angeles, CA 90089-1693, llorensm@usc.edu)

Reflex cough and speech are two behaviors that make use of the same end effectors—the respiratory and vocal tract subsystems. How do talkers engaged in conversation resolve conflicts between the communicative necessities involved in speech planning and their physiological reflexes? Coughing is a complex reflex arc triggered in the airway that leads to an urge to cough and, at some latency, the deployment of a coordinated vocal-respiratory response to that urge. However, research on factors that influence urge-cough latency is limited. Hypothesizing that talkers may actively delay coughing, this present study examines 200 instances of reflex cough from the Switchboard and CallHome corpora of spontaneous, dyadic telephone speech. Coughing events in both corpora are always flanked by acoustic silences, with the majority of coughing events occurring at interlocutor turn exchanges. Assuming that urges to cough can arise at anytime, we interpret the restriction of cough occurrence to pauses and turns as evidence that talkers can and do delay coughing until the end of their intonational phrases or to floor exchanges with their interlocutor. This further indicates a degree of linkage between the non-speech motor system and the prosodic speech planning system. [Work supported by NIH.]

4aSC17. Tongue bracing under bite block perturbation. Monika M. Luszczuk (Inst. of Speech Therapy and Appl. Linguist, Maria Curie Sklodowska Univ. (UMCS), Lublin, Poland, ul. Sowińskiego 17, Lublin 20-040, Poland, monika.luszczuk@gmail.com), Murray Schellenberg (Dept. of Linguist, Univ. of Br. Columbia, Vancouver, BC, Canada), Yadong Liu (Dept. of Linguist, Univ. of Br. Columbia, Hong Kong, Hong Kong), and Bryan Gick (Dept. of Linguist, Univ. of Br. Columbia, Vancouver, BC, Canada)

Speech sounds have been shown to adapt quickly but imperfectly under bite block perturbation, supporting opposing acoustic vs. articulatory compensation mechanisms [Gay *et al. JASA* 69: 802. 1981; Flege *et al. JASA* 83: 212. 1998]. The present study considers whether lingual bracing may provide insight into these apparently conflicting findings. Tongue bracing against the teeth or palate is a pervasive posture maintained during normal speech [Gick *et al. JSLHR*. 60:494. 2017]; we aim to test whether the tongue adapts its bracing position rather than adapting each speech movement individually, providing a single, postural parametric mechanism for responding to jaw perturbation. Results of an experiment will be presented in which native English-speaking participants read aloud passages normally and under bite block conditions translating the jaw in forward, backward or lateral directions, and to varying degrees of opening. Coronal ultrasound imaging results will be reported, measuring positions of the lateral tongue for indications of stable bracing postures. Implications of these findings will be discussed for models of speech production. [Funding from NSERC.]

4aSC18. Lateral bias in lingual bracing during speech. Bryan Gick, Megan Keough, Oksana Tkachman (Linguist, Univ. of Br. Columbia, 2613 West Mall, Vancouver, BC V6T1Z4, Canada, gick@mail.ubc.ca), and Yadong Liu (Linguist, Univ. of Br. Columbia, Hong Kong, Hong Kong)

Human bodies exhibit lateral biases between many laterally symmetrical body parts (e.g., hands, feet, eyes, and ears) that increase the efficiency of behaviour and functionality of a system. We report on lateral biases observed in the tongue during speech. The tongue is bilaterally braced against the back teeth and hard palate throughout speech, but is interrupted for the production of some laterals and occasional low vowels; some evidence suggests the movement away from the braced posture may be produced by lowering one side of the tongue first and that the leading side is consistent within speaker [Gick *et al.* 2017, JSLHR 60(3), 494]. We report findings on lateral bias in English speakers, on its correlation with other lateral biases of the speakers, and what this may imply about the origins of this bias. Preliminary results indicate some variation, with a population-level bias (preference for one side over the other), suggesting that the bias may develop with cortical modulation in much the same way that handedness is thought to arise. [Funding through NSERC.]

4aSC19. Cross-linguistic lateral bracing: An ultrasound study. Felicia Tong (Linguist, Univ. of Br. Columbia, 6368 Stores Rd., Rm. 4, Vancouver, BC V6T 1Z2, Canada, feliciatong@yahoo.com.hk), Yadong Liu (Linguist, Univ. of Br. Columbia, Hong Kong, Hong Kong), Dawoon Choi (Psych., Univ. of Br. Columbia, Vancouver, BC, Canada), Megan Keough, and Bryan Gick (Linguist, Univ. of Br. Columbia, Vancouver, BC, Canada)

Lateral bracing refers to intentional stabilizing of tongue contact with the roof of the mouth along the upper molars or the hard palate. Previous research has found evidence of lateral bracing in individual speakers of six languages [Cheng *et al.* 2017. *Can. Acoust.* 45, 186]. The current study examines lateral bracing cross-linguistically at a larger scale using ultrasound technology to image tongue movement. We tracked and measured the magnitude of vertical tongue movement at three positions (left, right, and middle) in the coronal plane over time using Flow Analyzer [Barbosa, 2014. *J. Acoust. Soc. Am.* **136**, 2105] for optical flow analysis. Preliminary results across all languages (Cantonese, English, French, Korean, Mandarin, and Spanish) show that the sides of the tongue are more stable than the center and maintain a relatively high position in the mouth throughout speech. The magnitude of movement at the sides is significantly smaller than at the center of the tongue. Further, lateral releases vary in frequency for different languages. This evidence supports the view that bracing is a physiological property of speech production that occurs irrespective of the language spoken. [Funding from NSERC.]

4aSC20. Coarticulation of speech and smile movements. Terrina Chan, Ryan C. Taylor, Esther Y. Wong, and Bryan Gick (Linguist, Univ. of Br. Columbia, 2613 West Mall, Vancouver, BC V6T 1Z4, Canada, terrinachan@gmail.com)

Facial expressions and speech movements can impose conflicting demands on articulators. For example, the lip spreading movement associated with smiling is incompatible with bilabial closures for /m/, /b/ or /p/. Anecdotal evidence suggests this conflict may resolve as labiodental stop

variants (see http://phonetic-blog.blogspot.ca/2012/03/u.html), though this discussion has been controversial [Ladefoged & Maddieson 1996, p. 18]. The simplest model of coarticulation—one of unmediated superposition of muscle activations [Gick *et al*., 2013, *POMA* 060207]—predicts that the outcome of this conflict should be determined by summing opposing forces due to competing muscle activations. If so, varying degrees of smile and varying degrees of closure force (e.g., for different stop consonants) should be expected to produce distinct outputs. Previous work suggests that closures for /m/, /b/ and /p/ vary (increasingly) in both intraoral pressure [Lubker & Parris 1970, *JASA* 47: 625] and muscle force [Gick *et al*. 2012, *JASA* 131: 3345]. An experiment will be presented in which bilabial stops are produced under varying smile conditions. Preliminary results indicate that labiodental stop variants occur more frequently for lower-force stops under higher-force smile conditions, as predicted. Implications for models of coarticulation will be discussed. [Funding from NSERC.]

4aSC21. A computational study of the role of laryngeal muscles in vocal fold posturing. Xudong Zheng, Ngoc Pham, Biao Geng, and Qian Xue (Mech. Eng., Univ. of Maine, 213 Boardman Hall, Orono, ME 04469, xudong.zheng@maine.edu)

This work aims at using a three-dimensional finite element method to simulate the contraction of different laryngeal muscles, and to study the role of each or groups of muscles in vocal fold posturing. A Hill-based contractile model is coupled in the finite element analysis to capture the active response of the vocal folds, and a fiber-reinforced model is employed to model the passive response. A 1D flow model and 1D acoustic solver are coupled with the vocal fold model to simulate the flow-structure-acoustics interaction. In the simulation, the geometry of the vocal folds and cartilages is reconstructed from a MRI scan. The coupled Hill-based contractile model and fiber-reinforced tissue model shows good agreement with literature data for simulating dynamics, concurrent tissue stimulation and stretching. It is found that the contraction of cricothyroid (CT) muscle alone lengthens the vocal folds by rotating the thyroid cartilage clockwise, around the cricoid-thyroid joint, whereas the contraction of cricoarytenoid (CA) muscle causes vocal fold abduction. The contraction of TA muscle and arytenoid muscles, on the other hand, adduct the vocal folds. The effect of muscle contraction on subglottic pressure during voice production is also shown in flow-structure-acoustics interaction simulations.

4aSC22. Effects of tongue elevation speed and glottal flow control on the production of sibilant /s/. Tsukasa Yoshinaga (Graduate School of Eng. Sci., Osaka Univ., 1-3 Machikaneyama, Toyonaka, Osaka 560-8531, Japan, t.yoshinaga@me.es.osaka-u.ac.jp), Kazunori Nozaki (Osaka Univ. Dental Hospital, Suita, Japan), and Shigeo Wada (Graduate School of Eng. Sci., Osaka Univ., Toyonaka, Japan)

To clarify the effects of tongue elevation speed and glottal flow control on the production of /s/ in VCV sequence, the experimental measurements were conducted using a simplified vocal tract model. The simplified model was constructed based on five cross-sectional shapes of the vocal tract of a subject pronouncing /s/, which were measured with CT scans. The tongue elevation speed and volume flow rate at the glottal inlet of the model were controlled by using a stepper motor and an electromagnetic proportional control valve, respectively. The flow velocity at the gap between front teeth and the sound propagating from the model were measured by a hot-wire anemometer and a microphone simultaneously. By controlling the tongue elevation speed and the flow rate of the simplified model, the spectrogram of /s/ in Japanese word /usui/ was reproduced. The result of the flow measurement showed that the velocity fluctuation and the sound propagation started almost at the same time when the spectrogram was reproduced. In contrast, by lowering the tongue elevation speed, the velocity fluctuation preceded appearance of the sound generation. This indicates that the tongue elevation speed has a significant effect on controlling the sound source generation of /s/ in word pronunciation.

4aSC23. Using ultrasound to examine contrastive hyperarticulation. Kathleen C. Hall, Geoff Fullerton (Linguist, Univ. of Br. Columbia, 2613 West Mall, Vancouver, BC V6T 1Z4, Canada, kathleen.hall@ubc.ca), and Kevin McMullin (Dept. of Linguist, Univ. of Ottawa, Ottawa, ON, Canada)

The current paper examines the extent to which there is cross-linguistic evidence for the hyperarticulation of more phonologically contrastive vowel sounds. Hall *et al*. [2017, *JCAA* 45: 15] found that tense vowels in English are produced with more total tongue movement when in positions where they are more phonologically contrastive. This finding was established through the use of Optical Flow Analysis [Horn & Schunck 1981] on ultrasound videos of the tongue. The "more contrastive" positions were those in which the vowels could contrast with their lax vowel counterparts, while the "less contrastive" positions were those in which no such contrast was possible. While there was evidence that the degree of contrast affected the tongue movements, the data were somewhat confounded by the fact that the more contrastive positions were largely closed syllables, while the less contrastive positions were largely open syllables. In the current paper, we first replicate the original results with more tightly controlled phonetic contexts in English and then examine analogous results for Canadian French. Crucially, in Canadian French, [e] vs. [ɛ] contrast in open syllables and not in closed, such that the effects of syllable position and phonological contrast can be teased apart. [Funded by SSHRC.]

4aSC24. Registration and fusion of 3D head-neck MRI and 3D/4D tongue ultrasound. Steven M. Lulich (Speech and Hearing Sci., Indiana Univ., 4789 N White River Dr., Bloomington, IN 47404, slulich@indiana.edu)

Static 3D head-neck MRI can provide excellent coverage of the entire vocal tract and surrounding anatomy with good spatial resolution and signal-to-noise ratios. Real-time MRI takes advantage of these strengths and adds temporal resolution for dynamic speech production, but depends on regular repetition of the same speech utterances for several minutes. (Dynamic or cine MRI also depends on regular repetitions, but the process is stroboscopic rather than reconstructive, and the end result is a reduced number of time frames with longer frame durations.) Real-time MRI is not suitable for non-repetitive speech utterances. In contrast, 3D/4D ultrasound can provide excellent spatial resolution with good temporal resolution of non-repetitive speech utterances, but with lower signal-to-noise ratios and poor anatomical coverage. This study presents methods for registering and fusing static head-neck MRI and 3D/4D tongue ultrasound data, taking advantage of the strengths of both MRI and 3D/4D ultrasound to investigate the production of non-repetitive speech utterances.

4aSC25. Towards higher precision in vocal tract length estimation. Stefon M. Flego (Linguist, Indiana Univ., Bloomington, 720 S College Mall Rd. Apt. N5, Bloomington, IN 47401, sflego@indiana.edu)

Approaches to vocal tract length (VTL) estimation typically differ in how much credibility is allocated to certain formants as predictors of length, but are alike in allocating equal credibility to all vowel spectra. The latter may be problematic, as asymmetries in the phonotactic frequency of certain vowels and measurement errors for higher formants can influence VTL estimates. Herein, an additional parameter is proposed which privileges vowel spectra that approximate a uniform tube. The proposed metric for this proximity is standard variance (σ) in Φ, where $\Phi{=}F_n/(2n\text{-}1)$, n represents the integer label of the formant, and σ_Φ for a uniform tube is 0. The five estimators detailed in Lammert & Narayanan (2015) were tested on the speech of two adult males. Each of the estimators were run using all vowel spectra as well as only those for which $\sigma_\Phi < 50$ Hz. Estimates from all vowel spectra had σ_{VTL} well above 1 cm, which is undesirably large, given that the adult vocal tract only ranges from 13 to 20 cm. However, estimates from vowel spectra with $\sigma_\Phi < 50$Hz had σ_{VTL} less than 1 cm. This higher precision is especially striking since spectra with $\sigma_\Phi < 50$ Hz constituted a small fraction of the total.

4aSC26. Individual differences and their effect on the nature of speech motor errors in younger and older adults. Katherine M. Dawson (Speech-Language-Hearing Sci., City Univ. of New York Graduate Ctr., 365 5th Ave., New York, NY 10016, kdawson2@gradcenter.cuny.edu), Mark Tiede (Haskins Labs., New Haven, CT), and D. H. Whalen (Speech-Language-Hearing Sci., City Univ. of New York Graduate Ctr., New York, NY)

Older adults have been reported to have a greater propensity towards speech errors [MacKay & James, *Psych & Aging*, *19*(1), 93–107 (2004)],but as with many age-related effects, there is considerable variability in results and a need to identify behavioral predictors. The present study employs a motor speech paradigm, whereby both younger and older adults repeated word pairs (e.g., "top-cop") timed to a metronome in an accelerating rate task. The metronome had a steady rate for half the task (8s) and then increased at a linear rate. Participants also completed evaluations of motor speech, sensory, and cognitive abilities, so that these individual characteristics can be correlated with speech error tendencies. Articulatory data of speech movements were collected in the form of ultrasound tongue contours and lip aperture derived from facial video. Articulatory results are discussed in terms of speech error gradience in the context of previous work [Pouplier, *Proc. XVth ICPhS,* 2245–2248 (2003)], with emphasis on how well speakers adapt to the changing metronome rate, and how this affects error incidence in younger versus older adults.

4aSC27. Reduced coarticulation and aging. Cécile Fougeron, Daria D'Alessandro, and Leonardo Lancia (Laboratoire de Phonétique et Phonologie, CNRS-U. Sorbonne Nouvelle, 19 rue des Bernardins, Paris 75005, France, cecile.fougeron@univ-paris3.fr)

Lifespan changes in speech have mostly been documented with respects to children's development, while little is known about its evolution throughout adulthood. More particularly our knowledge on the effect of aging on speech and voice is sparse. Changes have been found in voice quality [e.g., Xue & Hao, 2003], the speech rate has been described to slow down [e.g., Staiger *et al.* 2017), and pitch is said to raise for older males and to lower for older females [e.g., Harnsberger *et al*., 2008]. The present study aims to investigate the effect of aging on Vowel$_{\text{to_Vowel}}$ anticipatory coarticulation in French. This effect is tested according to vowel duration and to differences in regional varieties. Data from 240 speakers (half female) distributed across three age groups (20–45), (50–69), and (70–80) have been extracted from the MonPaGe$_{\text{HA}}$ database [Fougeron *et al.* 2018] which includes speakers from four regional varieties: France, Belgium, Switzerland, and Quebec. The influence of V2 (/a/ or /i/) on V1 (/a/) is measured as a lowering of F1 and a rise of F2. Results show that coarticulation and vowel duration varies with regional variety and that vowels of older speakers are lengthened. More interestingly, results show a reduction of coarticulation with age.

4aSC28. Improving the quality of speech in the conditions of noise and interference. Bozena Kostek (Audio Acoust. Lab., Gdansk Univ. of Technol., Faculty of ETI, Narutowicza 11/12, Gdansk 80-233, Poland, bokostek@audioakustyka.org) and Krzysztof Kąkol (PGS Software SA, Gdansk, Poland)

The aim of the work is to present a method of intelligent modification of the speech signal with speech features expressed in noise, based on the Lombard effect. The recordings utilized sets of words and sentences as well as disturbing signals, i.e., pink noise and the so-called babble speech. Noise signal, calibrated to various levels at the speaker's ears, was played over two loudspeakers located 2 m away from the speaker. In addition, the recording session included utterances in quiet, which constitute a reference to the received speech signal analysis with the Lombard effect. As a part of the analysis, the following parameters were examined with regard to prosody: fundamental frequency F0, formant frequencies of F1 and F2, duration of the utterance, sound intensity, etc., taking into account individual sentences, words, and vowels. The PRAAT program was used to process and analyze speech signals. Next, a method for modifying speech with the features of speech spoken in noise was proposed. Subsequent analyzes have shown that noisy speech modified by the Lombard effect features is characterized by higher values of the PESQ (perceptual evaluation of speech quality) speech quality indicator compared to noisy speech without the features incorporated.

4aSC29. How much formant variability in linear prediction-based measurements can be explained by F0 variability? Wei-Rong Chen, D. H. Whalen, and Christine H. Shadle (Haskins Labs., 300 George St. STE900, New Haven, CT 06511, chenw@haskins.yale.edu)

Previous studies have used speech variability as a measure of speech development; for instance, children reduce the variability in their formant frequencies as they grow, indicating increases in speech motor control. However, formant measurements in most of these studies are computed using variants of linear prediction coding (LPC), which is known to exhibit a bias towards the nearest harmonic of F0, especially at high F0 [Shadle *et al.* (2016). JASA. 139(2), 713–727]. The proportion of the reported formant variabilities that can be attributed to changes in F0 is still unknown; thus, the true variability in formant frequency is also unknown. Therefore, we used published values of formant and F0 variabilities for /a/ produced by children (age range: 4~19 + yrs), and, for each age group, we synthesized 1000 vowels with the same formants (F1~F3) fixed at the reported group means, but randomly varied F0 using a normal distribution (μ = F0 group mean, σ = F0 standard deviation (SD)). We then measured formants in those synthesized vowels using LPC; the SDs of the measured formants were considered to be the F0-biased formant variability (true formant SD = 0). A ratio of F0-biased SD to reported formant SD for each age group indicates how much of the reported formant variability can be explained by F0 changing in children's speech. The results show, on average, that the purported formant variabilities in previous studies that can be explained by F0 bias are 82.2 % for F1, 31.1% for F2, and 17.4% for F3.

4aSC30. Pitch duration as a cue for declination. Lihan Wu (School of Foreign Lang. Studies, Guangxi Univ. for Nationalities, Nanning, Guangxi 530006, China, linhuadance@hotmail.com) and Hua Lin (Linguist, Univ. of Victoria, Victoria, BC, Canada)

An utterance of a language often demonstrates the effect of declination, usually understood broadly as a reduction in certain physical or acoustic signals. Previous research on declination focuses primarily on the acoustic measures of pitch (such as the movement of pitch or the alternation of pitch span) or intensity. Neglected in the matter is the third acoustic dimension of speech duration. This paper reports on an experiment on declination focusing on *pitch duration*. The language studied is Mandarin Chinese. Six native Mandarin speakers are recruited. A total of 864 utterances of 2-to-9 syllables in four tones and four functional intonations are recorded and analyzed on Praat. The results show that the declination of pitch duration goes side by side along the pitch declination, both of which share the same physiological basis. Specifically, (1) the average pitch duration within the prosodic unit decreases gradually from the sentence-initial prosodic unit to the sentence-final one, (2) the pitch duration of the left-most syllable of the prosodic unit decreases gradually from the sentence-initial prosodic unit to the sentence-final one, and (3) the pitch duration of the right-most syllable of the prosodic unit decreases slightly from the sentence-initial prosodic unit to the sentence-final one.

4aSC31. Fully-automated tongue detection in ultrasound images. Elham Karimi (Dept. of Elec. Eng., École de technologie supérieure, 1100 Notre Dame Ouest, Bureau A-2464, Montreal, QC H3C 1K3, Canada, elham.karimi.1@etsmtl.net), Lucie Menard (Dept. of Linguist, Université du Québec à Montréal, Montreal, QC, Canada), and Catherine Laporte (Dept. of Elec. Eng., École de technologie supérieure, Montreal, QC, Canada)

Tracking the tongue in ultrasound images provides information about its shape and kinematics during speech. Current methods for detecting/tracking the tongue require manual initialization or training using large amounts of labeled images. This work introduces a new method for extracting tongue contours in ultrasound images that requires no training or manual intervention. The method consists in: (1) application of a phase symmetry filter to

highlight regions possibly containing the tongue contour; (2) adaptive thresholding and rank ordering of grayscale intensities to select regions that include or are near the tongue contour; (3) skeletonization of these regions to extract a curve close to the tongue contour; and (4) initialization of an accurate active contour from this curve. Two quality measures were also developed that predict the reliability of the method so that optimal frames can be chosen to confidently initialize fully automated tongue tracking. Experiments were run on 16 free speech ultrasound recordings from healthy subjects and subjects with articulatory impairments due to Steinert's disease. Fully automated and semi automated methods result in mean sum of distances errors of 0.92mm +/- 0.46 mm and 1.04 mm +/ 0.5513 mm, respectively, showing that the proposed automatic initialization does not significantly alter accuracy.

4aSC32. Speech adaptation to palatal perturbation: Evidence for sensorimotor reorganization across the workspace. Douglas Shiller, Guillaume Barbier (School of Speech-Lang. Pathol. & Audiol., Université de Montréal, PO Box 6128, succursale Centre-ville, Montreal, QC H3C 3J7, Canada, douglas.shiller@umontreal.ca), Lucie Menard (Linguist, Université du Québec à Montréal, Montreal, QC, Canada), and Shari Baum (School of Commun. Sci. & Disord., McGill Univ., Montreal, QC, Canada)

Competent speakers demonstrate a high degree of precision in speech production, but also considerable flexibility in speech motor patterns used to attain speech goals. This flexibility is evident in studies examining adaptation to sensory perturbations, including physical manipulations that alter both auditory and somatosensory feedback (e.g., using a palatal prosthesis to disrupt the production of the alveolar fricative /s/). While the acoustic effects of such perturbations (and subsequent adaptation) have been well explored, their underlying articulatory basis remains poorly understood. In this study, we explored speech adaptation to a palatal perturbation in 9 adult speakers using acoustic and articulatory measures of adaptation (using electromagnetic articulography) during the production of various consonants and vowels. Acoustic measures showed effects of both the perturbation and subsequent compensation (following 15 minutes of speech practice) across the consonants and vowels. Analysis of tongue kinematics confirmed the global nature of these effects, including significant changes in sounds involving no physical interaction with the palate. Furthermore, directional analyses revealed a degree of fine-tuning in motor adaptation across phonemes. The findings indicate that even localized changes in palate shape induce complex compensatory changes in speech motor control across the articulatory workspace.

4aSC33. Cepstral coefficients successfully distinguish the front Greek fricatives. Laura Spinu (Communications & Performing Arts, CUNY - Kingsborough Community College, 2001 Oriental Boulevard, Brooklyn, NY 11235-2398, Laura.Spinu@kbcc.cuny.edu), Jason Lilley (Ctr. for Pediatric Auditory & Speech Sci., Nemours Biomedical Res., Wilmington, DE), and Angeliki Athanasopoulou (School of Lang., Linguist, Literatures, and Cultures, Univ. of Calgary, Calgary, AB, Canada)

In the current study, we explore the factors underlying the well-known difficulty in acoustic classification of front fricatives (McMurray & Jongman, 2011; Maniwa *et al*., 2009) by taking a closer look at the production of 29 native Greek speakers. Our corpus consists of Greek fricatives from five places of articulation and two voicing values [f, v, θ, ð, s, z, ç, ʝ, x, ɣ] produced in nonce disyllabic words before [a, o, u] in stressed syllables (e.g., [ˈθakos]). We apply a relatively novel classification method based on cepstral coefficients, previously successful with obstruent bursts (Bunnell *et al*., 2004), vowels (Ferragne & Pellegrino, 2010), and Romanian fricatives (Spinu & Lilley, 2016). Our method yields the best correct classification rates reported to date for front fricatives: Present study: 88%; English: 66% (Jongman *et al.,* 2000), Greek: 55.1% (Nirgianaki, 2014). The important cues for the successful classification are the vowel following the target fricative and the second region of frication noise. Our study adds to the body of work aimed at identifying techniques for quantifying and categorizing large samples of speech. Obtaining higher classification rates than before takes us one step closer to understanding the properties of "difficult" sounds like the front fricatives.

4aSC34. Nasal rustle: An evidence-based description, II. Michael Rollins (Biomedical Eng., Univ. of Cincinnati, MSB 6303, 231 Albert Sabin Way, Cincinnati, OH 45267, rollinmk@mail.uc.edu), Liran Oren (Otolaryngol., Univ. of Cincinnati, Cincinnati, OH), Srujana Padakanti, Ephraim Gutmark (Aerosp. Eng., Univ. of Cincinnati, Cincinnati, OH), Ann W. Kummer (Speech-Lang. Pathol., Cincinnati Children's Hospital Medical Ctr., Cincinnati, OH), and Suzanne E. Boyce (Commun. Sci. & Disord., Univ. of Cincinnati, Cincinnati, OH)

This research extends a previous study investigating the sound source of audible nasal emissions associated with a small velopharyngeal (VP) opening (herein "nasal rustle," but also "nasal turbulence" elsewhere). This extension includes more speech sounds than the original study, which only considered /s/. Speech sounds susceptible to nasal rustle were recorded from ten pediatric patients with VP insufficiency via high-speed nasopharyngoscopic video at the superior VP port and simultaneous nasometry. The total pixel intensity was summed in each video frame to generate a video intensity signal. For each speech sound token, a cross-correlation was computed between this video intensity signal and each of the nasometer's acoustic signals; this measures similarity in frequency content between motion at the superior VP port and acoustics exterior to the lips and nares, respectively. The difference between the video-nasal cross-correlation and the video-oral cross-correlation over a frequency bandwidth including the dominant frequencies observed in the video intensity signal (<100 Hz) has a positive association with the number of mucus movements, meaning the motion of mucous secretions correlates more strongly with acoustic energy exterior to the nares than exterior to the mouth. This indicates that mucus motion could be a dominant sound source of nasal rustle.

4aSC35. Contribution of the tongue tip retraction in the articulation of high vowels. Hayeun Jang (Univ. of Southern California, 3601 Watt Way, Grace Ford Salvatori 301, Los Angeles, CA 90089-1693, hayeunja@usc.edu)

This study shows that the retraction of the tongue tip contributes to the raising of the tongue body in the articulation of high vowels /i/ and /u/ by using qualitative simulations of tongue deformation by using a biomechanical 3D tongue model of Artisynth (Lloyd *et al*. 2012) and the analysis of tongue configuration in X-ray microbeam data (Westbury 1994). The simulations qualitatively replicated the tongue shapes at the temporal mid-point of /i/ and /u/ in rtMRI data by manipulating the activation values of tongue muscles in Artisynth. The results show that the tongue tip retraction helps to raise the tongue body higher in both /i/ and /u/. In the simulations of /i/, the tongue tip retraction raises the tongue body even without moving the tongue body. The results of x-ray microbeam analysis confirm those findings of simulations. The mixed linear regression model of tongue configurations in the tasks of sentence and paragraph production shows that more tongue tip retraction (a shorter horizontal distance between two anterior tongue pellets) significantly correlates with a higher position of the tongue body (higher one between two posterior tongue pellets) in both /i/ and /u/ and the correlation is stronger in /i/ than in /u/.

4aSC36. Limitations of the source-filter coupling in phonation. Debasish R. Mohapatra and Sid Fels (Dept. of Elec. and Comput. Eng., Univ. of Br. Columbia, 2366 Main Mall, Vancouver, BC V6T 1Z4, Canada, d.mohapatra@alumni.ubc.ca)

The coupling of vocal fold (source) and vocal tract (filter) is one of the most critical factors in source-filter articulation theory. The traditional linear source-filter theory has been challenged by current research which clearly shows the impact of acoustic loading on the dynamic behavior of the vocal fold vibration as well as the variations in the glottal flow pulses' shape. This paper outlines the underlying mechanism of source-filter interactions; demonstrates the design and working principles of coupling for the various existing vocal cord and vocal tract biomechanical models. For our study, we have considered self-oscillating lumped-element models of the acoustic source and computational models of the vocal tract as articulators. To

understand the limitations of source-filter interactions which are associated with each of those models, we compare them concerning their mechanical design, acoustic and physiological characteristics and aerodynamic simulation. References: 1. Flanagan, J. L. (**1968**) "Source-system interaction in the vocal tract," Ann. N.Y. Acad. Sci. **155**, 9–17. 2. Lieberman, P., and Blumstein, S.E. (**1988**) Speech physiology, speech perception, and acoustic phonetics (Cambridge University Press, Cambridge, Mass). 3. Titze, I. R. (**2008**) "Nonlinear source-filter coupling in phonation: Theory," *J. Acoust. Soc. Am.* **123,** 2733–2749.

4aSC37. Articulatory feature extraction from ultrasound images using pretrained convolutional neural networks. Kele Xu (School of Comput., National Univ. of Defense Technol., 16 Rue Flatters, Paris 75005, France, kelele.xu@gmail.com) and Jian Zhu (Dept. of Linguist, Univ. of Michigan, Ann Arbor, MI)

Feature extraction is of great importance to ultrasound tongue image analysis. Inspired by the recent success of deep learning, we explore a novel approach to feature extraction from ultrasound tongue images using pretrained convolutional neural networks (CNN). The bottleneck features from different pre-trained CNNs, including VGGNet and ResNet, are used as representations of the ultrasound tongue images. Then an image classification task is conducted to assess the effectiveness of CNN-based features. Our dataset consists of 20,000 ultrasound tongue images collected from a female speaker of Mandarin Chinese, which were manually labeled as containing one of the following consonants: /p, t, k, l/. Experiment results show that the Gradient Boost Machines (GBM) classifiers trained on the CNN-based features achieve the best performance, with a classification accuracy of 92.4% for ResNet and 91.6% for VGGNet, outperforming the benchmark GBM classifier trained on the features extracted using Principal Component Analysis (PCA), which only achieves an accuracy of 87.5%. In this preliminary dataset, our method of feature extraction is found to be superior to the PCA-based method. This work demonstrates the potential of applying the pre-trained convolutional neural networks to ultrasound tongue image analysis task.

4aSC38. Hand gesture density in prosodically important speech regions. Samantha G. Danner (Linguist, Univ. of Southern California, 3601 Watt Way, Grace Ford Salvatori 301, Los Angeles, CA 90089, sfgordon@usc.edu)

Hand movements during speech are systematically linked with a variety of "landmarks" in the acoustic speech signal, including prosodically important regions in speech. While many researchers have studied qualitative aspects of how hand movement coordinates with speech, there are few studies that consider quantitative properties of manual gestures during speech and their coordination with speech prosody. We use optical flow to automatically detect manual gesture and identify velocity peaks in the right hand movement signal. Hand gestures for six speakers are z-scored and classified into three "sizes" as a function of velocity peak magnitude. We measure whether manual gesture density—i.e., the occurrence frequency of velocity peaks—differs among regions around prosodic landmarks in the accompanying acoustic speech signal, including at speech turns, phrase boundaries, acoustic amplitude prominences, and regions containing both a pitch accent and an amplitude prominence, with respect to a baseline of all other regions of speech. Results show that manual gestures of all sizes are more frequent at phrase boundaries and at acoustically and prosodically prominent regions of speech than at turn boundaries and non-prominent speech regions. This suggests that gesture density may be a communicative component highlighting prosodically important regions of speech. [Work supported by NIH.]

4aSC39. Articulatory and acoustic investigations into gestures of Mandarin retroflex fricatives . Yung-hsiang Shawn Chang (Dept. of English, National Taipei Univ. of Technol., Zhongxiao E. Rd., Sec. 3, No. 1, Taipei 106, Taiwan, shawnchang@mail.ntut.edu.tw)

This study investigates the inventory and acoustic realizations of articulatory gestures of the Mandarin retroflex (or arguably, post-alveolar) fricative /ʂ/. Tongue shapes were obtained with ultrasound images and categorized as retroflexed or bunched, in light of the literature on the North American English rhotic /ɹ/ production. Velarization, a secondary articulatory gesture of retroflex consonants, was also coded in the ultrasound imaging analysis. Lip rounding, found to be an optional enhancing gesture in Mandarin retroflexion, was identified with lip video data. The acoustic measurements included the spectral center of gravity obtained from the middle of the frication and F2 frequency measured at the onset of the vowel following the fricative. The analyses showed that the spectral center of gravity only varied as a function of the vowel context. F2 at vowel onset was affected by vowel context, velarization and the interaction of vowel context and lip rounding. The preliminary results failed to support previous speculations that the two tongue shapes in Mandarin retroflexes result in different acoustics. We consider including more acoustic measures to further explore the acoustic consequences of articulatory variations in Mandarin retroflex fricatives.

4aSC40. A mechanical deformation model of the tongue during speech production considering personalized anatomical structure. Shun Takenaka, Tsukasa Yoshinaga (Graduate School of Eng. Sci., Osaka Univ., 1-3, Toyonaka, Osaka 560-8531, Japan, s.takenaka@me.es.osaka-u.ac.jp), Kazunori Nozaki (Dental Hospital, Osaka Univ., Suita, Japan), Satoshi Ii (Graduate School of Systems Design, Tokyo Metropolitan Univ., Hachioji, Japan), and Shigeo Wada (Graduate School of Eng. Sci., Osaka Univ., Toyonaka, Japan)

In the previous studies, several tongue models have been developed to investigate the relationship between the contraction of tongue muscles and the deformation of the tongue during speech. However, the tongue deformation is highly subject-specific and their modellings are insufficient to uncover the detailed mechanism. In this study, a mechanical tongue model considering personalized anatomical structure of tongue myofiber is proposed. A set of reference geometries of tongue and jaw is constructed from the medical images of computer tomography (CT) and the magnetic resonance imaging (MRI) for a Japanese male adult. In addition, we reflect the information of tongue myofiber orientations evaluated from the medical images of diffusion tensor imaging (DTI) on the model using a smoothing extrapolation technique. The tongue deformation is calculated by solving a force-equilibrium equation using a continuous Galerkin finite element method with a hyperelastic material, where the deformation is driven by active contraction of the tongue muscles. We examined the tongue deformations under several contraction conditions during speech and confirmed reasonable numerical reproductions of actual tongue motion and shape. These results indicate that the proposed model reflecting personalized information including myofiber orientations has a potential to identify the tongue muscle contractions in speech production.

4aSC41. SOUND STREAM: Towards vocal sound synthesis via dual-handed simultaneous control of articulatory parameters. Pramit Saha, Debasish R. Mohapatra, Venkata Praneeth Srungarapu, and Sid Fels (Dept. of Elec. and Comput. Eng., Univ. of Br. Columbia, 2366 Main Mall, Vancouver, BC V6T 1Z4, Canada, pramit@ece.ubc.ca)

This paper introduces Sound stream: a low-cost, tangible and ambidextrous controller which drives a dynamic muscle-based model of the human vocal tract for articulatory speech synthesis. The controller facilitates the multidimensional inputs which are mapped to the tongue muscles in a biomechanical modeling toolkit Artisynth using a microcontroller. As the vocal tract is a complex biological structure containing many muscles, it is a challenging and computationally expensive task to accommodate control for every muscle in the proposed scheme. So, we have followed a simplified approach by controlling the selective muscles for the efficient articulatory speech synthesis. The goal for designing an ambidextrous controller is to create new possibilities of controlling multiple parameters to vary the tongue position and shape simultaneously for generating various expressive vocal sounds. As a demonstration, the user learns to interact and control a mid-sagittal view of the tongue structure in Artisynth through a set of sensors using both hands. The Sound-Stream explores and evaluates the appropriate input and mapping methods to design a controllable speech synthesis engine. 1. Wang, J. *et al.* (**2011**) "Squeezy: Extending a multi-touch screen with force sensing objects for controlling articulatory synthesis," in Proceedings on New Interfaces for Musical Expression, Oslo, Norway, pp. 531–532.

4aSC42. Co-mingling effects of dialectal and stylistic variation due to choice of speaker normalization. Wil A. Rankinen (Commun. Sci. and Disord., Grand Valley State Univ., 515 Michigan St. NE, Ste. 300, Office 309, Grand Rapids, MI 49503, wil.rankinen@gvsu.edu) and Kenneth de Jong (Linguist, Indiana Univ., Bloomington, IN)

This paper explores how a speaker normalization routine can result in co-mingling effects of dialectal and stylistic variation. To this end, the paper examines a database of vowel formants in a passage and a wordlist task collected from 88 monolingual American English speakers in Michigan's Upper Peninsula; this corpus is further stratified by ethnic-heritage, age, sex and educational attainment. Two different speaker normalization routines, Lobanov (Adank *et al.*, 2004) and Labov ANAE (Labov *et al.*, 2006), were used; however, each routine yielded differences in a paralleled analysis. The Lobanov normalization routine removes much of the ethnic-heritage effect that appears in the analysis using the Labov ANAE normalization routine. Examining the cause of these different results reveals that 1) the Lobanov routine removes any mean shifts in formant values that run orthogonal to anatomical variation (i.e., vocal tract length effects) and 2) the ANAE routine only removes shifts which are correlated across formants. The ethnic-heritage effect in this dataset involves mean shifts that are negatively correlated to typical anatomical variation, which results in the ANAE routine potentially amplifying such effects. We discuss the import of these findings for vowel space variation and language change.

4aSC43. Reconstruction of the glottal pulse using a subband technique on kazoo recordings. Alexandre M. Lucena, Mario Minami (CECS, Federal Univ. of ABC, Av. dos Estados, 5001, Santo André, São Paulo 09210-580, Brazil, ale.mlucena@gmail.com), and Miguel A. Ramirez (Electron. Systems, Univ. of São Paulo, São Paulo, São Paulo, Brazil)

The kazoo, a wind instrument, generates its typical sound when stimulated by voiced speech. Using this instrument, this paper proposes a novel technique to recover the glottal pulse excitation of its player. We applied multiband frequency techniques to the kazoo signal to compare the results with those obtained from the corresponding recordings of an electroglottograph (EGG). With the player's management over his embouchure on the instrument, one can make recordings for spoken and singing speech as well as recitative, at the instrument's resonator cap, which closely fit the EGG recordings. After a spectrogram analysis, it was possible to detect in the lower frequency band of the kazoo signals, the spectral envelope and, in the higher frequency band, the pitch harmonics mixed with the spectral decay of the glottal pulse. A quadrature mirror filter (QMF) was designed, providing this source-filter separation. Additionally, a reverse spectral band replication (SBR) technique was applied, which consists in recovering the lower frequency band by the demodulation of the higher frequency band followed by a total energy spectral gain adjustment, where a new signal was generated and then evaluated. At the end, a subjective evaluation, SNR, and SD measures prove the efficiency of the proposed method.

4aSC44. Toward the automatic detection of manually labeled irregular pitch periods. Olivia Murton (Speech and Hearing BioSci. and Technol., Harvard Med. School, One Bowdoin Square, 11th Fl., Boston, MA 02114, omurton@g.harvard.edu), Sophie Rosas-Smith (Comput. Sci., Wellesley College, Wellesley, MA), Jeung-Yoon Choi (Res. Lab. of Electronics, Massachusetts Inst. of Technol., Cambridge, MA), Daryush Mehta (Ctr. for Laryngeal Surgery and Voice Rehabilitation, Massachusetts General Hospital, Boston, MA), and Stefanie Shattuck-Hufnagel (Res. Lab. of Electronics, Massachusetts Inst. of Technol., Cambridge, MA)

Irregular pitch periods (IPPs) occur in a wide variety of speech contexts and can support automatic speech recognition (ASR) systems by signaling word boundaries, phrase endings, and certain prosodic contours. IPPs can also provide information about emotional content, dialect, and speaker identity. The ability to automatically detect IPPs is particularly useful because accurately identifying IPPs by hand is time-consuming and requires expertise. In this project, we use an algorithm developed for creaky voice analysis by Kane *et al.* (2013) incorporating features from Ishi *et al.* (2008) to automatically identify IPPs in recordings of speech from the American English Map Task (AEMT) database. Short-term power, intra-frame periodicity, inter-pulse similarity, subharmonic energy, and glottal pulse peakiness measures are input into an artificial neural network to generate frame-by-frame creak probabilities. To determine a perceptually relevant threshold probability, the creak probabilities are compared to IPPs hand-labeled by experienced raters. Preliminary results from four AEMT conversations yielded an area under the receiver operating characteristic curve of 0.898 on average. A creak probability threshold of 0.0083 yielded equivalent sensitivity and specificity of 81.1%. This study indicates generally good automatic detection of hand-labeled IPPs and explores the effects of linguistic and prosodic context.

4aSC45. Belching as a non-lexical speech object: Evidence from pop-culture. Brooke L. Kidner (Linguist, Univ. of Southern California, 620 McCarthy Way, #273, Los Angeles, CA 90089, bkidner@usc.edu)

Belching is normally considered an involuntary speech-tract vocalization. When speaking, involuntary speech-tract gestures such as belching often compete with speech for articulation by our vocal apparatus. There are non-lexical paralinguistic items (i.e., "ugh") that also convey speech meaning, but traditionally belching is not regarded as a speech object. This study presents a case where belching appears to be an intentional speech act, occurring with clearly defined parameters and not in competition with speech. Data comes from the character Rick from TV's Rick and Morty. Transcripts were collected and compared with the audio data. 105 belches were found in the 8 episodes investigated: all belches, except 1, were not written into the original scripts or transcriptions. Acoustic analysis showed no significant disruptions between belching and speech, supporting the conclusion that belches do not compete with speech. The most frequent occurrence of belching is word-medial, after the initial segment, which is a common infixation pattern cross-linguistically. Additional acoustic analysis showed belching occurred regularly before a stressed syllable, an established pattern for infixation in English. This pattern provides evidence that these belches are behaving phonologically as para-phonemic items. The results show that belching can be intentional and behave like paralinguistic items in human language and communication.

4aSC46. Accidental gaps in Mandarin Chinese tones. Shao-Jie Jin and Yu-an Lu (Foreign Lang. and Literatures, National Chiao Tung Univ., DFLL Humanities Bldg. 3, 1001 University Rd., Hsinchu 30010, Taiwan, courtney0419003.fl04@nctu.edu.tw)

Mandarin is a tone language with four phonemic tones (i.e., high-level Tone 1 [55]), rising Tone 2 [35], falling-rising Tone 3 [214], and falling Tone 4 [51] and with the maximum (C)(G)V(G)/(C) syllable structure. However, not all syllables can be combined with each of the tones (i.e., accidental gaps). For example, the syllable [ts^{h}u] is allowed to be combined with T1 ([ts^{h}u]55 "coarse"), T4 ([ts^{h}u]51 "vinegar"), but not T2 and T3. A calculation of all the 391 allowable syllables showed that there are 131, 185, 155, 110 accidental gaps in each of the four tones. A one-way chi-square test revealed that the accidental gaps in T2 were over-represented (stats here). A further investigation into these gaps in T2 according to different syllable types (CV: 52, CVN: 43, CGV: 38, and CGVN: 29) showed that these gaps were marginally under-represented in CGVN (stats here). We attributed these findings partially to the marked status of contour tones and to the typological preference for complex tonal targets to complex rimes due to their inherent longer durations (cf. Zhang 2001).

Session 4aSP

Signal Processing in Acoustics, Underwater Acoustics, Engineering Acoustics, and Physical Acoustics: Detection and Tracking of Mobile Targets I

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Kainam T. Wong, Cochair
Dept. of Electronic & Information Engineering, Hong Kong Polytechnic University, 11 Yuk Choi Road, Hung Hom KLN, Hong Kong

Invited Papers

8:00

4aSP1. Detection and tracking in very shallow water. William Jobst, Lawrence Whited (Appl. Marine Phys., LLC, 1544 Marina Dr., Slidell, LA 70458, amp_jobst@bellsouth.net), and David Smith (Appl. Marine Phys., LLC, Miami, FL)

In our previous work, we demonstrated a low power, m-sequence based, bistatic acoustic system that detected and tracked a -20 dB target at a range of 172 m in a water depth of 2 m. To our knowledge, similar results have not been obtained with non-biological systems. Unfortunately, we were not able to repeat these results under similar, or possibly better, environmental conditions. This work addresses the reasons for failure to reproduce our original experimental results and suggests an approach that should result in very shallow water detection and tracking at much longer range.

8:20

4aSP2. Simultaneous estimation of array tilt and source range using broadband ship noise and a vertical array. Hee-Chun Song (Scripps Inst. of Oceanogr., 9500 Gilman Dr., La Jolla, CA 92093-0238, hcsong@mpl.ucsd.edu), Gihoon Byun (Scripps Inst. of Oceanogr., San Diego, CA), and Jea Soo kim (Korea Maritime and Ocean Univ., BUSAN, South Korea)

The array invariant, a robust approach to source-range estimation in shallow water, is based on the dispersion characteristics of broadband signals in ideal waveguides. It involves time-domain planewave beamforming using a vertical line array (VLA) to separate multiple coherent arrivals (eigenrays) in beam angle and travel time. Typically, a probe signal (i.e., a cooperating source) is required to estimate the Green's function, but the array invariant has been recently extended to a ship of opportunity radiating random signals using a ray-based blind deconvolution. Still, one major drawback is its sensitivity to the array tilt, shifting the beam angles and adversely affecting the array invariant parameter that determines the source range. In this paper, a simple optimization algorithm for simultaneous estimation of the array tilt and the source range is presented. The method is applied to a ship of opportunity (200–900 Hz) circling around a 56-m long VLA at a speed of 3 knots (1.5 m/s) at ranges of 1.8-3.6 km in approximately 100-m deep shallow water. It is found that the standard deviation of the relative range error significantly reduces to about 4%, from 14% with no compensation of the array tilt.

8:40

4aSP3. Tracking a surface ship via cascade of blind deconvolution and array invariant using a bottom-mounted horizontal array. Gihoon Byun, H. C. Song (Scripps Inst. of Oceanogr., La Jolla, San Diego, CA 92093-0238, gbyun@ucsd.edu), Jea Soo Kim (Korea Maritime and Ocean Univ., BUSAN, South Korea), and Ji Sung Park (Korea Inst. of Ocean Sci. and Technol., Busan, South Korea)

The array invariant developed for robust source-range estimation typically requires a priori knowledge about the source signal (i.e., cooperating source) to estimate the Green's function. However, the array invariant method has been recently extended to a surface ship radiating random signals (200-900 Hz) by extracting the Green's function via blind deconvolution using a vertical array [*J. Acoust. Soc. Am.* **143**. 1318–1325 (2018)]. In this paper, the blind deconvolution is applied to a 60-m long, bottom-mounted horizontal array to extract the Green's function of the same ship circling around the array in a square spiral pattern at ranges of 300–1500 m. The overall tracking performance shows good agreement with GPS measurements except when the ship is towards the broadside with respect to the horizontal array. Further, simultaneous localization of multiple ships is discussed.

Contributed Paper

9:00

4aSP4. Monitoring of moving surface ship using a four-element planar hydrophone array installed on the seabed. Sung-Hoon Byun and Sea-Moon Kim (KRISO, 32 1312beon-gil, Yuseong-daero, Yuseong-gu, Daejeon 34103, South Korea, byunsh@kriso.re.kr)

This talk presents the analysis result of underwater sound measurement data which were recently gathered in the shallow water near South Korea. For the underwater sound measurement, a four-element planar hydrophone array developed by KRISO was installed on the seabed and recorded a month-long acoustic data. The result shows that the measured sound level is usually higher in the morning time and this is ascribed to the surface boats which are presumably in fishing activity. We deal with the identification of the sound sources via various modulation analysis methods which can discriminate the sound induced by rotating propellers and also discuss the direction finding by beamforming of the planar hydrophone array data. [This work was financially supported by the research project PNS3120 funded by KIGAM.]

Invited Paper

9:15

4aSP5. Comparing passive localization methods for ocean vehicles from their underwater sound received on a coherent hydrophone array. Chenyang Zhu (Northeastern Univ., 360 Huntington Ave., Boston, MA 02115, zhu.che@husky.neu.edu), Delin Wang (Mech. Eng., Massachusetts Inst. of Technol., Boston, MA), Alessandra Tesei (Eng. Dept., Ctr. for Maritime Res. and Experimentation (CMRE), La Spezia, Italy), and Purnima Ratilal (Northeastern Univ., Boston, MA)

Simultaneous localization and tracking of multiple ocean vehicles over instantaneous continental-shelf scale regions using the passive ocean acoustic waveguide remote sensing (POAWRS) technique employs a large-aperture densely-sampled coherent hydrophone array system to monitor the underwater sounds radiated by ocean vehicles. Here, particle filtering is implemented for bearings-only localization and tracking of surface ships where the estimated bearing-time trajectories of ship-radiated underwater sound detections are used as inputs to provide two-dimensional horizontal position estimates. Coherent beamforming of the acoustic data received on the hydrophone array not only provides estimates of ship bearing, but also significantly enhances the signal-to-noise ratio. Results of passive acoustic surface ship localization and tracking from recordings in the Gulf of Maine and the Norwegian Sea are presented. The particle filtering approach for passive source localization are compared with three other approaches: moving array triangulation (MAT), array invariant (AI), and modified-polar-coordinates extended Kalman filter (MPC-EKF). The passive source localization accuracies, determined by comparison with the GPS-measured position of the surface ships, are dependent on numerous factors such as source-receiver geometry, range, and relative speed. By combining several of these approaches, which have respective pros and cons under different circumstances, the surface ships can be localized with improved accuracy.

Contributed Paper

9:35

4aSP6. Application of preprocessing methods to improve time delay estimation for synthetic aperture sonar motion estimation. Julia Gazagnaire (NSWC PCD, 2525 Pelican Bay Dr., Panama City Beach, FL 32408, jgazagnaire@hotmail.com) and Pierre-Philippe Beaujean (Ocean and Mech. Eng., Florida Atlantic Univ., Dania Beach, FL)

Synthetic aperture sonar (SAS) provides the best opportunity for side-looking sonar mounted on unmanned underwater vehicles to achieve high-resolution images at longer ranges. However, SAS processing requires maintaining a coherent phase history over the entire synthetic aperture, driving strict constraints on resolvable platform motion. This has driven the development of motion estimation and compensation techniques that use the received ping data, in addition to the onboard navigation solution, to resolve ping-to-ping platform motion. The most common approach is to use the redundant phase center technique. Here the ping interval is set, such that a portion of the array is overlapped. The accuracy of the motion estimation depends on the accuracy of the time delay estimation between the data received on the overlapping channels. Given the stochastic nature of the operational environment some level of decorrelation between these two signals is likely, even without residual platform motion. This decorrelation results in inaccurate time delay estimation and image quality degradation. In this research various preprocessing techniques have been applied to the sonar data to reduce the influence of stochastic noise with the goal of improving the accuracy of the time delay estimates.

Invited Papers

9:50

4aSP7. Mobile targets and sensor mobility in bat biosonar. Rolf Müller (Mech. Eng., Virginia Tech, ICTAS Life Sci. District (Mail Code 0917), Virginia Tech, Blacksburg, VA 24061, rolf.mueller@vt.edu)

Many species of echolocating bats pursue highly mobile prey such as flying insects. Target mobility poses a challenge when a prey executes escape maneuvers. In these cases, bats and their prey find themselves locked in evolutionary arms races. However, prey mobility also provides the bats with opportunities for solving the problem of detecting prey buried in clutter. This applies to active as well as passive biosonar. In passive biosonar, bats frequently listen to sounds that are generated involuntarily by the motion of a prey. In active

biosonar, certain bat groups have evolved highly specialized biosonar systems to detect flying prey insects by virtue of unique Doppler signatures that are generated by the insect's wing beat. Since the Doppler shifts concerned are very small, this sensory feat requires a well-integrated suit of evolutionary specializations that include sonar signal design, cochlear filtering, representations all through the auditory system, and even adaptive behavioral control of the emitted frequencies. In addition to this complexity, bats with such a biosonar system also employ a peripheral mobility where the baffles for pulse emission and reception move rapidly during biosonar operation. It remains to be seen how these mobilities fit together to support the animal's sensory abilities.

10:10–10:25 Break

10:25

4aSP8. Underwater acoustic localization of sperm whales with a pair of hydrophones. Emmanuel Skarsoulis, George Piperakis (Inst. of Appl. and Computational Mathematics, Foundation for Res. and Technol. - Hellas, N. Plastira 100, Heraklion GR-70013, Greece, eskars@iacm.forth.gr), Michael Kalogerakis (Technolog. Education Institute-Crete, Heraklion, Greece), Emmanuel Orfanakis, Panagiotis Papadakis (Inst. of Appl. and Computational Mathematics, Foundation for Res. and Technol. - Hellas, Heraklion, Greece), Stan E. Dosso (School of Earth and Ocean Sci., Univ. of Victoria, Victoria, BC, Canada), and Alexandros Frantzis (Pelagos Cetacean Res. Inst., Vouliagmeni, Greece)

An iterative Bayesian approach using a pair of hydrophones was applied for the localization of vocalizing (click producing) sperm whales off the southern coast of Crete in the Eastern Mediterranean Sea in June 2015. The localization method relies on the time difference between direct and surface reflected arrivals at each hydrophone as well as between the direct arrivals at the two hydrophones, and on the knowledge of the hydrophones depths, whereas it accounts for refraction due to a depth-dependent sound-speed profile. The method provides range and depth estimates as well as estimates for the localization uncertainty reflecting measurement errors and environmental uncertainties. Further, by assuming a simple model for the array geometry bearing estimates were obtained. The inherent left-right ambiguity in bearing estimation can be resolved by changing the array orientation through maneuvering. The localization results were verified through encounters with the sperm whales when they ascended to the surface. [Work supported by EU-Greece through the Aristeia-II program and the NSRF 2014-2020/PERAN project.]

Contributed Paper

10:45

4aSP9. Tracking sperm whales *(Physeter macrocephalus)* detected with irregular time intervals. Tian Bai and Paul White (Inst. of Sound and Vib. Res., Univ. of Southampton, University Rd., Southampton, Hampshire SO171BJ, United Kingdom, t.bai@soton.ac.uk)

Passive acoustic monitoring (PAM) is a method that has been proved to be a powerful tool for monitoring sperm whales. Sperm whales frequently emit loud, short duration, clicks for the purposes of echolocation. The clicks are emitted at intervals of typically once per second, but the exact interval between pulses varies irregularly. To achieve the monitoring goals, it is necessary to localise the vocalising animal. This is typically achieved by detecting the animal's echolocation clicks on a hydrophone array, estimating time delays between clicks and then employing a localisation algorithm. By adopting a tracking-based approach we are able to smooth the results, based on a model of whale motion, to remove noise and improve the estimates of location. The paper will consider the problem of tracking whales using the data from fixed bottom mounted sensors and using a tracking algorithm to follow the motion of such animals. The tracking will be implemented using a Kalman filter, but in this instance that filter needs to be designed to allow for data measured at irregular intervals. The method uses a version of the Kalman filter using a varying time-step. The time-step varying Kalman filter is applied to track data from simulated and measured datasets. The measured dataset is publicly available as part of the 2nd International Workshop on Detection and Localisation of Marine Mammals, Monaco, 2005 and comprises of a single animal moving within an array of 5 sensors.

Invited Paper

11:00

4aSP10. Comparing neural networks with conventional classifiers for fin whale vocalizations in beamformed spectrograms of coherent hydrophone array. Heriberto A. Garcia (Elec. and Comp Eng., Northeastern Univ., 151 Mystic St., Apt. 35, Arlington, MA 02474, garcia.he@husky.neu.edu), Seth Penna (Elec. and Comp Eng., Northeastern Univ., Boston, MA), Jess Topple (Sci. and Technol. Organisation - Ctr. for Maritime Res. and Experimentation (STO CMRE), La Spezia SP, Italy), and Purnima R. Makris (Elec. and Comp Eng., Northeastern Univ., Boston, MA)

A large variety of sound sources in the ocean, including biological, geophysical and man-made activities can be simultaneously monitored over instantaneous continental-shelf scale regions via the passive ocean acoustic waveguide remote sensing (POAWRS) technique by employing a large-aperture densely-sampled coherent hydrophone array. Millions of acoustic signals received on the POAWRS system per day can make it challenging to identify individual sound sources. An automated classification system is necessary to enable sound sources to be recognized. Here a large training data set of fin whale and other vocalizations are gathered after manual inspection and labelling. Next, multiple classifiers including neural networks, logistic regression, support vector machine (SVM) and decision tree are built and tested for identifying the fin whale and other vocalizations from the enormous amounts of acoustic signals detected per day. The neural network classifier will use beamformed spectrograms to classify acoustic signals, while logistic regression, SVM, and decision tree classifiers will use multiple features extracted from each detection to perform classification. The multiple features extracted from each detection include mean, minimum, and maximum frequencies, bandwidth, signal duration, frequency-time slope, and curvature. The performance of the classifiers are evaluated and compared using multiple values including accuracy, precision, recall, and F1-score.

Contributed Paper

11:20

4aSP11. Estimation of Shark's biomass through active acoustics. Roee Diamant (Marine Technologies, Univ. of Haifa, Rm. 280, Multi Purpose Bldg., 199 Aba Khoushy Ave., Haifa 3498838, Israel, roee.d@univ.haifa.ac.il), Eyal Bigal (Marine Biology, Univ. of Haifa, Haifa, Israel), Adi Pinhasi (Marine Technologies, Univ. of Haifa, Haifa, Israel), and Aviad Scheinin (Marine Biology, Univ. of Haifa, Haifa, Israel)

We study sharks biomass in open-sea using non-invasive active acoustics. The importance of continuous long-term monitoring of top-predator biomass is vital in understanding the healthiness of the ecosystem. Instead of the traditional fishery data, catch-and-release methods and visual inspections, which are problematic to supply reliable statistics, we rely on acoustic tools for the quantitative estimation of the number of sharks in a given area, their size, and the evaluation of their motion patterns and behavior. We take a blind classification approach and identify sharks' related reflections from sea boundary reflections based on a track-before-detect approach. Specifically, by emitting a series of wideband acoustic signals, we create a time-delay image whose rows correspond to the received reflection response. We rely on the observation that sea boundary reflections are characterized by a random clutter-like pattern, while shark's related reflections are continuous and steady. Thus, we detect a shark in a clutter by identifying in the time-delay image continuous but curved lines whose structure meet certain limitations, namely, the shark's maximal speed and its expected carangiformmotion pattern. In this paper, we will describe our method in details and show results from a sea experiment that included a verified detected shark.

THURSDAY MORNING, 8 NOVEMBER 2018 OAK BAY 1/2 (VCC), 8:30 A.M. TO 11:45 A.M.

Session 4aUWa

Underwater Acoustics, Acoustical Oceanography, Signal Processing in Acoustics, and Physical Acoustics: Sediment Acoustics–Inferences from Forward Modeling, Direct, and Statistical Inversion Methods I

Charles W. Holland, Cochair
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Chair's Introduction—8:30

Invited Papers

8:35

4aUWa1. Quantifying seabed and other environmental information contained in ocean ambient noise. Martin Siderius (ECE Dept., Portland State Univ., P.O. Box 751, Portland, OR 97207, siderius@pdx.edu) and John Gebbie (Adv. Mathematics Applications, Metron Inc., Portland, OR)

The ocean ambient noise field contains a surprising amount of information about the environment. Information about the seabed includes, layering structure, sound speed, density, and attenuation. There is also information about the ocean environment such as water column sound speed, volume attenuation, and the sea-state. Estimators for these quantities have been developed based mostly on vertical hydrophone arrays and beamforming. Although this has led to some simple and direct estimates for important quantities such as bottom loss, these estimates may be less than ideal in some cases. For example, when using inadequately sampled arrays or in certain unfavorable measurement geometries. The Fisher information can be used to quantify the basic information available in the noise measurements; and its inverse, the Cramér-Rao lower bound (CRLB), provides the lower limit on the variance of any unbiased estimator. The CRLB can be used to study the feasibility of various measurement configurations and parameter sensitivities. In this presentation, an overview will be given specifying the variety of environmental information contained in ocean noise. Parameter sensitivities and performance of beamforming estimators relative to the CRLB will also be discussed. [Work supported by the Office of Naval Research Ocean Acoustics Program.]

8:55

4aUWa2. Sequential filtering and linearization for inversion in the Seabed Characterization Experiment. Zoi-Heleni Michalopoulou (Mathematical Sci., New Jersey Inst. of Technol., 323 ML King Blvd., Newark, NJ 07102, michalop@njit.edu)

Using particle filtering we can track arrival times at spatially separated phones and identify them as distinct paths. This method is very well suited to data collected during the Seabed Characterization Experiment at 16 vertically separated phones of an MPL vertical line array. The source signals are mid-frequency linear frequency modulated pulses. Several paths can be identified in the received time-series including the bottom bounce and sediment reflection. These provide a plethora of information on the seabed properties. We combine the particle filter with a linearization approach that first allows us to resolve the geometry of the experiment. For that we build a Jacobian matrix with time derivatives with respect to the unknown parameters. We proceed with an exhaustive search for sediment parameters such as sound speed and thickness. Probability density functions of arrival times are propagated backwards through the inverse model, providing estimates of densities for the unknown parameters. [Work supported by ONR.]

9:15

4aUWa3. Insights into sediment acoustics from analytic solutions of forward problems. Oleg A. Godin (Phys. Dept., Naval Postgrad. School, 833 Dyer Rd., Bldg. 232, Monterey, CA 93943-5216, oagodin@nps.edu)

Closed-form analytic solutions of wave equations are available for a rather limited subset of geoacoustic models of practical interest. Apart from elucidating the challenging underlying physics of wave propagation, the analytic solutions prove to be useful in sediment acoustics when these predict distinct, identifiable features of measured acoustic fields. Theoretical predictions can be used then to identify or constrain the type of sediment stratification, lead to quick, low-parameter inversions, guide application of statistical inversion methods, and help with interpretation of their results. In this paper, application of analytic solutions of forward problems to acoustic remote sensing of marine sediments will be illustrated by results obtained in a few experiments, which involve observations of slow interface waves near the seafloor and/or resonance peaks in either bottom-reflected energy or in the power spectrum of ambient noise. Coupling between shear and compressional waves in the stratified ocean bottom plays a key role in both examples. [Work supported, in part, by ONR.]

9:35

4aUWa4. Trans-dimensional geoacoustic inversion on the New England Mud Patch using modal dispersion data. Julien Bonnel (Woods Hole Oceanographic Inst., 266 Woods Hole Rd., MS# 11, Woods Hole, MA 02543-1050, jbonnel@whoi.edu), Stan E. Dosso (School of Earth and Ocean Sci., Univ. of Victoria, Victoria, BC, Canada), Dimitrios Eleftherakis (ENSTA Bretagne, Brest, France), N. Ross Chapman (School of Earth and Ocean Sci., Univ. of Victoria, Victoria, BC, Canada), David R. Dall'Osto, and Peter H. Dahl (Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

This paper presents trans-dimensional geoacoustic inversion of two low-frequency broadband signals recorded during the Seabed Characterization Experiment (SBCEX). The considered sources are a chirp emitted by a towed J15 source and an underwater impulse created by a combustive sound source (CSS). Corresponding received signals are recorded on single hydrophones, and the two source/receiver configurations are reciprocal, on the same track. Input data for inversion are modal time-frequency dispersion curves, estimated using warping. Inversion is then performed within a Bayesian framework. A trans-dimensional inverse algorithm is used to estimate the model parametrization (i.e., number of seabed layers), and the mode covariance matrices are estimated as a first-order autoregressive error process. Inversion results on the two reciprocal tracks are consistent with the modal information available in each dataset. Results are also consistent with what is known about the area.

9:55

4aUWa5. Application of geoacoustic inference for ecosystem monitoring of a seagrass meadow. Megan S. Ballard, Jason D. Sagers (Appl. Res. Labs. at the Univ. of Texas at Austin, P.O. Box 8029, Austin, TX 78758, meganb@arlut.utexas.edu), Gabriel R. Venegas (Dept. of Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), Kevin M. Lee, Andrew R. McNeese (Appl. Res. Labs. at the Univ. of Texas at Austin, Austin, TX), Preston S. Wilson (Dept. of Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), Abdullah Rahman (Sch. of Earth, Env., and Marine Sci., The Univ. of Texas Rio Grande Valley, South Padre Island, TX), and Justin T. Dubin (Dept. of Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

Seagrasses provide a multitude of ecosystem services: they alter water flow, cycle nutrients, stabilize sediments, support the food web structure, and provide a critical habitat for many animals. However, due to threats to seagrass meadows and their associated ecosystems, these habitats are declining globally. Since the biological processes and physical characteristics associated with seagrass are known to affect acoustic propagation due to bubble production, which results in dispersion, absorption and scattering of sound, acoustical methods are proposed to assess the health of seagrass meadows. For this purpose, an experiment was conducted in the Lower Laguna Madre where the seabed was covered by a dense growth of Thalassia testudinum. During the experiment, a combustive sound source was used to produce broadband signals at ranges of 20 m to 1000 m from the receiver location. Three sensors were positioned at the receiver location: two hydrophones located within and above the seagrass canopy, and a single-axis geophone. The data were analyzed for the purposes of inferring environmental parameters in the seagrass meadow and to investigate the feasibility of using acoustical methods to monitor ecosystem health. Initial results indicate that water column void fraction can be inferred. [Work sponsored by ARL:UT IR&D and ONR.]

10:15–10:30 Break

Contributed Papers

10:30

4aUWa6. Geoacoustic parameter variations and inversion estimates with a silt-suspension theory of mud. Elisabeth M. Brown (Mathematical Sci., Rensselaer Polytechnic Inst., Rensselaer Polytechnic Inst., Mathematical Sci., Troy, NY 12180, browne6@rpi.edu), Allan D. Pierce (Cape Cod Inst. for Sci. and Eng., East Sandwich, MA), Ying-Tsong Lin (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., Woods Hole, MA), Jason D. Chaytor (Coastal and Marine Sci. Ctr., U.S. Geological Survey, Woods Hole, MA), and William L. Siegmann (Mathematical Sci., Rensselaer Polytechnic Inst., Troy, NY)

Marine mud sediments can be modeled by a recent silt-suspension theory [Pierce, *et al.*, *JASA*, **142**, 2591 (2017) (A)], in which silt particles are embedded in a suspension of flocculated clay particles. This presentation investigates the influence on attenuation predictions from uncertainties in parameters such as the effective grain density and the distribution of grain sizes. For example, the matrix of clay flocs may effectively lower the density of silt grains and cause decreased attenuation. Effects from such parameters are determined on the regime where attenuation is nearly linear with frequency. The value of porosity is critical for specifying sound speed and attenuation in the mud layer. Porosity measurements from sediment cores can have significant uncertainty, particularly in the upper region, from the core extraction process. Inversions for porosity and other physical parameters of the silt-suspension theory are performed along a WHOI AUV track from the 2017 Seabed Characterization Experiment. Characteristics of cores guide parameter modeling and ranges for the inversions. This approach produces estimates of sound speed, attenuation, and density that are related by physical constraints. In addition, the approach allows for validity assessment of current geoacoustic mud models using measured data. [Work supported by ONR.]

10:45

4aUWa7. Maximum likelihood geoacoustic inversion from surface generated noise. John Gebbie (Adv. Mathematics Applications, Metron, Inc, 2020 SW 4th Ave., Ste. 2020, Portland, OR 97201, gebbie@metsci.com) and Martin Siderius (Dept. of Elec. and Comput. Eng., Portland State Univ., Portland, OR)

Noise generated at the surface from wind and breaking waves is incident upon the seabed at a wide range of angles and frequencies. The reflectivity of the seabed thus plays an important role in determining the vertical directionality of the noise field. It is therefore not surprising that from an information theory perspective, the noise field encodes a significant amount of information about the geoacoustic properties of the seabed that govern reflectivity, such as layering structure, sound speed, density, and attenuation. Using a physics-based ambient noise model, the Cramér-Rao lower bound (CRLB) can be computed, which specifies the lowest possible variance that an unbiased estimator can have. Existing estimators have primarily involved vertically beamforming the noise field to recover the vertical noise directionality and bottom loss, which is then processed to estimate geoacoustic seabed properties. The method described here takes a different approach by estimating the properties directly from the measured data (i.e., without beamforming). This presentation will show that maximizing the proposed likelihood function can perform better than beamforming-based techniques, and approaches the CRLB. Further, it remains unbiased in low signal-to-noise ratio conditions, is tolerant to array tilt, and can operate beyond the nominal array design frequency.

11:00

4aUWa8. Classification of acoustic and seismic signals based on the statistics of their wavelet sub-band coefficients. Costas Smaragdakis (Dept. of Mathematics and Appl. Mathematics & IACM, Univ. of Crete and FORTH, Heraklion, Greece), John Mastrokalos (Inst. of Appl. and Computational Mathematics, FORTH, Heraklion, Greece), and Michael Taroudakis (Dept. of Mathematics and Appl. Mathematics & IACM, Univ. of Crete and FORTH, Voutes University Campus, Heraklion 70013, Greece, taroud@uoc.gr)

Statistical characterization of acoustic signals is a pre-processing technique aiming at the definition of signal observables, which can be used as input data in the formulation of inverse problems of acoustical oceanography aiming at the estimation of critical parameters of the marine environment. Statistical signal characterization is also a means to classify the signals and compare their properties without reference to any physical model that determines the signal observables. Thus, it can in principle be used for the comparison of the signals of any type, including seismic recordings, thus opening the area to many applications of geophysical monitoring, using signals of any type. The paper summarizes the first attempts to study the efficiency of a signal characterization method based on wavelet transform of the signal at various levels, followed by the statistical description of their sub-band coefficients. It is shown that A-stable symmetric distributions are capable of defining the statistics of these coefficients for signals used in applications of ocean acoustic tomography and geoacoustic inversions. It is further investigated if these distributions are capable of defining the statistics of seismic signals and it is shown by preliminary results, that they can indeed be considered as possible candidates in this respect.

11:15

4aUWa9. Estimation of subbottom geoacoustic properties from offshore airgun surveys in Atlantic Canada. Graham A. Warner (JASCO Appl. Sci., 2305-4464 Markham St, Victoria, BC V8Z 7X8, Canada, graham.warner@jasco.com) and Bruce Martin (JASCO Appl. Sci., Dartmouth, NS, Canada)

This paper estimates seabed geoacoustic profiles at 14 sites offshore of the eastern Canadian coast using sound levels from a single airgun. A 210 in^3 airgun was operated along two track lines at each site and a calibrated seabed-mounted Autonomous Multichannel Acoustic Recorder equipped with a single omnidirectional hydrophone recorded the airgun sounds. High-resolution bathymetry was acquired along the tracks using a ship-mounted multibeam system and conductivity-temperature-depth profiles were measured at each site. Seabed geoacoustic profiles were estimated in a non-linear trans-dimensional Bayesian inversion using energy spectral density levels at six frequencies between 10 and 320 Hz as a function of range. The inversion estimated the unknown number of subbottom layers, geoacoustic properties (including compressional and shear wave speeds and attenuations, density, and layer thicknesses), airgun source levels, a range-correction factor, and error statistics. Data were fit with error standard deviations of 0.6–5.6 dB. Convergence of the posterior probability density function was not reached within the time limits of the study so parameter uncertainty was estimated using a linearized formulation of the inverse problem centered around the model that minimized the Bayesian Information Criterion.

11:30

4aUWa10. Classifying seabed parameters from normal incidence reflections. Megan Frantz, Martin Siderius (ECE, Portland State Univ., P.O. Box 751, Portland, OR 97207, mefrantz@pdx.edu), and Scott Schecklman (Adv. Mathematics Applications, Metron, Inc., Portland, OR)

A technique is being investigated to estimate and classify seabed parameters, including interface roughness, based on normal incident reflections. These parameters can be characterized by single beam echo sounder data or normal beams from a side scan sonar. Previous sediment classification methods used either empirical relationships based on grain size (estimated based on the strength of the echo return) or by matching data with a parameterized model of the echo envelope. [Snellen, Siemes, Simons. J Acoust Soc Am. 2011]. Here, seabed classification (including interface roughness) is estimated using a combined approach. Both the strength of the reflection and a model of the echo envelope are used to determine seabed properties. The modelled envelope is obtained from the parameterization of the scattering cross section derived from the high-frequency Kirchhoff approximation near vertical incidence [Jackson, Richardson. High-Frequency Seafloor Acoustics]. To determine the validity of the modelling approximations, a merit function has been developed to quantify the similarity between the data returned and the model results. In this presentation, the model-based approach to classifying seabed properties will be described along with results showing the exact and approximate scattering results. [Work supported by the Office of Naval Research.]

THURSDAY MORNING, 8 NOVEMBER 2018 SAANICH 1/2 (VCC), 9:00 A.M. TO 11:40 A.M.

Session 4aUWb

Underwater Acoustics, Signal Processing in Acoustics, and Structural Acoustics and Vibration: Acoustic Vector Field Studies I

Kevin B. Smith, Cochair
Department of Physics, Naval Postgraduate School, 833 Dyer Rd., Bldg. 232, Rm. 114, Monterey, CA 93943

Robert Barton, Cochair
NUWC, 1176 Howell St, Newport, RI 02841

Chair's Introduction—9:00

Invited Papers

9:05

4aUWb1. Some comments on acoustic sensing at a single point in space. Gerald L. D'Spain (Marine Physical Lab, Scripps Inst. of Oceanogr., 291 Rosecrans St., San Diego, CA 92106, gdspain@ucsd.edu), Camille Pagniello, and Timothy Rowell (Marine Physical Lab, Scripps Inst. of Oceanogr., La Jolla, CA)

The directionality provided by an acoustic vector sensor can be viewed in the context of a Taylor series expansion of the acoustic pressure field. From this perspective, acoustic pressure and acoustic particle velocity are but two of an infinite number of acoustic field variables. The purpose of this presentation is to examine some of the properties of these higher order field variables, including their energetics and the behavior of their spatial cross spectra in homogeneous, isotropic noise. Analytical expressions for those spatial cross spectra involving the normal strain rate parallel to the direction of separation decrease rapidly with increasing separation, and then continue to oscillate with a sinc function behavior. In contrast, those involving the normal strain rate perpendicular to the direction of separation decrease more slowly so that significant positive coherence exists at half-wavelength spacing, but then decay to, and remain near, zero at larger spacings. Accurate estimates of the range to a near-field point source can be obtained from the in-phase and quadrature components of normal strain rate with respect to pressure. Do animals in the ocean exploit the properties of these higher-order field variables?

9:25

4aUWb2. Exploiting ambient noise for coherent processing of mobile vector sensor arrays. Karim G. Sabra, Brendan Nichols, James S. Martin (Mech. Eng., Georgia Inst. of Technol., 771 Ferst Dr., NW, Atlanta, GA 30332-0405, karim.sabra@me.gatech.edu), and Christopher M. Verlinden (US Coast Guard Acad., La Jolla, CA)

A network of mobile sensors, such as vector sensors mounted to drifting floats, can be used as an array for locating acoustic sources in an ocean environment. Accurate localization using coherent processing on such an array dictates the locations of sensor elements must be well-known. Achieving this for a mobile, submerged array composed of individual drifting sensors is usually challenging. However the coherent processing of the ambient acoustic noise between sensor pairs can provide the separation distance between them and thus an opportunity to correct sensor location errors. Here a stochastic search algorithm is presented for identifying hidden coherent noise arrivals when the separation distance is changing faster than the required averaging time to extract such arrivals from a fixed sensor pair. The accuracy of this method matches that of GPS-derived array element positioning and its performance is shown to be improved when using directional vector sensors instead of omnidirectional hydrophones. The proposed approach is demonstrated

4a THU. AM

experimentally using ambient noise recorded by drifting vector sensors deployed in the Long Island Sound and is used to enable tracking of surface ships by coherent processing of the drifting sensors.

9:45

4aUWb3. Characteristics of very-low-frequency pulse acoustic fields measured by vector sensor and ocean bottom seismometer in shallow water. Renhe Zhang, Shihong Zhou, Yubo Qi, Yuquan Liang (State Key Lab. of Acoust., Inst. of Acoust., Chinese Acad. of Sci., Bei-si-huan-Xi Rd. No.21,Haidian District, Beijing 100190, China, shih_zhou@mail.ioa.ac.cn), and Yuanjie Sui (Qingdao Branch, Inst. of Acoust., Chinese Acad. of Sci., Qingdao, China)

In the marine environment, the effects of the seafloor and subbottom elastic media on the propagation of very low frequency (VLF) sound wave should be considered. In order to understand the mechanism of seismoacoustic field propagation, a very-low-frequency sound propagation experiment was conducted in shallow water using seafloor-located acoustic vector sensor and ocean bottom seismometers (OBS) in December 2017. The water depth is about 100 m. Pulse sources detanoted at depths of 7 m and 50 m were employed to emit the VLF signals. Comparing the signals received by OBS with that of the acoustic vector sensor, a narrow-band wave package group at VLF had been observed, which was excited by the interaction of acoustic waves in water with the layered shear seabed. In this paper, the pulse signals are used to analyze the characteristics of sound propagation and spatial correlation, amplitude and phase structure of separated normal modes based on warping transform, horizontal and vertical acoustic energy flow at different ranges. The time arrival structure and time expansion of the elastic-bottom-induced wave package group and its relationship with source distance and depth are also analyzed.

10:05

4aUWb4. A primer on vector hydrophones. Bruce A. Armstrong (GeoSpectrum Technologies Inc., 10 Akerley Blvd., Unit 19, Dartmouth, NS B3B 1J4, Canada, bruce.armstrong@geospectrum.ca)

Vector hydrophones are used by the military to track submarines, and, more recently, by scientists attempting to measure the influence of particle velocity on the behaviour of fish. Despite being in use for decades, vector hydrophones remain poorly understood by both communities. The problem is particularly acute for those aiming to measure particle velocity because vector hydrophones do not measure particle velocity directly; rather they measure pressure difference, the force that creates particle velocity. Topics covered will be the two types of vector hydrophone and why only one works well at low frequencies, the importance of knowing the phase of the directional channels relative to each other and to the pressure hydrophone, how bearings are determined, calibration of vector hydrophones and near-field effects, the magnitude of particle velocity for typical sound pressure levels, why bigger is better for mechanical noise, why neutral buoyancy is unnecessary and undesirable, and best practice for mounting.

10:25–10:40 Break

Contributed Papers

10:40

4aUWb5. Frequency warping transform of the vertical energy flux and its use for passive source ranging. Yubo Qi, Mengxiao Yu, Shihong Zhou, Renhe Zhang, Shuyuan Du (State Key Lab. of Acoust., Inst. of Acoust., Chinese Acad. of Sci., Bei-si-huan-Xi Rd., No.21, Haidian District, Beijing 100190, China, qyb@mail.ioa.ac.cn), and Mei Sun (School of Phys. and Electron. Eng., Taishan Univ., Taian, China)

The frequency warping transform of the vertical energy flux and a passive impulsive source ranging method with a single acoustic vector sensor are presented in this paper. The cross-correlation component of two different modes in the vertical energy flux is transformed into separable impulsive sequence. With a guide source, the source range is extracted from the relative delay time of the impulsive sequence. Comparing with warping the pressure autocorrelation function, there is no need to delete the modal autocorrelation component for warping the vertical energy flux, especially for its real part. Besides, the source ranging result based on the time delay of the warped vertical energy flux is much better for closer range. The frequency warping transform of the vertical energy flux and the passive source ranging method are verified by experimental data.

10:55

4aUWb6. A bottom-diffracted surface-reflected arrival in the North Pacific observed from 15 to 3200 km. Ralph A. Stephen (RASCON Assoc. LLC, PO Box 567, West Falmouth, MA 02574, rasconassoc@aol.com), S. T. Bolmer (Geology and Geophys., Woods Hole Oceanographic Inst., Woods Hole, MA), Peter F. Worcester, and Matthew Dzieciuch (Scripps Inst. of Oceanogr., La Jolla, CA)

Bottom-diffracted surface-reflected (BDSR) arrivals are a ubiquitous feature in long-range ocean acoustic propagation and are not predicted by existing forward models based on available bathymetric and bottom properties data. They were first identified in the LOAPEX Experiment in the North Pacific in 2004 where a BDSR from the side of "Seamount B" was observed from 500 to 3200 km range for an M-sequence centered at 75 Hz. In a follow-up research cruise in 2013, a BDSR was also observed from the side of "Seamount B" for ranges from 15 to 35 km and the same azimuth. Are the seafloor diffractors observed in 2004 and 2013 the same? Various measures of the diffractor location and their resolution are reviewed. Both diffractors fall within a 1km radius region. Elsewhere in the 2013 survey pairs of BDSRs were observed within a kilometer of each other. So it is possible that the two BDSR arrivals arise from two distinct and unresolvable diffractors. It is quite likely however that the two diffractor points are the same. The location of the 2004 diffractor is sufficient to explain the arrival times of the BDSR from the 2013 diffractor within the scatter of the data. [Work supported by ONR.]

11:10

4aUWb7. Underwater vector sensor communication in KOREX-17. Kang-Hoon Choi, Sunhyo Kim, Jee Woong Choi (Marine Sci. and Convergence Eng., Hanyang Univ., 55, Hanyangdaehak-ro, Sangnok-gu, Ansan-si, Gyeonggi-do 15588, South Korea, choikh0210@hanyang.ac.kr), Su-Uk Son (The 6th R&D Inst., Agency for Defense Development, Chanwon-si, Gyeongnam, South Korea), Peter H. Dahl, David R. Dall'Osto (Appl. Phys. Lab., Univ. of Washington, USA, Seattle, WA), and Ho Seuk Bae (The 6th R&D Inst., Agency for Defense Development, Changwon, South Korea)

The underwater acoustic communication channel is subject to multiple interactions with the ocean boundaries and refraction due to sound speed structure of water column, which produces significant time spread. This phenomenon is referred to as inter-symbol interference (ISI), which results in degradation of error performance. Recently, a time reversal technique has been

used for reducing the ISI. However, this requires a large-size array with spatially separated receivers to obtain higher spatial diversity gain, and it becomes a limitation to its application in space-constrained environment. An acoustic vector sensor (combined pressure and particle velocity) can potentially yield a better communication performance relative to a system using only hydrophones. In this talk, communication data collected using a vector sensor known as IVAR (Intensity Vector Autonomous Receiver) during Korea Reverberation Experiment (KOREX-17) conducted in shallow water located at 34° 43′ N, 128° 39′ E on May 23–31, 2017. The characteristics of the channel as probed by a pressure-only versus combined pressure plus particle velocity system are discussed along with some performance results of a vector sensor communication system. [Work supported by the ADD(UD170022DD) and the National Research Foundation of Korea(NRF-2016R1D1A1B03930983).]

11:25

4aUWb8. Implementation of Nuttall's vector-sensing target motion analysis algorithm. Benjamin Cray (NUWC, 1176 Howell St., Newport, RI 02841-1708, benjamin.cray@navy.mil)

Direction cosine measurements, obtained from an array of vector sensors, are processed and used to estimate the location and motion of a moving source. Nuttall derived this unpublished TMA algorithm in 2006, which recently was reviewed, implemented, and evaluated using newly collected experimental data. Both stationary and moving sources are examined. It is assumed that the array element positions are known; however, at any given time instant, the measured direction cosines are presumed noisy. Originally, Nuttall's formulation only utilized the three orthogonal components of acoustic particle velocity at each array location, however adding acoustic pressure is straight-forward and resolves bearing ambiguity. For a non-moving source, the TMA processor (linear in source position variables) generates a 3-by-3 set of simultaneous equations, with solution vector for the source position (x_s, y_s, z_s). Range from any array element to the source can then be estimated. For a moving source, a 6-by-6 set of simultaneous equations is obtained; the solution vector is now the initial source position (x_s, y_s, z_s) and the source velocity components (a, b, c). It is presumed that the source velocity is constant over the observation interval.

THURSDAY AFTERNOON, 8 NOVEMBER 2018 — THEATER (VCC), 1:15 P.M. TO 4:50 P.M.

Session 4pAA

Architectural Acoustics and Noise: Validation of Modeling and Analysis: Predictions and Outcomes

Logan D. Pippitt, Cochair
none, Marvin Hall, 1465 Jayhawk Blvd., Lawrence, KS 66045

Ben Bridgewater, Cochair
Architecture, University of Kansas, 1536 Ogden Street, Denver, CO 80218

Invited Papers

1:15

4pAA1. Practical use and extension of methods for improved source realism in auralizations. Marcus R. Mayell, Nicolaus Dulworth, and Gregory A. Miller (Threshold Acoust., LLC, 141 West Jackson Blvd., Chicago, IL 60604, mmayell@thresholdacoustics.com)

The effectiveness of an auralization as a learning and decision-making tool is greatest when listeners are able to actively engage in a plausibly realistic experience. Accuracy of the spatially varying timbral behavior of various sound sources within acoustic models has been found to greatly improve the perceived realism in auralizations. Multi-channel/phantom source modeling techniques offer the ability to better emulate sources through improved timbral representation, more authentic 3D imaging, and dynamic directivity—characteristics limited or lost by traditional point source representation methods. This technique realizes and expands upon methods using multichannel recording techniques that have been previously proposed. It offers the ability to expedite the source capture and implementation process, and to more easily represent larger/multiple sound sources while improving the feasibility and opportunity for high quality source content.

1:35

4pAA2. Speech Privacy Class calculations and post-construction measurements of a corporate workplace tenant improvement project. David Manley (DLR Group, 6457 Frances St., Omaha, NE 68106, dmanley@dlrgroup.com)

LEED interior design and construction projects IEQ acoustical credits are particularly difficult to achieve due to broad scope requirements and restrictive credit language. Specifically, the high composite sound transmission requirement between private offices and hallways. DLR Group acoustical staff proposed an alternative compliance path and evaluation focused on lower STC rated partitions with the addition of sound masking systems to increase the background noise levels, with the goal of achieving an equivalent level of speech privacy between offices and hallways. Speech Privacy Class as detailed in ASTM E2638-10 was used as the evaluation metric. Difficulties in analysis and design stage assessment included unknown level difference of constructed partitions and entry doors, correlating subjective Client expectations to objective SPC ratings, and Client sign-off on sound masking approach. Post-construction measurements

were conducted before and after sound masking system installation. Significant improvements in objective SPC ratings were measured, although targeted design goal SPC ratings were not achieved in all cases. Measured noise level differences of the glass storefront office partitions and measured sound masking noise spectra were compared to design stage values for further process refinement. Completed space measurement results and design stage comparisons will be reviewed and discussed.

1:55

4pAA3. A tale of three models, two loudspeakers, and one install. Liz Lamour Croteau (Cavanaugh Tocci Assoc., 327F Boston Post Rd., Sudbury, MA 01776, elamour@cavtocci.com)

It's commonplace to use an acoustic modeling software to confirm room conditions and loudspeaker coverage when designing a sound system. It is also becoming more commonplace for different loudspeaker manufacturers to publish—and require—the use of their own "homemade" proprietary software for modeling. In this survey, three separate software applications were utilized to model two loudspeaker options for a lecture hall at a local university with one loudspeaker solution ultimately being selected and installed. The university required graphic evidence of adequate coverage from both solutions and an on-site loudspeaker "shoot out" to determine final loudspeaker selection. Data collected along the journey includes room coverage maps for each loudspeaker with graphic elements adjusted for close-to-equal comparison, rough pink noise octave-band measurements during the demo, and final installed analysis and comparisons.

2:15

4pAA4. Learning about hall acoustics from multi-directional audio recordings. Benjamin Markham, Jonah Sacks, and Kelsey Hochgraf (Acentech Inc., 33 Moulton St., Cambridge, MA 02138, bmarkham@acentech.com)

Multi-channel audio recording, for playback in ambisonic format, is common when documenting a newly completed auditorium or an existing auditorium to be renovated. These recordings provide opportunities for comparative listening of different auditoria, and also between an acoustical model and the built auditorium. To allow meaningful comparison, recordings have been made using the same musicians playing the same musical excerpts on the same instruments. Anechoic recordings made similarly are used for convolution with synthetic (modelled) impulse responses. Comparative listening experience has yielded a number of lessons for acoustical design of auditoria and for acoustical modeling in design, including (a) the importance of instrument directivity to the sound of music in an auditorium, and perceptual responses to variations thereof, (b) the importance and challenge of loudness calibration during comparative listening, (c) inter-dependency of room size, perceived intensity and distribution of reverberation, and source directivity, (d) limitations in typical geometric acoustical modeling techniques, and (e) usefulness of directive loudspeakers when acquiring measured impulse responses for the purpose of convolving with anechoic audio. The process and results of these efforts will be presented and discussed. Selected listening segments will be available following the presentation.

Contributed Papers

2:35

4pAA5. Method of predicting Composite Sound Transmission Class rating of composite wall assemblies that do not extend to deck. Steve Pettyjohn (The Acoust. & Vib. Group, Inc., 5765 9th Ave., Sacramento, CA , spettyjohn@acousticsandvibration.com)

Predicting the sound transmitted through a partition can be very difficult because of the range of unknowns. Typically, Sound Transmission Class, STC, data provided by vendors is available to select the appropriate wall or floor/ceiling assembly. However, when partitions are made of multiple building elements, the difficulty increases. Calculating the area of each element and having an STC rating for each element allows the Composite STC rating to be calculated from the total area divided by the Total area times the transmission loss coefficient. The larger the STC rating, the smaller the Transmission Loss Coefficient. Thus, even small areas with very low STC ratings diminish the Composite STC rating. For a wall partition that stops 15 cm above the ceiling, the area and STC rating of the common area must be included in the calculation. This means that the area of the ceiling, the penetrations of the ceiling, and the area from the top of the partition must be computed. A method has been developed to account for all the paths between two spaces that provides a good estimate of the Composite STC/NIC rating of any partition. Examples and details will be provided.

2:50

4pAA6. Regression and correlation analysis of measured and simulated reverberation time in two Taiwanese Basilica churches. Chia-Fen Lee and Lucky S. Tsaih (Dept. of Architecture, National Taiwan Univ. of Sci. and Technol., 3F., No. 87, Ningbo W. St., Zhongzheng Dist., Taipei City 10075, Taiwan, rebecca830111@gmail.com)

There are few historical Basilica churches in Taiwan. None of them have been studied thoroughly on their acoustic characteristics with on-site measurements and model predictions. Wanchin Basilica of the Immaculate Conception and Tainan Theological College and Seminary Chapel were built between 1860 to 1960 AD. On-site measurement with ISO-3382 were performed to obtain reverberation time and background noise level. To obtain and document additional acoustic parameters and mappings of C_{80}, D_{50}, and STI, Odeon Acoustics was used. Since both churches are the historical buildings, it is difficult to identify the absorption coefficient of materials used in these churches for model simulation purpose. Thus, after selecting the approximate materials for the simulated model, validating the measured and simulated reverberation time with Linear Regression model is necessary. With 392 data points; standard error, least squares, and RMSD were calculated to derive the regression model with correlation (R) and determination coefficient (R^2). R^2 of regression models for both churches are 0.73 and 0.84. R values for both churches are 0.86 and 0.92. Both R and R^2 indicated that the high correlation between predicted model and measured data. Thus, the confidence of predicated C_{80}, D_{50} and STI can be raised and reliable.

3:05–3:20 Break

3:20

4pAA7. Compatibility study between building information modeling and acoustic simulation software. Kaveh Erfani, Sara Mahabadipour, Joonhee Lee, and Mazdak Nik-Bakht (Dept. of Bldg., Civil and Environ. Eng., Concordia Univ., EV 6.231, 1515 Rue Sainte-Catherine O, Montreal, QC H3H1Y3, Canada, joonhee.lee@concordia.ca)

Geometric and non-geometric data from Building Information Models (BIM) need to be manually reconstructed in acoustic simulation software to examine building acoustic performance. This process is time-consuming, error-prone, and the accuracy of acoustic analysis results will depend on the expertise of the user. The process is also unidirectional (i.e., from BIM to

the acoustic software); meaning that simulation outputs can't be easily retrieved back to the BIM. Thus, over the last decade, several attempts have been made to integrate acoustic analysis into BIM-related software during the conceptual design phase. This paper aims to facilitate the interoperability between BIM and acoustic simulation engine by improving the information exchange between them. A case study with an educational space at Concordia University in Montreal is carried out to validate the developed system and examine the practicality and efficiency of this process in action. The paper will investigate: 1) extraction of geometry data from the BIM software; 2) enriching the BIM with data regarding acoustic absorption coefficients, via an external database; 3) calculation of acoustic qualities including reverberation time, and 4) visualization of the simulation results in the BIM software.

3:35

4pAA8. Sound insulation of a single finite panel—Comparison, validation, and parametric studies. Yihe Huang, Fangliang Chen, and Tejav DeGanyar (Virtual Construction Lab, Schuco-USA, 260 W 39th St., Ste. 1801, New York, NY 10018, yhuang@schuco-usa.com)

Sound transmission characteristics of a single finite panel is of significant value for both theoretical development and engineering application in building acoustics. Extensive studies and numerous models have been developed to study the sound transmission loss behavior of a single panel over the past decades. Though most of them have been validated separately with specific test results in their developing phases, evidences showed that considerable discrepancy exit from one model to another. Numerous comparisons among different prediction models have been presented in the literature. However, quantitative comparisons among them is still lacking. To guide the design of real practice, an impartial evidence of the applicability and accuracy of different models is highly desired. For this purpose, a quantitative comparison among more than ten widely adopted existing models are presented in this study. Test results collected from different labs for different materials, that found in the literature and conducted by the authors as well, are provided first to validate different models for different scenarios. Comprehensive parametric studies are then conducted to investigate the effects of different design parameters, such as boundary conditions, damping factor, panel dimension, and aspect ratio, on the sound transmission loss of a single panel.

3:50

4pAA9. Basic study on three-dimensional sound field auralization using a scale model. Kazunori Suzuki, Yoshinari Yamada, Shinichiro Koyanagi, and Takayuki Hidaka (Res. and Development Inst., Takenaka corp., 1-5-1, Ohtsuka, Inzai-shi, Chiba 270-1395, Japan, suzuki.kazunori@takenaka.co.jp)

In order to achieve ambisonic three-dimensional sound field auralization using a scale model, the ambisonic recording procedure was reformulated based on a partly modified microphone arrangement. Specifically, a technique of measuring room acoustic response by changing a small microphone to various places in a scale model was introduced. In converting room acoustic responses to ambisonic signals, taking signal differences reduces the dynamic range of the low frequency range and deteriorates the sound quality of auralization. This is avoided by determining the microphone arrangement interval and measurement frequency range with the characteristics of the model sound source and the microphone SN ratio taken into consideration. The room acoustic responses measured in the divided frequency range was synthesized to obtain all the frequency range. To confirm the effectiveness of this method, ambisonic signals up to the secondary ones were prepared, and a fundamental psychological experiment on localization in the horizontal plane was conducted, and localization reproducibility equivalent to a full-scale model was confirmed. The presented sound contains no auditorily harmful noise and is expected to be applicable to subjective evaluation of the sound field of a hall.

4:05

4pAA10. Measurement and simulation of the sound field in a concrete grain silo. Daniela T. Helboe (Acoust., COWI AS, Karvesvingen 2, Oslo 0579, Norway, datl@cowi.com)

This paper presents impulse response measurements of a concrete grain silo and results from simulations of the measured sound field. The purpose of the measurement and simulations has been to assess the validity of the evaluation tools used for the acoustic design of an art museum, a project comprising the renovation of an existing reinforced concrete grain silo building from 1935 into a space for exhibition of art. The architects' concept involves cutting the lower half of the silos to create a central space (approximate volume 6500 m^3, height 28 m) and exhibition spaces (approximate volume 2600 m^3, average height 12 m), while keeping the upper half of the cylinders with an open end to these spaces. This paper presents how the measurements and simulations of a single concrete grain silo (approximate volume 450 m^3, height 30 m) were used to adjust relevant settings in the acoustic simulations. Uncertainties related to the simulation tools are presented and acoustical challenges imposed by the geometry of the proposed architectural concept are discussed.

4:20

4pAA11. Public lobby reverberation control using modelling software design. Ellen R. Buchan and Philippe Moquin (TSB, AB Infrastructure, 6950-113 St. NW, Edmonton, AB T6H5V7, Canada, ellen.buchan@gov.ab.ca)

A public lobby with high ceilings and hard surfaces is often found to be an excessively reverberant space. Sound-absorbing finishes to ensure a good acoustic environment are required. This case study examines the acoustical analysis of the public lobby and adjacent corridors at the Edmonton Law Courts. Improving the acoustics of this workplace/public space is important to create comfortable working conditions, and minimize noise disturbances in the nearby courtrooms. Design parameters included reverberation time and sound pressure level (SPL). ODEON Auditorium software was used to determine the optimum sound-absorbent finishes required. An array of acoustical ceiling panels and wall panels were installed in the Lobby from this simulation. The results of acoustical measurements conducted after the acoustical renovation and the ODEON Modelling predictions are discussed and compared.

4:35

4pAA12. On-stage energy enhancement in a vineyard concert hall verified by scale model measurements. Anh-Nguyen Phan, Weihwa Chiang, and Yi-Run Chen (College of Design, National Taiwan Univ. of Sci. and Technol., 43 Keelung Rd., Section 4, Taipei 106, Taiwan, whch@mail.ntust.edu.tw)

Reflective surface area near the stage of a vineyard concert hall is largely reduced when surrounded by seating terraces as opposed to by perimeter walls in a rectangular hall. Lacking of upper wall surfaces would dramatically reduce reflective energy directed back to the performer himself and to the others. The missing energy could, however, be enhanced by reflective energy from overhead, detached reflectors and other terrace walls in front of the stage. Computer simulations and scale model measurements are conducted considering tilting of side stage walls, form of the overhead reflector and height of the overhead reflector. The enhancement in both early support and reflective sound strength on stage is roughly 2.5 dB. Making the reflector surface highly diffusive causes slight drop in early support but increase in hall-averaged sound strength.

Session 4pAB

Animal Bioacoustics and Signal Processing in Acoustics: Combining Passive and Active Acoustics for Ecological Investigations

Simone Baumann-Pickering, Cochair
Scripps Institution of Oceanography, University of California San Diego, 9500 Gilman Dr., La Jolla, CA 92093

Ana ÕiroviÃ, Cochair
Scripps Institution of Oceanography, 9500 Gilman Drive MC 0205, La Jolla, CA 92093-0205

Chair's Introduction—1:30

Invited Paper

1:35

4pAB1. The ocean has music for those who ping AND listen. Kelly J. Benoit-Bird (Monterey Bay Aquarium Res. Inst., 104 COEAS Admin Bldg., Corvallis, Oregon 97331, kbb@mbari.org), Brandon L. Southall (Southall Environ. Assoc., Inc., Aptos, CA), and Mark Moline (Univ. of Delaware, Lewes, DE)

Sound is a key sense for animals in the ocean, including humans. The integration of passive and active acoustics in behavioral and ecosystem studies has revealed much about how ocean systems work. Here, I will discuss how the integration of echosounders with various passive acoustic approaches has contributed to our understanding to the biology and ecology of marine mammals. For example, explaining the habitat selection of beaked whales, the foraging ecology, behavior, and communication of spinner dolphins, and the ability of dolphin sonar to discriminate fish species. Careful integration of passive and active acoustics has also contributed to sonar signal design and processing approaches. A recent study of Risso's dolphins using passive acoustic recording tags on the dolphins themselves and echosounders deployed from a ship and an underwater vehicle provided new insights into how these animals make foraging decisions, what information is available to them and when, and how changes in their prey over space and time affect their foraging behavior. The combination of passive and active acoustics is providing insight into the ecology and dynamics of predator and prey at the scale of the individual and the population.

Contributed Papers

1:55

4pAB2. Sonar signal analysis: Biological consequences of out-of-band acoustic signals from active sonar systems. Paul A. Lepper (Wolfson School, Loughborough Univ., Loughborough LE113TU, United Kingdom, p.a.lepper@lboro.ac.uk) and Denise Risch (Scottish Assoc. of Marine Sci., Oban, Argyll, United Kingdom)

Within the aquatic environment acoustic monitoring systems are widely used to monitor and assesses anthropogenic noise sources and potential impacts on many aquatic species. Passive acoustic studies are, however, often limited to acoustic observation of changes in the species vocalization. Changes in vocalizations such as vocal calls or use of echolocation signals can be interpreted as both presence/absence, i.e., the animals have moved away or a change in acoustic behaviours, for example, not vocalizing as before. Active sonar systems can offer an attractive alternative in behavioural response studies potentially providing high-resolution spatial and temporal tracking of non-vocalizing or currently "quite" species. The main frequency of the sonar signal chosen to be outside the hearing range of the species of interests. The potential, however, exists for signal generation away from the design frequency of the sonar based on both the transducer/electronics as well as the pulsed nature of the signal. These signals might be perceived by the species of interest and therefore alter the behavioural response under study. Typical active sonar signals are analysed and compared to known hearing and integration periods for a range of species and the potential biological consequences of this out-of-band energy evaluated.

2:10

4pAB3. The use of active and passive acoustics for early detection and control of invasions. Francis Juanes (Biology, Univ. Victoria, 3800 Finnerty Rd., Victoria, BC V8P 5C2, Canada, juanes@uvic.ca)

As the threat of global invasions increase it is critically important to develop tools for early detection and control of invasive species. The earlier invasives can be detected the more feasible their control and potential eradication will be. Surveys involve the use of sensors to find and verify the identity of new invaders quickly and efficiently. Here, I summarize what features of acoustic sensors make effective tools for early detection of invasives and present examples of their use in both terrestrial and aquatic environments. Acoustic sensors combined with visual sensors allow for species identification and localization when sounds are unknown or undescribed. Technology for use of passive acoustic sensors for early detection of invasives is still in its infancy, but use will expand as methodology for automatic detection and identification develops particularly in remote settings. Once an invasive species is detected, active acoustic methods can be used to prevent range expansion or facilitate biocontrol. I will present various examples of such control methods from studies targeting fish, bird, mammals and amphibians.

2:25

4pAB4. Vertical line array tilt revealed through snapping shrimp noise.
Edward Richards, Zhuqing Yuan, Hee-Chun Song, and William S. Hodgkiss (Scripps Oceanogr., Univ. of California, San Diego, 9500 Gilman Dr., La Jolla, CA 92093, edwardlrichards@ucsd.edu)

The shallow-water acoustic variability experiment (SAVEX15), conducted in the East China Sea, unexpectedly recorded a great deal of snapping shrimp noise on a 16-element (56-m length) vertical line array deployed in 100-m water depths. These impulsive events can be used to find the tilt of the moored array which was caused by the strong ocean currents in the area. In this environment, the recorded snaps of bottom-dwelling shrimp within a radius of more than 500 m were largely separated from one another in time, and coherent time domain beamforming was successful in localizing snaps in time and range. The sparse and impulse nature of the snaps allowed for a simple normalization and gave tens of snap detections per second after a threshold detector. The results from the automated detector showed it was necessary to search with the beamformer over a three-dimensional space for reliable performance: (1) arrival time, (2) source range, and (3) array tilt. The results of the beamformer search in array tilt space are consistent with independent acoustic tilt measurements of the same array, showing that snapping shrimp noise in this region can provide valuable inference for the acoustic environment.

2:40–2:55 Break

Invited Paper

2:55

4pAB5. Relationship between cetacean presence, their prey, and oceanographic conditions at an offshore pelagic environment.
Simone Baumann-Pickering, Jennifer S. Trickey, Ally Rice, Ana Širović (Scripps Inst. of Oceanogr., Univ. of California San Diego, 9500 Gilman Dr., La Jolla, CA 92093, sbaumann@ucsd.edu), Josiah Renfree, and David A. Demer (Southwest Fisheries Sci. Ctr., National Oceanic and Atmospheric Administration, La Jolla, CA)

Large mobile predators take advantage of offshore pelagic environments that tend to be surveyed at a low temporal scale. We present results from a one-year observation of passive and active acoustics, as well as oceanographic measures, taken at a moored location 150 nmi offshore southern California in 4000 m deep pelagic waters. The passive acoustic data shows presence of delphinids and baleen, sperm, and beaked whales throughout the recording period with species-specific seasonal occurrence. Variability in small pelagics, krill, and diel vertical migrators is evident from the active acoustic data. Multiple upwelling events and associated changes in dissolved oxygen at 500 m water depth, as well as in surface waters, were also recorded over the year. We will present results on the relationships among oceanographic variability, presence of pelagic species, and top predator presence. This study will provide valuable information on ecosystem processes in meso- and epipelagic waters on a relatively small spatial scale but very fine temporal scale.

Contributed Papers

3:15

4pAB6. Predator-prey interactions in the Southern California Bight.
Ana Širović (Texas A&M Univ. Galveston, 9500 Gilman Dr. MC 0205, La Jolla, CA 92093-0205, asirovic@ucsd.edu), Simone Baumann-Pickering (Scripps Inst. of Oceanogr., La Jolla, CA), and Joseph Warren (Stony Brook Univ., Southampton, NY)

The Southern California Bight (SCB) is an important foraging area for a variety of cetacean species, including blue whales and beaked whales. However, the interactions of these species with their prey (krill and deep-water squid, respectively) can be very difficult to observe. Moorings combining passive and active acoustics were deployed at two locations in the SCB in two consecutive fall and winter seasons. A High-frequency Acoustic Recording Package sampling at 200 kHz recorded the full range of baleen and beaked whale signals. The active acoustic system on each mooring included two Simrad Wide Band Acoustic Transceivers (WBAT). A bottom-mounted WBAT had an upward looking 70 kHz transducer and the one at 300 m depth had upward looking 70 kHz and 200 kHz transducers. Over the course of four months, blue whale songs and social calls were recorded on both moorings. Beaked whale echolocation clicks were commonly detected on the mooring deployed offshore, with only rare beaked whale encounters on the coastal mooring. Backscatter data indicated overlap in the presence of cetacean predators and their prey. Scattering layer dynamics were dominated by regular diel migration, with differences in scatterer abundance between mooring sites, as well as on 2–5 day time scales.

3:30

4pAB7. Acoustic estimation of the biodiversity of fish and invertebrates.
Xavier Mouy (School of Earth and Oceans Sci., Univ. of Victoria, 2305–4464 Markham St., Victoria, BC V8Z7X8, Canada, Xavier.Mouy@jasco.com), Fabio Cabrera De Leo (Ocean Networks Canada, Victoria, BC, Canada), Francis Juanes (Biology, Univ. of Victoria, Victoria, BC, Canada), and Stan E. Dosso (School of Earth and Oceans Sci., Univ. of Victoria, Victoria, BC, Canada)

The increase in climate and anthropogenic stressors in the marine environment can lead to important losses in habitat for fishes and invertebrates. Consequently, monitoring marine biodiversity has become a critical task for ecologists. Traditional biodiversity measurements are costly and logistically challenging and there is an increasing need to develop new techniques that are more suitable for long-term and large-scale monitoring. The objective of this work is to assess the efficiency of active and passive acoustics to monitor the presence and diversity of fish and invertebrates. This work uses a multi-instrument platform, deployed on Ocean Networks Canada's VENUS cabled observatory in the Strait of Georgia (British Columbia, Canada), comprised of a high-definition video camera with a pair of LED lights, a dual-frequency imaging sonar and a hydrophone. Fish and invertebrates are counted and identified using the data from the video camera and sonar. Several acoustic indices such as acoustic complexity indices and similarity sound clusters are computed from the hydrophone data. The time series of these indices estimated from the passive acoustic data are then compared to the camera and sonar recordings to assess the ability of passive acoustics alone to determine the presence and diversity of fish and invertebrates.

3:45

4pAB8. Continuous monitoring and classification of odontocete echolocation clicks from an ocean observatory. Samuel S. Urmy, Kelly J. Benoit-Bird, John P. Ryan, and Danelle E. Cline (Monterey Bay Aquarium Res. Inst., 7700 Sandholdt Rd., Moss Landing, CA 95039, urmy@mbari.org)

Passive acoustic monitoring is a valuable technique for detecting the presence and inferring the activities of odontocetes, but is often limited by power and data-storage constraints of self-contained recorders. Since July 2015, a broadband (10 Hz–128 kHz) hydrophone has been recording continuously at the Monterey Accelerated Research System (MARS), a cabled observatory in Monterey Bay, California, USA. This acoustic record is notable for its combination of duration, bandwidth, and completeness. An automated detector identified more than 200 million unique clicks in this dataset, and extracted a variety of time-domain, spectral, and cepstral features from each click. Depending on the criteria used, clustering algorithms identified 4-8 click classes consistent with those of local odontocete species, including dolphins and beaked whales. Clicking rates were highly variable, with a median of 0.2 clicks minute^{-1} and a mean of 165 ($\pm$ 130 standard error) clicks minute^{-1}. Echolocation activity was 1-2 orders of magnitude higher at night than during daytime, and was seasonally higher in fall and winter. Prior work has shown that mesopelagic sound-scattering layers in Monterey Bay are densest at these times of year, suggesting higher availability of prey. Continuous passive monitoring has great potential to improve our understanding of these species' foraging ecology, especially when integrated with environmental measurements and active acoustic measurements of their prey fields.

4:00–4:20 Panel Discussion

THURSDAY AFTERNOON, 8 NOVEMBER 2018 ESQUIMALT (VCC), 1:00 P.M. TO 3:30 P.M.

Session 4pAO

Acoustical Oceanography: Topics in Acoustic Oceanography

Boris Katsnelson, Chair
Marine Geosciences, University of Haifa, Mt. Carmel, Haifa 31905, Israel

Contributed Papers

1:00

4pAO1. Broadband acoustic observations of individual naturally occurring hydrate-coated bubbles in the Gulf of Mexico. Elizabeth F. Weidner (Earth Sci., Univ. of New Hampshire, 24 Colovos Rd., Durham, NH 03824, eweidner@ccom.unh.edu), Kevin Jerram (Ctr. for Coastal and Ocean Mapping, Univ. of New Hampshire, Durham, NH), and Thomas C. Weber (Mech. Eng., Univ. of New Hampshire, Durham, NH)

Methane has been observed escaping from the seafloor as free gas bubbles. During ascent through the watercolumn, these bubbles undergo several processes, including accumulation of surfactants on the bubble surface. Surfactants have been theorized to increase the longevity of bubbles, facilitating methane transport to the upper ocean. Surfactants take many forms, including biological detritus, oil coatings, and hydrate shells. In 2011 and 2012, hydrate shelled bubbles were visually observed by ROV Hercules in the Northern Gulf of Mexico at the Biloxi Dome. The NOAA Ship *Okeanos Explorer* returned to the Biloxi Dome in March of 2018 with a broadband (10 kHz bandwidth) echosounder with vertical range resolution of approximately 7 cm. Individual bubbles were acoustically observed, rising from the seafloor several hundred meters before exiting the echosounder beam. In addition, the *Okeanos Explorer* made multiple survey passes over the seep site, mapping the vertical rise of the bubble plume at almost 1000 m. The dataset supports detailed modeling of the effects of hydrate coatings on acoustic response and provides an opportunity to estimate the dissolution rate of naturally occurring hydrate-coated bubbles.

1:15

4pAO2. Observations of acoustic intensity fluctuations and ambient noise near the surf zone. Kaustubha Raghukumar (Marine Sci. and Eng., Integral Consulting, 200 Washington St., Ste. 201, Santa Cruz, CA 95060, kraghukumar@integral-corp.com) and Kara Scheu (Marine Sci. and Eng., Integral Consulting, Portland, OR)

The recently completed Inner Shelf Direct Research Initiative (IS-DRI) experiment examined in great detail the physical oceanographic processes involved in the exchange of heat and momentum from outside the surf zone to the inner continental shelf, with a focus on features such as rip currents, fronts, and nonlinear internal waves. During this experiment, 27 kHz acoustic transmissions were also conducted with the goal of understanding nearshore acoustic fluctuations and their impact on acoustic communications and sonar performance. Acoustic intensity fluctuations are examined during two week-long periods during which intensity fades greater than 20 dB were observed. Variability spectra were found to be dominated at lower frequencies by tidal oscillations and higher frequency variability is attributed to intense nonlinear internal wave activity. Ambient noise spectra were found to be anisotropic, with low frequency spectral levels dominated by noise from wave breaking. When significant wave heights exceeded 3 m, a significant drop in spectral levels is observed, likely due to attenuation of sound by bubble plumes from breaking waves which are then ejected offshore by rip currents. Images from airplane flight missions confirmed the presence of offshore bubble plumes during swell events.

1:30

4pAO3. Determining spectral properties of Bowhead whale vocalizations through automated methods. Thomas V. Caero (U.S. Coast Guard, 31 Mohegan Ave., Chase 7162, New London, CT 06320, vinson.caero@gmail.com), Aaron Thode, and Christopher M. Verlinden (Scripps Inst. of Oceanogr., La Jolla, CA)

Determining spectral properties of whale vocalizations is valuable to the study of marine mammal behavior. However, manually measuring these properties is tedious, which motivates the search for automated methods. Linear regression and Radon transform methods were developed to automatically measure the slope of Bowhead whale frequency-modulated (FM) sweeps from spectrogram data. The Radon transform method was further expanded to include classification of whale vocalizations to aid in further analysis. The data collected using these methods were used to study the acoustic dispersive nature of marine mammal vocalizations in an ocean waveguide. Possible relationships were explored between the change in slope of the whale vocalization and range between each animal and the receiver.

1:45

4pAO4. Estimation of gassy layer parameters in shallow water sediment using low frequency ship noise. Boris Katsnelson (Marine GeoSci., Univ. of Haifa, Mt. Carmel, Haifa 31905, Israel, bkatsnels@univ.haifa.ac.il) and Andrey Lunkov (A.M.Prokhorov General Phys. Inst., Moscow, Russian Federation)

Sound speed in gassy sediments can reach very low values (~100 m/s) which provides high reflection coefficient for not very low frequencies (above a few tens of Hz) and specific properties of frequency dependence in the sound reflection/propagation phenomena in shallow water which allows us to estimate parameters of gassy layer (thickness and gas concentration). In the present study, shipping noise is used for estimation of mentioned parameters in Lake Kinneret (Israel). The sound noise from R/V "Hermona" was recorded at a vertical line array, moored in the lake. R/V was moving toward VLA at the speed ~4 m/s along 37 m isobath. Fourier components of the noise in the band 22–76 Hz were identified and analyzed. Mode selection was implemented to determine the mode amplitude as a function of range to the source for up to 4 normal modes. Matching mode attenuation coefficients with modeled ones for several frequencies, the thickness of the gassy layer and bubble concentration were estimated. The obtained values (thickness ~40 cm and gas content ~0.25%) correlate well with the results of direct measurements using X ray CT for cores. [This work was supported by RFBR 16-32-60194.]

2:00

4pAO5. Ray identification and ranging issues in deep water. John Boyle (OASIS Inc., 5 Militia Dr., Lexington, MA 02421, oceanpattern@gmail.com) and Kevin D. Heaney (OASIS Inc., Fairfax Station, VA)

Ocean acoustic positioning systems require accurate measurements of travel time from several acoustic beacons. When receiver position is unknown, the uncertainties in the sound speed, source position, currents, internal waves, and tides must be addressed and bounded. High travel time accuracy has been achieved in the deep ocean in ocean acoustic tomography experiments via a combination of long-duration acoustic arrival time-series, accurate source/receiver positioning, and comprehensive, post-experiment ocean modeling and data assimilation. For offline autonomous underwater positioning, those are not available. Strategies to reduce acoustic forward model uncertainties and improve acoustic ranging in deep water will be discussed.

2:15

4pAO6. Target strength measurements of pointhead flounder, *Cleisthenes pinetorum*, a flatfish without a bladder. Tohru Mukai (Faculty of Fisheries Sci., Hokkaido Univ., 3-1-1 Minato, Hakodate, Hokkaido 0418611, Japan, mukai@fish.hokudai.ac.jp), Naizheng Yan (Graduate School of Fisheries Sci., Hokkaido Univ., Hakodate, Hokkaido, Japan), Jun Yamamoto (Field Sci. Ctr. for Northern Biosphere, Hokkaido Univ., Hakodate, Japan), and Kohei Hasegawa (Faculty of Fisheries Sci., Hokkaido Univ., Hakodate, Japan)

There are only a few acoustic surveys of the bladderless fish. As is known by all, Atlantic mackerel was surveyed these years by echo-sounder and showed the high target strength at high frequency. In northern Japan, an interesting fish attracted the attention these years which called pointhead flounder, *Cleisthenes pinetorum,* without a bladder. It is a flatfish but captures prey in the middle water column. And the body shape is not the normal spindle-shaped, but a flat shape, which may cause different acoustic characteristics with Atlantic mackerel. We did the surveys in Funka Bay, northern Japan and keep the samples to the tank to measure the acoustic characteristics of live flounder. The acoustic characteristics of pointhead flounder measured both using the tether method (38, 120 kHz) and free swimming method (120 kHz). During the measurements of free swimming method, we also observed the swimming actions of pointhead flounder and calculated the swimming angle at the same time. We also discussed the characteristics of TS of the swimming angle.

2:30

4pAO7. Effect of the wind-generated bubble layer on forward scattering from the ocean surface. Yuzhe Fan, Haisen Li, and Chao Xu (Harbin Eng. Univ., No. 145, Nantong St., Nangang District, Harbin, Heilongjiang 150001, China, yuzhe.fan93@gmail.com)

Wave scattering from pressure release surfaces is a classical subject with a long history. The standard approach which treat the surface probabilistically has been well developed. Over the last decade, some problems are more effectively solved with the deterministic approach. However, the complexities of upper ocean affect acoustic scattering from the ocean surface, and wind-generated bubbles are one of the most important complexities. In this paper, the effect of wind-generated bubble layers on forward scattering from the ocean surface is studied. The extended Hall-Novarini model is used to describe the population spectral density of the bubble layer. The time-domain Helmholtz-Kirchhoff integral is used to describe the acoustic scattering from pressure release surfaces. The free-field Green's function for an inhomogeneous medium is formulated using ray method. The advantages of such conjunction are as follows: (1) the geometric shadowing effects generally neglected in the H-K integral can be handled by ray method; (2) the Helmholtz-Kirchhoff integral can provide a correct solution even in caustics zones. The numerical simulation reveals that, for large wind speeds, the amplitude of scattering signal is negligible in contrast to the amplitude of direct path signal. In such situation, the reverberation will be dominated by volume scattering of wind-generated bubbles.

2:45

4pAO8. Analysis of frontal eddies effect and horizontal refraction in Gulf of Mexico with Hybrid Coordinate Ocean Model (HYCOM). Yao Xiao (Inst. of Acoust., Chinese Acad. of Sci., 801 Ferst Dr., George W. Woodruff School of Mech. Eng., Atlanta, GA 30332, xyao62@gmail.com), Karim G. Sabra (School of Mech. Eng., Georgia Inst. of Technol., Atlanta, GA), and Zhenglin Li (Inst. of Acoust., Chinese Acad. of Sci., Beijing, China)

The cold and warm water carried by the eddies can significantly change the structure of the sound speed field, which is the key factor affecting long range deep-water sound propagation. Investigations of individual eddies have typically employed extensive deep bathy thermography readings. However, these procedures typically require a high density of readings throughout the oceanic region containing the eddy, requiring fairly extensive ship time and equipment. Further, results from these procedures are restricted to regions within the eddy where data were acquired. Fortunately, with the development of ocean models, it is convenient to obtain high-resolution ocean state estimates in global ocean as well as in regional ocean areas. In this presentation, we use the Hybrid Coordinate Ocean Model (HYCOM) extensively for high-resolution simulations to quantify the effects of vertical and horizontal refraction caused by a realistic ocean environment and actual bathymetry in Gulf of Mexico. For fixed source and receiver configurations, the impact of frequency, source depth in received pressure field and influence of frontal eddies on the travel time (and arrival angle) are examined. Moreover, the scenario and computational approach for a simulated long-range transmitted signal through a dynamic ocean with eddies is presented. Depth-averaged transmission loss, which computed by averaging the acoustic pressure-squared in the water over depth, is used to facilitate the observation of 3D horizontal refraction through frontal eddies.

3:00

4pAO9. The arrival time of mode one in a stochastic ocean. Arthur B. Baggeroer (Mech. and Elec. Eng., Massachusetts Inst. of Technol., Rm. 5-206, MIT, Cambridge, MA 02139, abb@boreas.mit.edu)

The travel time for end of the final finale is often used is in inversion algorithms for acoustic tomography experiments when there is impreciseness due to ship and mooring positions and/or motions. The rationale is the first mode has the maximum slowness and higher order ones must arrive earlier, so the finale must solely be composed of energy from the first mode. This places a constraint on the tomographic sections, e.g., on the summation of the ray path or mode group slownesses. In a two papers Dozier and Tappert (JASA, 1978) examined the re population of the mode space for signals propagating in a stochastic ocean described by a Garrett & Munk model for internal waves. In the limit of an energy conserving ocean, i.e. no loss through boundaries, the limit of the population goes to an equipartition population density which was verified by numerical experiments. For more realistic oceans with boundary losses, the limit is a race between loss and scattering. This complicates what can be well identified as mode one at long ranges at low SNR's. We perform a numerical experiment by spatial filtering for mode one along the range dependent path to select just its energy in the finale. Earlier NUMERICAL studies by the author, (JASA, 1998) suggests just five percent of the energy remained at one megameter ranges with a 1/2 Garrett/Munk ocean. [Work supported by ONR.]

3:15

4pAO10. Study on acoustic characteristics of pointhead flounder *Cleisthenes pinetorum* and juvenile walleye pollock *Gadus chalcogrammus*. Naizheng Yan (Graduate School of Fisheries Sci., Hokkaido Univ., 3-1-1, Minatocho, , Hakodate, Hokkaido 041-8611, Japan, ynz_1992@outlook.com), Tohru Mukai (Faculty of Fisheries Sci., Hokkaido Univ., Hakodate, Hokkaido, Japan), Jun Yamamoto (Field Sci. Ctr. for Northern Biosphere, Hokkaido Univ., Hakodate, Japan), and Kohei Hasegawa (Faculty of Fisheries Sci., Hokkaido Univ., Hakodate, Japan)

In this study, the acoustic differences between the pointhead flounder and the juvenile walleye pollock were examined using a quantitative echo-sounder around the Funka Bay, Japan and the acoustic characteristics of the pointhead flounder have been surveyed. Acoustics data of the fish were monitored using a Simrad EK60 (38, 120, 200 kHz) split-beam echo sounder in the field. The target strength (TS) and swimming angle of free swimming pointhead flounders were measured in a seawater tank (length: 10 m, width: 5 m, and height: 6 m). As the result, pointhead flounder schools presented a patch shaped echo on the echograms, whereas the distribution patterns of the juvenile walleye pollock schools were layered. The volume backscattering strength (SV) of the target schools extracted from the echograms showed that the pointhead flounder presented a higher SV at high frequency. In contrast, the juvenile walleye pollock showed higher SV at a low frequency. For pointhead flounder, the distribution of pitch angle was measured both by camera and echo-sounder at the experiments in the tank and shown the same distribution pattern. The TS of pointhead flounder is large bigger than other bladderless fish.

THURSDAY AFTERNOON, 8 NOVEMBER 2018 — SIDNEY (VCC), 1:30 P.M. TO 3:45 P.M.

Session 4pBA

Biomedical Acoustics: Biomedical Acoustics II

Jeffrey A. Ketterling, Chair
Riverside Research, 156 William St., New York, NY 10038

Contributed Papers

1:30

4pBA1. Focusing ultrasound into the kidney using 3D patient models. Magda Abbas, Constantin Coussios, and Robin Cleveland (Dept. of Eng. Sci., Univ. of Oxford, Inst. of Biomedical Eng., Old Rd. Campus Res. Bldg., Oxford OX3 7DQ, United Kingdom, magda.abbas@eng.ox.ac.uk)

The incidence of kidney cancer worldwide is on the rise particularly in older patients. The current gold standard of treatment is surgical resection, which has significant morbidity. HIFU is an attractive candidate for treatment of kidney tumours, as there is acoustic access and it has fewer risks then resection. However, clinical trials have had disappointing outcomes with intervening layers of fat identified as a significant obstacle. Here CTs from three patients treated in a clinical trial were segmented into different tissue types. Then acoustic simulations were undertaken using the k-Wave Matlab toolbox for a single element 0.95 MHz transducer as was used in clinical trials. Similar to previous reports we find that refraction due to fat layers results in fragmentation of the focal volume, in some but not all patients. The fragmentation was found to be sensitive to transducer placement. Therefore, a planning step was introduced where a virtual source was

placed at the target location and then the predicted acoustic field outside the body was used to determine the best position for the transducer based on the received phase and amplitude. After optimal placement of the transducer there was up to 25% increase in the focal gain.

1:45

4pBA2. Ultrasonic bioreactor to monitor 3D culture human engineered cartilage. Juan M. Melchor (Structural Mech., Univ. of Granada, Politecnico de fuentenueva, Granada, Granada 18071, Spain, jmelchor@ugr.es), Elena Lopez-Ruiz (BioPathol. and Regenerative Medicine Inst. (IBIMER), Ctr. for Biomedical Res., Granada, Spain), Juan Soto (Dept. of Optics, Faculty of Physical Sci., Complutense Univ. of Madrid, Madrid, Spain), Gema Jimenez, Cristina Antich (BioPathol. and Regenerative Medicine Inst. (IBIMER), Ctr. for Biomedical Res., Granada, Spain), Jose Manuel Baena (Breca Health Care, Granada, Spain), Macarena Perán (BioPathol. and Regenerative Medicine Inst. (IBIMER), Ctr. for Biomedical Res., Jaén, Spain), Guillermo Rus (Structural Mech., Univ. of Granada, Granada, Spain), and Juan Antonio Marchal (BioPathol. and Regenerative Medicine Inst. (IBIMER), Ctr. for Biomedical Res., Granada, Spain)

Engineered cartilage tissue is one of the most promising treatments for articular cartilage pathologies. In this work, an ultrasonic bioreactor was designed to implement a non-invasive real-time monitoring of the neo-cartilage tissue formation processes through signal analysis. Polylactic acid (PLA) scaffolds were printed and seeded with human chondrocytes. Then, they were cultured in an ultrasound (US)-integrated bioreactor. The readings from the ultrasonic sensors were analyzed by numerical models of the ultrasound-tissue interaction and by a stochastic treatment to infer the extracellular matrix (ECM) evolution. To reconstruct the velocity and attenuation from the recorded signals, a genetic-algorithm based inverse problem (IP) was combined with an iterative computational propagation. The ultrasonic data were validated against evolution measurements of the *in vitro* 3D chondrocyte cultures assessed by proliferation and morphological observations, qualitative and quantitative biochemical parameters and gene expression analysis. The significant correlation shown between glycosaminoglycans (GAG) and collagen II (Col II) expression with the elastic damping evolution of the novo ECM ($R=0.78$; $p<0.001$) and ($R=0.57$; $p<0.01$), respectively, reinforces the feasibility of using ultrasound to evaluate chondrocyte functionality. Consequently, US can be used to monitor chondrocyte proliferation and ECM formation in the context of 3D cartilage engineering.

2:00

4pBA3. Analysis of a dual aperture approach and standing wave suppression pulse sequences for controlled transvertebral focused ultrasound delivery in *ex vivo* human thoracic vertebrae. Stecia-Marie P. Fletcher and Meaghan A. O'Reilly (Physical Sci. Platform, Sunnybrook Res. Inst., 2075 Bayview Ave., Rm. M7-302, Toronto, ON M4N 3M5, Canada, sfletcher@sri.utoronto.ca)

Focused ultrasound (FUS), in conjunction with microbubbles (MB), can open the blood-brain barrier to aid therapeutic delivery. Its feasibility for blood-spinal cord barrier opening (BSCBO) has been demonstrated in small animals. Controlled transvertebral delivery of FUS to the human vertebral canal requires low frequencies to penetrate thick vertebral bone. The resulting focal depth of field (DoF) is comparable to the size of the canal, promoting the formation of standing waves (SW) and potentially compromising treatment safety. Through k-Wave simulations and experimental acoustic field scans, a confocal, dual aperture approach was investigated in combination with SW suppressing pulses (linear chirp, short bursts and random phase shift keying (PSK)) to simultaneously reduce DoF and mitigate SWs. Two transducers (470/530 kHz, angle 90°) reduced the DoF by 85% compared to a single transducer (500 kHz). While all modified pulses reduced SWs relative to a 30 cycle sinusoid, short bursts performed significantly better than longer burst methods such as linear chirps ($p=0.03$, (59 ± 15) vs (34 ± 11)% reduction). In combination with PSK, short bursts implemented in the dual aperture configuration mitigated SWs, while producing a uniform focal spot. The feasibility of using confocal, dual-aperture FUS to create a controlled focus within *ex vivo* vertebrae has been demonstrated. Next steps will include investigating MB emissions under short burst, PSK exposures, and characterizing spectral content associated with BSCBO and tissue damage.

2:15

4pBA4. Passive elastography monitoring of a high intensity ultrasound treatment: A study of feasibility in *in vitro* liver. Bruno Giammarinaro, Victor Barrère, David Melodelima, and Stefan Catheline (Univ. Lyon, Université Lyon 1, INSERM, LabTau, 151 cours Albert Thomas, Lyon Cedex 03 69424, France, bruno.giammarinaro@inserm.fr)

High intensity focused ultrasound (HIFU) is a non-invasive modality of intervention allowing thermal ablation in soft tissues by locally increasing temperature. Thermal lesions can be observed as a change in tissue elasticity properties, and so in shear wave velocity, by elastography. Most of studies based on ultrasound imaging used shear waves created by acoustic radiation force. Some of these studies presented elasticity measurements during a treatment in tissues like livers. They demonstrated that the elasticity increases during the treatment and so that the lesion becomes progressively harder than the original tissue. Acoustic radiation force method still presents some difficulties to obtain elasticity in deep tissues. However, in the human body, a natural noise due to cardiac activity or arterial pulsatility can be used to characterize the elasticity in using noise correlation techniques, it corresponds to passive elastography. This method depends on the imaging technique and so can image deep tissues. The objective is so to study the feasibility of using passive elastography technique during a treatment. Here, experiments are performed in *in vitro* porcine and bovine livers, heated with an acoustic transducer till 80°C and imaged with a high framerate ultrasound imaging device.

2:30–2:45 Break

2:45

4pBA5. Characterizing tissue calcifications using the color Doppler ultrasound twinkling artifact. Scott A. Zinck and Julianna Simon (Acoust., The Penn State Univ., 201E Appl. Sci. Bldg., University Park, PA 16802, 19scottz@gmail.com)

Rapid Doppler shifts highlight some kidney stones with a rapid change of color in ultrasound imaging in what is termed the "twinkling artifact." While many hypotheses exist to describe the origin of twinkling, the currently accepted hypothesis is that surface microbubbles are stabilized in the crevices on the surface of the kidney stone. The objective here is to evaluate the distribution of bubbles stabilized on the kidney stone surface and to determine whether other calcified tissues display the twinkling artifact. A Verasonics® research ultrasound system with the Verasonics® L22-14 and Philips/ATL L7-4 and P4-2 transducers was used to quantify twinkling on kidney stones and in minerals deposited by osteogenic stem cells over a range of frequencies. Preliminary results on kidney stones suggest that surface roughness and chemical composition influence the magnitude of twinkling. Osteogenic stem cells were also found to twinkle when minerals were present; correlating twinkling magnitude with the quantity and distribution of mineral deposition is a topic of ongoing research. The appearance of twinkling on even small tissue calcifications indicates ultrasound is very sensitive to mineral deposition and suggests that bubbles may be byproduct of the calcification process.

3:00

4pBA6. Stress reduction effects using deep breathing rhythm pattern based on speech utterance system. Seonggeon Bae (Div. of Comput. Media Information Eng., Kangnam Univ., 40, Kangnam-ro, Giheong-gu, Youngin-si, Gyeonggi-do, Youngin 446-702, South Korea, sgbae@kangnam.ac.kr) and myungjin bae (Information & TeleCommun., Soongsil Univ., Seoul, South Korea)

A breathing with a certain pattern plays an important role in heart breathing, and it is very important for daily life by relieving physical tension and finding psychological stability. In order to verify this, we analyzed the phonetically evoked organs and studied the appropriate method. In the

general study, a reliable effect of 83% was confirmed in 200 experimental groups. In the future, we will perform many studies from the viewpoint of speech signal processing using these features.

3:15

4pBA7. A study on voice enhancement using palm reflections. Hyung Woo Park and Myungjin bae (IT, Soongsil Univ., 1212 Hyungham Eng. Bldg. 369 Snagdo-Ro, Dongjak-Gu, Seoul 06978, South Korea, pphw@ssu.ac.kr)

Peoples communicates with others using voice that is essayist way. Voice is a means of conveying your opinions or thoughts. In complex social life, voices become rough and tone changes due to various causes such as condition, stress, singing, effect of drugs and changing the pitch due to changes in the numerical value of hormones. This will result in the other party not being able to communicate. In order to solve this treatment from the hospital. In this paper, we introduce the phenomenon of improving the clarity of human voice by utilizing the reflected sound waves of the palm. In previous research, We proposed the methods that using voice recorded or a solid surface mug. In this study, we propose method that peoples own voice reflected from the palm of their vocal organ. Speak the voice for "ah," "e," "o", "ooo" for a minute and make a sound wave in each mouth and massage it. The reflected voice from the palm stimulates delicate massage of the skin cells of the mouth, further enhancing the original voice enhancement function.

3:30

4pBA8. *Ex vivo* tissue preservation solutions for ultrasonic atomization. Julianna C. Simon (Graduate Program in Acoust., The Penn State Univ., Univ. Park, PA and Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab, Univ. of Washington, Penn State, 201E Appl. Sci. Bldg., University Park, PA 16802, jcsimon@psu.edu), Yak-Nam Wang, Wayne Kreider (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., Univ. of Washington, Seattle, WA), Ziyue Liu (Dept. of Biostatistics, Indiana Univ. Schools of Public Health and Medicine, Indianapolis, IN), and Michael R. Bailey (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

Tissue water content has been observed to influence ultrasonic atomization efficiency, which raises questions regarding *ex vivo* tissue storage time and viability. Here, we investigate the influence of tissue preservation solutions on acoustic properties and atomization efficiency. Samples from ten bovine livers were immediately submerged in phosphate-buffered saline (PBSaline), phosphate-buffered sucrose (PBSucrose), phosphate-buffered raffinose (PBRaffinose), or ViaSpan® clinical organ transplantation solution; no solution was also tested. Acoustic attenuation, sound speed, water content, and atomization efficiency were evaluated on either day 1 or day 2. For atomization efficiency, samples were partially submerged in water with the liver-air interface aligned with a 2-MHz focused transducer operating with 10-ms pulses at 1 Hz and peak pressures of 65/-16 MPa. Overall, no difference in atomization efficiency was observed. Water content in tissue was 66–78%, with PBSaline showing a significant increase between days ($p = 0.0013$). Only tissues in no solution showed a significant difference between days in acoustic attenuation ($p = 0.0003$). Sound speed measurements had more variation with only ViaSpan and PBSucrose showing no significant differences. As ViaSpan and PBSurcrose showed the most similarities in the tested acoustic properties, PBSurcose may be a low-cost alternative to the clinical ViaSpan solution for preserving *ex vivo* tissues. [Work supported by NIH DK043881 and EB017857.]

THURSDAY AFTERNOON, 8 NOVEMBER 2018 CRYSTAL BALLROOM (FE), 1:30 P.M. TO 3:55 P.M.

Session 4pMU

Musical Acoustics, Structural Acoustics and Vibration, and Signal Processing in Acoustics: Computational and Experimental Investigations of Flow in Musical Instruments

Whitney L. Coyle, Chair
Department of Physics, Rollins College, 1000 Holt Ave., Winter Park, FL 32789

Chair's Introduction—1:30

Invited Papers

1:35

4pMU1. Speckle imaging of air flow. Thomas Moore and Whitney L. Coyle (Dept. of Phys., Rollins College, 1000 Holt Ave., Winter Park, FL 32789, tmoore@rollins.edu)

Imaging airflow in ambient air is difficult, and it is especially difficult if the flow covers a large area. Schlieren photography is often used to image airflow, but it requires that the optics be at least as large as the area of observation, often making the cost prohibitive. Particle image velocimetry can also be used to visualize flow, but it requires specialized equipment, is computationally intensive, and imaging large areas can be challenging. We present an alternative method for imaging flow that is a variation of electronic speckle pattern interferometry. The system is easy to construct, can image large areas, and can be used to visualize both transient and steady-state flow.

1:55

4pMU2. Using speckle imaging of air flow inside musical instruments to improve computational models. Whitney L. Coyle and Thomas Moore (Dept. of Phys., Rollins College, 1000 Holt Ave., Winter Park, FL 32789, wcoyle@rollins.edu)

Measuring air flow inside a clarinet mouthpiece is a difficult task but an important characterization step if the computational model of the instrument is to be as accurate as possible. Several studies have focused on solving this problem computationally, and some experimentally, using Schlieren imaging techniques. Though effective, the Schlieren technique is tricky and time consuming and does not offer the spatial resolution necessary to detect the small changes in flow. For the clarinet, there are two flow contributions in the clarinet mouthpiece (AC/DC flow and reed-induced flow) that need to be better understood experimentally in order to include them accurately in the computational models. Here, this flow characterization begins with a simpler, non-moving-reed instrument—an organ pipe. Building off of the previous presentation, this work details the physical measurements possible with the ESPI technique and will briefly discuss the implications of this discovery on one particular computational model.

2:15

4pMU3. Navier-Stokes-based computational studies of a lip reed instrument. Nicholas Giordano (Phys., College of Sci. and Mathematics, Auburn Univ., Auburn, AL 36849, njg0003@auburn.edu)

Direct numerical solutions of the Navier-Stokes equations have been used to calculate the behavior of the flow velocity and density in a trumpet-like instrument [1]. An essential ingredient of this study is a dynamical model of the player's lips, which are described by the Adachi-Sato model [2] and incorporated into the numerical treatment using the immersed boundary method. The results are used to test previous models which rely on the Bernoulli approximation to estimate the pressure near the lips. That approximation is found to fail badly under the conditions commonly found in these instruments. Our numerical method can be readily extended to other wind instruments such as the clarinet. Work supported by NSF grant PHY1513273. [1] N. Giordano, "Physical modeling of a conical lip reed instrument," JASA 143, 38–50 (2018), to be published. [2] S. Adachi and M. Sato, "Trumpet sound simulation using a two-dimensional lip vibration model," JASA 99, 1200–1209 (1996).

Contributed Papers

2:35

4pMU4. Using Navier-Stokes modeling to design a better recorder. Jared W. Thacker and N. Giordano (Phys., Auburn Univ., 210 East Thach Ave., Apt. 24E, Auburn, AL 36830, jwt0024@tigermail.auburn.edu)

We use the Navier-Stokes equations to model how air flows in the mouthpiece region of the recorder. Different geometrical designs for this region are investigated, and the effect of these modifications on the tonal properties are studied. We find that simple changes from the standard mouthpiece geometry can have large effects on the resulting spectra which we have traced to changes in the vortex patterns near the labium. Our modeling results have been tested by using 3D printing to make instruments suggested to be most promising. The modeling results also reveal how subtle changes in features like the sharpness of the labium can have a profound effect on the behavior. [Work supported by NSF grant PHY1513273.]

2:50–3:10 Break

3:10

4pMU5. Experimental investigation of acoustic radiation from a flute with an artificial blowing device. Ai Natsubori (Dept. of Mech. Eng., Toyohashi Univ. of Technol., 1-1 Hibarigaoka,Tempaku-cho, Toyohash-shi, Aichi-ken 4418580, Japan, natsubori@aero.me.tut.ac.jp), Hiroshi Yokoyama (Dept. of Mech. Eng., Toyohashi Univ. of Technol., Toyohashi,Aichi, Japan), Akiyoshi Iida (Dept. of Mech. Eng., Toyohashi Univ. of Technol., Toyohashi-shi, Aichi-ken, Japan), and Keita Arimoto (Res. & Development Div., YAMAHA Corp., Hamamatsu-shi, Shizuoka-ken, Japan)

To clarify the effects of air jet conditions controlled by flutists on acoustic radiation, experiments were performed by an artificial blowing device with an artificial oral cavity. The experimental parameters are the flow rate, the jet angle between the jet issuing from the cavity exit and the surface of the mouth hole, the jet offset to the edge and the distance from the cavity exit to the edge. Each parameter was changed independently. The reference values for the conditions were determined by referring to the actual playing conditions, of which jet angle was measured by the Schlieren method in this research. The Reynolds number based on the cross-sectional mean velocity and jet height was changed from 1700 to 2560 by adjusting the flow rate. In this range, as the flow rate was increased, the fundamental frequency became higher, keeping the first acoustic mode. The fundamental frequency became lower for the larger jet angle within the intense radiation. When the cavity exit was biased outside the edge by the jet height, the second harmonic became more intense while the fundamental frequency was approximately constant. The shift of the fundamental frequency to the higher mode was observed for a longer jet distance.

3:25

4pMU6. Direct aeroacoustic simulation of acoustic radiation in recorders with different windway geometries. Kimie Onogi (Dept. of Mech. Eng., Toyohashi Univ. of Technol., 1-1 Hibarigaoka, Tempaku-cho, Toyohashi-shi, Aichi-ken 441-8580, Japan, onogi@aero.me.tut.ac.jp), Hiroshi Yokoyama (Dept. of Mech. Eng., Toyohashi Univ. of Technol., Toyohashi, Aichi, Japan), Akiyoshi Iida (Dept. of Mech. Eng., Toyohashi Univ. of Technol., Toyohashi-shi, Aichi-ken, Japan), and Tetsuro Shoji (Res. & Development Div., YAMAHA Corp., Hamamatsu-shi, Shizuoka-ken, Japan)

To clarify the effects of the windway geometry on the aeroacoustic feedback in the jet fluctuations in recorders, direct aeroacoustic simulations were performed along with experiments. The simulations were based on the compressible Navier-Stokes equations to predict the fluid-acoustic interactions. The volume penalization method was used to reproduce the flow and acoustic fields around the complex shape of the recorders. Two recorders with straight and arch-shaped windway were explored. The occurrence of mode change was observed at the higher velocity for an arch-shaped windway model compared with the straight windway model. The modified formulation of the negative displacement model (N.H.Fletcher *et al.*, 1976, J. Acoust.) was proposed based on the predicted jet fluctuations, where the jet fluctuations were divided into hydrodynamic and acoustic components. The ratio of the hydrodynamic component to the acoustic component near the windway exit was lower in the arch-shaped windway model than that in the straight windway model, whereas the amplification factor of the jet fluctuations was larger in the arch-shaped windway model. The relevance of these results and the mode change along the jet velocity is to be discussed.

3:40

4pMU7. Violin surface velocities and radiativity. Chironjeev Kanjilal and Chris Waltham (Phys. & Astronomy, UBC, 6224 Agricultural Rd., Vancouver, BC V6T 1Z1, Canada, chironjeev.kanjilal@gmail.com)

Complete surface velocity maps and radiativity measurements have been made of several violins of varying quality. The instruments were excited laterally at the top of the bridge with a small impact hammer, giving a strong signal up to 4 kHz. The wood velocities were measured with a 0.2 g accelerometer placed sequentially on a grid, all over the surface. The grid was small enough to allow a detailed look at the bridge island. The f-hole velocities were measured with a 6 mm diameter microphone at nine positions per hole, and the acoustic pressures were converted to velocities with a propagating wavefront model. The surface measurements were compared to radiation measurements made with a 92 cm radius microphone array in an anechoic chamber. The surface measurements could be completed in a day, the radiativities in two hours. The resulting maps of operating deflection shapes, calibrated in m/s per N, allow a detailed examination of the vibrational behavior of each instrument, at frequencies up to and including the first torsional mode of the bridge.

THURSDAY AFTERNOON, 8 NOVEMBER 2018 SALON C (VCC), 1:00 P.M. TO 2:15 P.M.

Session 4pNSa

Noise, Speech Communication, and Psychological and Physiological Acoustics: Effects of Noise on Human Performance II

Joonhee Lee, Cochair
Department of Building, Civil and Environmental Engineering, Concordia University, EV 6.231, 1515 Rue Sainte-Catherine O, Montreal, QC H3H1Y3, Canada

Z. Ellen Peng, Cochair
Waisman Center, University of Wisconsin-Madison, 1500 Highland Avenue, Madison, WI 53711

Contributed Papers

1:00

4pNSa1. Comparing resonance frequencies in muscle tissue before and after exercise induced damage. Garrett Jones, Cameron Smallwood (Mechanical Eng., Brigham Young Univ., CTB 435Q, Provo, UT 84602, garrett-jones.co@gmail.com), Brent Feland (Exercise Sci., Brigham Young Univ., Provo, UT), and Jonathan Blotter (Mechancial Eng., Brigham Young Univ., Provo, UT)

There are various approaches to measuring muscle fatigue, including such simple measures as surveys for qualitative data, and measuring power output for quantitative data. Electromyography is consistently used to determine muscle fatigue. However, a quantitative measure of muscle recovery after exercise-induced damage is yet to be defined. The objective of this study is to measure the resonance frequency of a muscle group using a 3D laser vibrometer system and make a correlation with both elastography stiffness measurements, and with recovery of a muscle group based on standard muscle fatigue measurement techniques. Measurement of the resonance frequency of a muscle group has been successfully performed, and there is good reason to believe that the resonance frequency of the muscle group does shift after exercise-induced damage occurs. The main outcome of this study will determine whether the shift in resonance frequency of a muscle group can be used to track recovery after a damage protocol.

1:15

4pNSa2. Workplace noise assessments by open-plan office occupants: Relationship with ISO 3382-3 metrics, and psychoacoustic parameters. Manuj Yadav, Densil Cabrera, James Love, Jungsoo Kim, and Richard de Dear (School of Architecture, Design and Planning, The Univ. of Sydney, Rm. 589, Wilkinson Bldg., 140, City Rd., NSW 2006, Australia, manuj.yadav@sydney.edu.au)

ISO 3382-3:2012 is the international standard for measuring the acoustics of open-plan offices. These measurements are performed in *unoccupied* offices, and it has recently been shown that some of the ISO 3382-3 metrics, especially distraction distance, are useful in predicting the occupants' perceived disturbance due to noise. The current research compares the ISO 3382-3 metrics, with several psychoacoustic parameters (loudness, fluctuation strength, etc.) derived from measurements done in an *occupied* state of open-plan offices, in terms of their usefulness in predicting occupants' perception of several aspects of workplace noise. This comparison was based on using measurements performed in several offices in both occupied and unoccupied states, along with a workplace noise survey that was conducted in these offices. The results will enable a more comprehensive understanding of noise-related issues in open-plan offices, as it involves both physical acoustic, and psychoacoustic considerations.

1:30

4pNSa3. A hearing-care toolkit for young musicians: Combined use of a noise dose measurement app and acoustical manekin. Romain Dumoulin (Ctr. for Interdisciplinary Res. in Music Media and Technol., CIRMMT, 527 Sherbrooke St. West, Montreal, QC H3A 1E3, Canada, romain.dumoulin@mcgill.ca), Francesco Tordini (Ctr. for Interdisciplinary Res. in Music Media and Technol., Montréal, QC, Canada), Jeremie Voix (École de technologie supérieure, Université du Québec, Montréal, QC, Canada), and Isabelle Cossette (Ctr. for Interdisciplinary Res. in Music Media and Technol., Montréal, QC, Canada)

Within a joint initiative with the Schulich School of Music at McGill University, the authors developed an original approach to raise awareness about hearing health by combining a participative assessment with a novel noise exposure risk indicator. The overall noise exposure is estimated using two complementary measurement systems addressing, respectively, the contribution determined by *(i)* the use of portable music players and *(ii)* music and non-music activities. The first system is a "listening-level measuring kiosk," a freely accessible station on which students can place their own headphones or earphones and playback (at their preferred listening levels) music excerpts from their own playlist on an acoustical manikin. The second system is a personal, smartphone-based, "noise exposure" portable measurement device. Users start with an initial training and an audiometry screening, followed by the collection of exposure data during a 4-week assessment period. Personalized results are provided based on individual exposure profile and initial audiogram. The validation of the design of a novel noise exposure risk indicator is based on the feedback from a focus group of young musicians. This paper describes the measurement systems, assessment process, and preliminary results from the ongoing pilot studies.

1:45

4pNSa4. Inciting our children to turn their music down: The AYE concept. Jeremie Voix and Romain Dumoulin (École de technologie supérieure, Université du Québec, 1100 Notre-Dame Ouest, Montréal, QC H3C 1K3, Canada, jeremie.voix@etsmtl.ca)

According to the World Health Organization (WHO), more than 1.1 billion people are currently at risk of losing their hearing due to excessive exposure to noise. Of this, a significant proportion consists of children, youth and young adults who are exposing themselves to excessive levels of sound through various leisure activities (music players, concerts, movies at the theatre, dance clubs, etc.). To address this issue, many approaches have been developed, ranging from general awareness messages to volume limiters on personal music players. Nevertheless, it seems that, as with all dangerous behaviors, the most promising approach is a direct one in which a personalized assessment of the risk has been made and the person is made aware of the associated detrimental consequences. The present paper describes a counseling approach whereby the sound exposure is associated with an increased loss of hearing sensitivity. The proposed "Age of Your Ears (AYE)" metric is computed using the ISO 1999 multiregression model and predicts for a given age and a given sound exposure the resulting accelerated aging of the person's hearing. Preliminary results from pilot studies and focus groups with youths will be presented together with the underlying mathematical and statistical foundations.

2:00

4pNSa5. Debunking unusual false noise damage claims. Marlund E. Hale (Adv. Eng. Acoust., 663 Bristol Ave., Simi Valley, CA 93065, mehale@aol.com)

We live in a society where noise seems to be everywhere and some of it is relatively loud, very annoying or even painful. In some cases we use noise to mask out noise, or we use ear plugs or we create quite spaces or "sound proof" rooms, just to get some peace and quiet. Nowadays, nearly everyone is aware of the annoyance or potential harm caused by other people's noise. This has been a liability issue for employer's and insurance companies. Laws, ordinances, regulations and standards (LORS) have been passed or otherwise put into place in an attempt to remedy excessive noise situations. Occupational hearing loss claims, violation fines, jail time and civil lawsuits have all taken place because of noise conflicts. This paper presents three examples where litigation was attempted for financial gain or to impose a heavy financial burden and force a business closure, when the plaintiffs enacted noise damage legal claims. Examples of noise from (1) a pick-up mounted railroad engine horn, (2) a very large rural private gun club with SWAT, law enforcement and military "Afghanistan-type" progressive urban and desert conflict scenarios, and (3) an EMT training class explosion simulation. Claims and outcomes are discussed.

Session 4pNSb

Noise and Psychological and Physiological Acoustics: Soundscapes

Kerrie G. Standlee, Chair
Daly-Standlee & Associates, Inc., 4900 SW Griffith Drive, Suite 205, Beaverton, OR 97005

Chair's Introduction—2:45

Contributed Papers

2:50

4pNSb1. Reducing the noise of tramways in urban areas. Rene Weinandy and Percy Appel (Noise Abatement in Transport, German Environment Agency, Woerlitzer Platz 1, Dessau-Rosslau 06844, Germany, rene.weinandy@uba.de)

One of the most important environmental issues in densely populated areas is the problem of noise. Traffic noise from cars, railway vehicles and airports located in close proximity to the city is not only annoying for residents; it also leads to serious health issues and has an enormous negative economic impact. Due to this, it is of primary importance for city planners, engineers and politicians to make our cities quieter. The research project "Reducing the noise of tramways in urban areas"—initiated and supervised by the German Environment Agency—investigates potential noise abatement technologies and its implementation for tramways in Germany. Urban areas are growing worldwide and due to the resulting progressive consolidation, public transport including tramways will expand rapidly and subsequently raise severe environmental problems, i.e., noise. In order to protect the people, it is important to operate public transport as quiet as possible. However, the current legal, technical and economic conditions in Germany are only partially suitable to achieve that. Therefore, the research project investigates technical and operational concepts for noise abatement of tramways, possibilities of constructional sound insulation and legal aspects for the implementation of noise abatement measures. The presentation identifies and describes obstacles for the market penetration, operational problems, maintenance, and legislation.

3:05

4pNSb2. Measurement of the sound directivity of tonal and broadband reversing alarms in various controlled environments. Olivier Robin, Tamara Krpic (Groupe d'Acoustique de l'Université de Sherbrooke, Université de Sherbrooke, Université de Sherbrooke, Sherbrooke, QC J1K 2R1, Canada, Olivier.Robin@USherbrooke.ca), Hugues Nélisse (Institut de recherche Robert-Sauvé en santé et en Sécurité du travail (IRSST), Montreal, QC, Canada), and Alain Berry (Groupe d'Acoustique de l'Université de Sherbrooke, Université de Sherbrooke, Sherbrooke, QC, Canada)

Tonal reversing alarms are a recognized source of noise annoyance and recent broadband alarms target a reduction of the environmental impact of backup sound alarms. From the worker and safety point, the sound field generated behind vehicles by broadband alarms is considered more uniform and its source easier to localize in space, a main drawback of tonal alarms being to exhibit large amplitude variations in the produced sound field pattern. Very few datasets are nevertheless available in order to objectively compare these two alarm types. The zone where the generated sound field is studied is also almost all the time restricted to the rear of the vehicle, even though additional diffraction effects can occur around the vehicle due to its finite volume and geometry. For two commercial alarms (tonal and broadband), 3D directivity patterns are obtained from microphone array measurements performed in controlled conditions, i.e., in a hemi-anechoic room and a reverberant room. The backup alarms are either placed on a stand at the center of the room, or on a parallelepiped surface representing the rear part of a small fork lift. The effect of the floor condition, i.e., reflective or absorbing, is also studied in the hemi-anechoic room.

3:20

4pNSb3. Derivation of network-wide surface condition corrections for rail noise modelling. Briony Croft (SLR Consulting, 200-1620 West 8th Ave., Vancouver, BC V6J1V4, Canada, bcroft@slrconsulting.com), Tasia Balding (Translink, Vancouver, BC, Canada), Pascal Everton, and Jasen Stein (SLR Consulting, Calgary, AB, Canada)

Noise modelling and mapping techniques are increasingly being used to understand noise impacts to communities across large geographic areas. Rail transit noise around a network can vary considerably with factors such as track type, train speed, rail roughness/surface condition, among others. Reference source levels for a particular rolling stock type are normally derived by measurement at a particular location or locations, and applied around the network. When modelling noise network wide, it is important to understand how variable rail roughness can affect predicted noise levels at locations other than those used for measurements. With elevated guideway, it can also be difficult to access appropriate external measurement locations. This study describes a study undertaken for the Vancouver SkyTrain network using the Nord2000 rail noise prediction algorithm. Variations in rolling noise level of the order of 15 dB were identified and attributed to rail surface condition. Noise measurements inside a test train were used to determine frequency dependent train speed coefficients and corrections to apply for rail condition around the network. This paper describes the approach used and the outcomes in relation to this city-wide rail noise mapping project.

3:35

4pNSb4. Sound quality metrics applied to road noise evaluation. Karolina Marciniuk (Multimedia Systems Dept., Gdansk Univ. of Technol., Faculty of ETI, Gdansk, Poland) and Bozena Kostek (Audio Acoust. Lab., Gdansk Univ. of Technol., Faculty of ETI, Narutowicza 11/12, Gdansk 80-233, Poland, bokostek@audioakustyka.org)

Road noise monitoring systems typically measure sound levels in specific time periods. The more insightful approach suggests to measure also the nature of noise. Sound quality of sounds such as car noise can be objectively evaluated by several parameters. One of them is psychoacoustic annoyance, described by loudness, tone color, and the temporal structure of sound. In this paper the assessment of several sound quality parameters, such as: loudness of time-varying noise, sharpness, fluctuation strength, and roughness, as well some additional parameters borrowed from the Music Information Retrieval area are presented. Then a comparison between parameter values obtained by means of the professional measurement system and a tool working in the Matlab environment is performed. The conducted investigations are carried out using recorded samples of an individual vehicle

pass-by in close proximity of the road, organized as a database of road traffic noise recordings. [This research was partly supported by the Polish National Centre for Research and Development within the Grant No. OT4-4B/AGH-PG-WSTKT.]

3:50

4pNSb5. Soundscape design connecting people with the environment. Taiko Shono (Office Shono, 4-32-6-614, Nishishinjuku, Tokyo, Shinjuku-ku 160-0023, Japan, shsh@aaa.email.ne.jp) and Hidemaro Shimoda (Acoust. Planning Corp., Tokyo, Minato-ku, Japan)

We have been engaged in soundscape design that enables people to connect with the environment and, in time, with the world beyond our immediate surrounds through sounds. We create sounds employing the elements on site: waves, wind, rain, spring water, etc., and strive to bring people into creative contact with our earth's environment and make people aware of something which may be present yet typically goes unnoticed. All of these works have been designed not as temporary performances but as permanent constructions responding to their environments. We have also dealt with "sound as media," employing not only audio from the natural world but also the familiar sounds of daily life, communities, and events, creating opportunities for people to rediscover the world around them in a new light. In addition, We have offered programs designed to awaken listening sense, including workshops at museums and other cultural facilities and university classes, with themes such as "aural adventure." In this paper, we show some examples of our activities and how we design the mechanisms of gathering or creating sounds.

4:05

4pNSb6. Urban soundscape: A case study in Taipei Dome. ChiaYuan Chuo and Lucky S. Tsaih (Dept. of Architecture, National Taiwan Univ. of Sci. and Technol., No. 43, Keelung Rd., Sec.4, Da'an Dist., Taipei City, Taipei 10607, Taiwan, isnoopy307@gmail.com)

Taipei Dome (TD) is under construction and located in the heart of Taipei City. It is surrounded by prominent shopping district and popular cultural centers. The capacity of TD is 40,575 seats. Songshan Cultural and Creative Park is within the same block as TD and has retained the remaining green area within the block. Once it is in operation, the impact of the subsequent vehicular and crowd noise to the nearby buildings and green area is inevitable. To understand the nearby soundscape of TD, an itinerary with 8 locations were used. In-situ visual/aural observation with video recording were performed in 4 different time periods during one weekday and one weekend. A list of taxonomy of sounds was derived. While vehicular noise is the dominant sound category, natural sound such as bird chirping and crickets calling were also identified. Traffic counts were conducted. Scooters and cars were the two main vehicular types and with peak volumes during weekend evenings. Sound pressure level of each location was measured. Maximum peak sound pressure level of car noise reached as high as 109.5 dB. Mitigating expected vehicular noise and enhancing natural sound around TD should be considered by its design team.

4:20

4pNSb7. Soundscape approach integrating noise mapping in Namsan Promenade. Jisu Yoo (Energy Environment System Eng., Univ. of Seoul, 309 2nd Eng. Bldg., 163 Seoulsiripdae-ro, Seoul, Dongdaemun-gu 02504, South Korea, luvxjisu@gmail.com), Kyong Seok Ki (Sangji Univ., Seoul, South Korea), Hunjae Ryu (Energy Environment System Eng., Univ. of Seoul, Seoul, South Korea), Ji Suk Kim (Jungbu Parks & Landscape Management Office, Seoul Metropolitan Government, Seoul, South Korea), and Seo I. Chang (Environment Eng., Univ. of Seoul, Seoul, South Korea)

Namsan is a 262 m high mountain located in the center of Seoul. Its easy accessibility and tourist attractions such as a cable car and a tall tower on the top allure native and foreign people. It functions as a park rather than a mountain and has a 7.5 km promenade around it with a moderate slope where even the old and the weak enjoy walking. Some parts of the promenade are surrounded by trees and flowing waters but other parts are influenced by shuttle buses, roundabout road-traffic and various machines etc. Noise maps for the sources were generated to find the spatial distribution of the noise over the mountain including the promenade. By observation along the promenade, natural sound including mainly birdsong and water streaming sound were identified and sound maps were generated. To collect individual responses from 4 acousticians and 6 non-acoustic people about an on-site questionnaire of the sound environment along the promenade, a total of 37 spots were selected apart from each other by approximately 200 m. The responses were used to generate soundscape maps. Separate and combined analyses of the three acoustic maps were performed to propose some measures to improve sound quality of the promenade.

4:35

4pNSb8. Sound grade classification with sound mapping of national park trails in South Korea. Hunjae Ryu (Energy and Environ. System Eng., Univ. of Seoul, 2nd Eng. Bldg. 309, Seoulsiripdae-ro, Dongdaemun-gu, Seoul 02504, South Korea, pgryuno1@uos.ac.kr), Kyong Seok Ki (Horticulture and Landscape Architecture, Sangji Univ., Wonju-si, South Korea), Jisu Yoo, Seo I. Chang (Energy and Environ. System Eng., Univ. of Seoul, Seoul, South Korea), and Bo-Hyun Kim (Conservation Planning Div., Korea National Park Service, Wonju-si, South Korea)

Most of the national parks in South Korea are exposed to noise pollution caused by urban noise from adjacent cities. Especially, the road-traffic noise from the roads passing through or surrounding the parks has increased the exposure. In addition, the military and tele-communication facilities located in ecologically sensitive highlands are noise sources and affect nearby ecosystems. Therefore, it is important to preserve and maintain a comfortable sound environment to ensure the quality of the park trails for visitors and the habitat of animals and plants. The purpose of this study is to map a sound environment of national parks in South Korea and to classify sound grades of park trails. The park trails were categorized into 5 different groups based on acoustic factors such as noise level, loudness, sharpness, roughness, and tonality, and environmental factors such as land-use, ground coverage, vegetation type, and relative location which can be obtained from biotope map and GIS DB. This classification was based on factor analysis and cluster analysis. It is expected that this sound grade classification of the national park trails meets the right of visitors to know the sound environment, and that the managers of national parks can use it as a guideline to create a positive sound environment.

4:50

4pNSb9. A study on the hearing reverence psychology using rhythm patterns. SangHwi Jee, Myungjin Bae, and Myungsook Kim (Sori Sound Eng. Lab, Soongsil Univ. Seoul, Seoul 06978, South Korea, slayernights@ssu.ac.kr)

Humans are exposed to noise from the moment they are born until they die. Noise is a sound that humans do not like to hear. In the case of automobile horn noises, it produces a noise of more than 100 dB, which causes stress to people, which leads to reprisal driving and becomes a social problem. In the case of a light machine, it produces a very sharp sound, which causes stress when people listen. Therefore, we propose a new sound that moves the rhythm and pattern based on the existing sound. If we make rhythmic sounds using a rhythm rather than a constant rhythm, it induces less stress than conventional rhythmic sounds.

Session 4pPA

Physical Acoustics and Biomedical Acoustics: Interactions of Sound Beams with Objects II

Likun Zhang, Cochair
University of Mississippi, 145 Hill Drive, Oxford, MS 38677

Grant Eastland, Cochair
System Acceptance & Operational Readiness, Naval Undersea Warfare Center Division Keyport, 610 Dowell St., Keyport, WA 98345

Invited Papers

1:00

4pPA1. Strongly focused vortex beams by using flat Fresnel-spiral lenses. Noé Jiménez (Instituto de Instrumentación para Imagen Molecular (I3M), Consejo Superior de Investigaciones Científicas (CSIC), Edificio 8B, Acceso N, 1[a] Planta, Camino de Vera s/n, Valencia 46022, Spain, nojigon@upv.es), Vicente Romero-Garcìa (Laboratoire d'Acoustique de la Université du Mans UMR 6613, CNRS, Le Mans cedex 9, France), Lluis M. García-Raffi (IUMPA, Universitat Politècnica de València, València, Spain), Francisco Camarena (Instituto de Instrumentación para Imagen Molecular (I3M), Universitat Politècnica de València, Valencia, Spain), and Kestutis Staliunas (ICREA, Universitat Politècnica de Catalunya, Tarrasa, Spain)

We report geometrically-optimal diffraction gratings for sharp vortex beam focusing using Fresnel-spiral curves. The lenses are built based on the Fresnel-spiral, a spiral curve that combine the focusing properties of Fresnel zone plates and the phase dislocations produced by spiral gratings. On the one hand, the constructive and destructive interferences between open and opaque zones in the grating, in analogy to the Fresnel zone plate, allow sharp beam focusing. On the other hand, the spiral shape of the grating retains the helicity, rotating the phase of the diffracted waves and creating a phase dislocation along the axis. This allows the generation of geometrically optimal focused vortex beams, enhancing the field intensity at the focus up to 170 times. In particular, this system offers a tunable topological charge of the vortex beam by using different arms in the Fresnel-spiral diffraction grating, being the topological charge equal to the number of arms. Two different Fresnel-spiral diffraction gratings with topological charge of 1 and 5 are experimentally tested showing excellent agreement with theory and simulations. These diffraction gratings will allow the design of effective wave-matter interaction systems, with direct applications in industry and biomedical engineering.

1:20

4pPA2. Bessel beam superposition for analyzing off-axial scattering, forces, and torques. Likun Zhang (Univ. of MS, 145 Hill Dr., Oxford, MS 38677, zhang@olemiss.edu)

Illumination of an arbitrary-order Bessel beam on an off-axial object is expanded as illuminations of a series of Bessel beam components of different orders on an axially centered object [L. Zhang, J. Acoust. Soc. Am. 143(5), 2796–2800 (2018)]. The expansion follows from a parallel-axis relation that is derived as a generization of Graf's addition theorem of Bessel functions. The off-axial Bessel beam expansion enables to understand the off-axial scattering, forces, and torques in terms of superposing that in the on-axial illuminations by the series of beam components. The superposition reveal negative axial radiation force and negative radiation torque in terms of contributions of different beam components in the on-axial configuration. The exact locations of the object for torque direction reversal are analytically identified for a small particle. The superposition also reveals that the off-axial radiation torque on a small particle exists even for higher-order vortex beams or vortices.

1:40

4pPA3. Expansion of linear focused sound fields using Bessel beams. Timothy D. Daniel (Phys., WSU, Phys. and Astronomy Dept., Washington State Univ., Pullman, WA 99164, timothy.daniel@email.wsu.edu), Ivars P. Kirsteins (NUWCDIVNPT, Newport, RI), and Philip L. Marston (Phys., WSU, Pullman, WA)

Previous work on scattering by Bessel beams shows that expansion of incident sound fields in term of these beams has application to scattering [P. L. Marston, J. Acoust. Soc. Am. 122, 247–252 (2007)]. In this work, we expand a focused cylindrically symmetric sound field in terms of Bessel beam components. In particular, the authors are interested in a focused beam radiated from a spherical cap source. A physical optics model is applied to sound propagation close to the source to facilitate the calculation of the Bessel beam expansion coefficients. This type of model is useful for focused scattering [P. L. Marston and D. S. Langley, J. Acoust. Soc. Am. 73, 1464–1475 (1983)]. Once the expansion coefficients are found the sound field can be evaluated by the appropriate superposition. The model gives results in agreement with O'Neil [H. T. O'Neil, J. Acoust. Soc. Am. 21, 516–526 (1949)] and Chen [X. Chen, K. Q. Schwarz and K. J. Parker, J. Acoust. Soc. Am. 94, 2979–2991 (1993)]. Comparison is also made with direct integration of a Kirchhoff integral for both on and off axis pressures with good agreement. [Work supported by ONR.]

2:00

4pPA4. Experimental analysis of the instantaneous mechanical torque applied to objects by means of acoustic beams. Ruben D. Muelas-Hurtado (School of Civil Eng. and Geomatics, Universidad del Valle, Cali, Valle del Cauca, Colombia), Joao L. Ealo (School of Mech. Eng., Universidad del Valle, Ciudad Universitaria Meléndez. Bldg. 351, Cali 760032, Colombia, joao.ealo@correounivalle.edu.co), and Karen Volke-Sepulveda (Instituto de Física, Universidad Nacional Autónoma de México - UNAM, Mexico City, Mexico D.F., Mexico)

In this work, we quantify the mechanical torque and the angular momentum transferred by an acoustic beam (AB) to disk-like samples of flat surface and chiral objects. Also, two types of acoustic beams are generated, i.e., quasi-planar and helical beams. To produce the AB, we have fabricated a multitransducer of 123 ultrasonic elements (40 kHz) deployed on a triangular lattice which allows us an electronic wavefront synthesis by modifying the phase of each emitter. The samples to be freely rotated are supported at its center of mass by a sharp tip of a thin metallic needle vertically located at the geometrical center of the multitransducer. The instantaneous angular response of samples of different size, made of cork and plastic, was extracted using a fixed video camera. The instantaneous torque exerted on each sample and the transferred angular momentum are estimated by fitting the output of a dynamic model of the sample behavior to empirical data. The experimental apparatus proves to be useful to estimate the acoustic torque induced by AB on objects and its dependency on the geometries of the samples and the AB wavefront, their relative size and the intensity of the acoustic beam.

2:20–2:35 Break

2:35

4pPA5. Acoustic radiation force manipulation of tissue, cell, and interactions with neurons. Hairong Zheng (Shenzhen Institutes of Adv. Technol.,Chinese Acad. of Sci., 1068 Xueyuan Ave., Shenzhen University Town, Shenzhen, China, hr.zheng@siat.ac.cn) and Feiyan Cai (Shenzhen Institutes of Adv. Technol.,Chinese Acad. of Sci., Shenzhen, China)

It is well known that an object in a sound field that absorbs, scatters, or reflects sound energy is subjected to the acoustic radiation force, and thus can be manipulated without contact. This unidirectional force has profound application in biomedicine. For example, the acoustic radiation force can be used to generate localized displacements in tissue, and the magnitude of the tissue displacement is inversely proportional to the local stiffness of the tissue. Based on this principle, the acoustic radiation force interaction with the tissue can be used for elasticity Imaging. The acoustic radiation force can also be used to manipulate cell's movement and simulate isolated neuron, which may have potential application in tissue engineering and ultrasound neuro-modulation. In this study, our recent research on the application of the acoustic radiation force for manipulation of tissue, cell and interactions with neurons, will be presented. New approaches or systems for generation acoustic radiation force with special application will also be shown.

2:55

4pPA6. Orbital motion of a particle levitated in a standing-vortex acoustical trap. Jhon F. Pazos-Ospina (Marco Fidel Suarez Military Aviation School (EMAVI), Colombia Air Force, Universidad del Valle, School of Mech. Eng. Edif. 351, Cali, Colombia, jhon.f.pazos@correounivalle.edu.co), Joao L. Ealo (School of Mech. Eng., Universidad del Valle, Cali, Colombia), and Karen Volke-Sepulveda (Instituto de Física, Universidad Nacional Autónoma de México - UNAM, Mexico City, Mexico D.F., Mexico)

Stable acoustical tweezers and particle manipulation using vortex beams have recently been accomplished in air [Nature Commun. 6: 8661 (2015)] and in water [Phys. Rev. Lett. 116: 024301 (2016)]. In both configurations, a particle is trapped along the propagation axis of a focused vortex, where the acoustic field exhibits a pressure node. More recently, a stable orbital motion of a particle trapped off-axis was achieved by alternating between two counter-rotating vortices [*Phys. Rev. Lett.* 120: 044301 (2018)]. However, this kind of orbital motion had not been possible so far with a steady helical field having a single angular momentum state. In this work, we present the first experimental evidence of the simultaneous 3D trapping and stable orbiting of millimeter-sized particles in air due to angular momentum transfer in a standing wave trap created by the interference of two vortices. In contrast with previous work, the two vortices have the same helicity with respect to the laboratory reference frame, and thus the angular momentum of this field is well-defined. A description of the particle kinematics as a function of different control parameters is presented, along with a comparison between experimental measurements of the field and theoretical simulations.

3:15

4pPA7. Acoustic manipulation and actuation of bubbles in complex environments: Beyond the Bjerknes force . Diego Baresch (Dept. of Chemical Eng., Imperial College London, 4, Pl. Jussieu, Paris 75005, France, diego.baresch@upmc.fr) and Valeria Garbin (Dept. of Chemical Eng., Imperial College London, London, United Kingdom)

Micron-sized gas bubbles are notoriously difficult to isolate, handle and remotely control. Their large buoyancy in common liquids will usually force them to rise and burst at any gas/liquid interface or remain trapped against a solid boundary until dissolution. While bubble stability issues against dissolution have found numerous practical workarounds, the challenge remains at isolating and maneuvering a single bubble in free space to, for instance, perform precise single bubble dynamics experiments with applied ultrasound or to use them as active carriers for a specific payload deliverable on demand. Here we demonstrate that single-beam acoustical tweezers [D. Baresch *et al,* Phys. Rev. Lett., **116,** (2016)] can trap and manipulate in 3D a single bubble with the radiation pressure of helicoidal ultrasonic beams. Contrary to the situation where bubbles are trapped in the antinodes of a standing wave, the trapping vortex beam does not require oscillating volume changes of the bubble to generate a force, *i.e*., the trapping mechanisms cannot be explained in terms of Bjerknes forces. Viscous boundary layer effects and large bubble stability will be discussed. Opportunities for our manipulation technique to probe the high speed dynamics and rheology of particle-laden armored bubbles will also be presented [V. Poulichet *et al.,* PNAS, **112,** (2015)].

3:35

4pPA8. Three-dimensional acoustical radiation forces in the view of momentum conservation. Likun Zhang (Univ. of MS, 145 Hill Dr., Oxford, MS 38677, zhang@olemiss.edu)

Analysis of acoustic radiation forces is essential for tailoring parameters for desired manipulations or for interpreting special force conditions. Previous theoretical derivation for acoustic radiation forces was from integration of stress tensor of the total fields. Given that the force is a second-order effect, interactions between the incident and scattered fields or between different beams contribute to the forces and complicate the analysis. Here the three-dimensional forces exerted by an arbitrary field is presented in the framework of momentum conservation where force superposition is possible for flexible analysis. The force is related to momentum transfers by ingoing and outgoing spherical waves and to momentum transfers associated with extinction and scattering. Superposition of forces based on orthogonality of momentum and insight into force dependence on beam and scattering parameters for special manipulations are addressed.

THURSDAY AFTERNOON, 8 NOVEMBER 2018 SALON B (VCC), 1:00 P.M. TO 2:55 P.M.

Session 4pPP

Psychological and Physiological Acoustics: Acoustics Outreach: Linking Physiology and Behavior for Future Collaborations II

Amanda Lauer, Cochair
Otolaryngology-HNS, Johns Hopkins University School of Medicine, 515 Traylor, 720 Rutland Ave., Baltimore, MD 21205

Anna C. Diedesch, Cochair
Communication Sciences & Disorders, Western Washington University, 516 High St., MS 9171, Bellingham, WA 98225

Invited Papers

1:00

4pPP1. Insights from individual differences: Uncovering the code for frequency modulation. Kelly L. Whiteford, Heather A. Kreft, and Andrew J. Oxenham (Psych., Univ. of Minnesota, 75 East River Parkway, Minneapolis, MN 55455, whit1945@umn.edu)

Perceiving changes in frequency (frequency modulation; FM) and amplitude (amplitude modulation; AM) are essential for human and animal communication. Our goal was to utilize individual differences in sensitivity for FM and AM to better understand how FM is represented in the auditory system. Previous studies suggest that FM of low-frequency carriers (f_cs < ~4 kHz) at slow modulation rates (f_m < 10 Hz) is coded by precise, phase-locked spike times in the auditory nerve (time code). The same f_cs at faster rates may utilize a tonotopic (place) code, whereby FM is converted to AM through cochlear filtering. We measured FM and AM detection for a low f_c at slow (f_m = 1 Hz) and fast (f_m = 20 Hz) rates across three large groups: Young, normal-hearing (NH) listeners (n = 100), NH listeners varying in age (n = 85), and listeners varying in degree of sensorineural hearing loss (SNHL; n = 49). Results from all groups show high multicollinearity amongst FM and AM tasks. Data from SNHL but not NH listeners show strong correlations between the fidelity of cochlear place coding (frequency selectivity) and FM detection at both rates. Overall, the evidence suggests a unitary place code for FM. [Work supported by NIH Grant No. R01 DC005216.]

1:20

4pPP2. Tinnitus related to physiological changes in the auditory nerve in chronically noise-exposed mice. Laurel A. Screven, Kali Burke (Psych., Univ. at Buffalo, B80 Park Hall, Buffalo, NY 14260, laurelsc@buffalo.edu), Amanda Lauer (Otolaryngol. HNS, Johns Hopkins Univ. School of Medicine, Baltimore, MD), Matthew A. Xu-Friedman (Biological Sci., Univ. at Buffalo, Buffalo, NY), and Micheal L. Dent (Psych., Univ. at Buffalo, Buffalo, NY)

Tinnitus is a pervasive auditory dysfunction affecting up to 10% of the adult population. The perception of ringing or hissing in the absence of a physical stimulus in one or both ears can be caused by acoustic trauma and other factors. Mice are a commonly used model for auditory disorders in humans, although the behavioral examination of tinnitus in mice has primarily been limited to reflexive measures involving inhibition of the acoustic startle by gaps in noise. Using an identification paradigm, we behaviorally tested whether mice show symptoms of tinnitus following long-term moderate noise exposure. Tinnitus was demonstrated by a shift in categorizing silence as narrowband noise. This experiment demonstrated that tinnitus can be induced in mice using noise exposure, similar to that caused by salicylate. Physiological and anatomical experiments reveal synaptic changes including response facilitation, increased reliability, increased synaptic terminal area, and increased number of release sites in mice housed in chronic background noise compared to standard-housed controls. These mice show normal auditory brainstem responses and a normal complement of peripheral auditory nerve synapses. This combination of behavioral and physiological methodologies enables researchers to examine both the behavioral manifestations and the underlying physiological mechanisms of tinnitus.

1:40

4pPP3. Neuronal frequency selectivity in the inferior colliculus and cochlear nucleus of the awake behaving macaque monkey. Jane A. Burton (Neurosci., Vanderbilt Univ., 111 21st Ave. S, 301 Wilson Hall, Nashville, TN 37240, jane.ann.burton@gmail.com) and Ramnarayan Ramachandran (Hearing and Speech Sci., Vanderbilt Univ., Nashville, TN)

Frequency selectivity relates to the ability to process complex signals and can be measured through auditory filters. Behavioral filters show broader tuning compared to cochlear and auditory nerve fiber tuning. To test whether filters evolve across the auditory pathway or if they are established in the periphery, we estimated neural filters in the cochlear nucleus (CN) and inferior colliculus (IC) and compared with simultaneously measured behavioral filters in macaques. Three macaques were trained to detect tones (signal = unit characteristic frequency (CF)) in spectrally notched maskers of varying width while single unit responses were recorded in the CN and IC. Filter shapes and bandwidths were estimated from the masked thresholds using the rounded exponential fit. Behavioral and neural filters increased in bandwidth with increasing CF. Behavioral and neural bandwidths were significantly correlated and not significantly different from each other for the CN and IC. Neural filter bandwidths were variable across units and structures, possibly reflecting heterogeneity of neuronal encoding strategies. These findings support a model in which behavioral frequency selectivity is established early in the auditory pathway. These data form the baseline for ongoing studies of macaques with noise-induced hearing loss and future studies of emerging hearing loss therapeutics.

1:55

4pPP4. Relations between speech perception in noise, high-frequency audiometry, and physiological measures of cochlear synaptopathy. Hannah Guest, Kevin Munro, and Christopher J. Plack (Manchester Ctr. for Audiol. and Deafness, Univ. of Manchester, Ellen Wilkinson Bldg., Oxford Rd., Manchester, Greater Manchester M13 9PL, United Kingdom, hannah.guest@manchester.ac.uk)

Cochlear synaptopathy, a loss of synapses between inner hair cells and auditory nerve fibers, is associated with age and noise exposure in animal models. However, the functional consequences of synaptopathy for humans are unclear. We pooled data from two recent studies to answer the question: are the common physiological measures of cochlear synaptopathy related to speech-perception-in-noise (SPiN) performance? Eighty-three audiometrically normal participants (ages 18–39) took part. Measures of synaptopathy were as follows: auditory brainstem response (ABR) wave I amplitude (102 dB peSPL click); ABR wave I:V amplitude ratio; envelope following response (EFR) amplitude (4 kHz carrier, 105 Hz modulation frequency); EFR amplitude growth with stimulus modulation depth; and middle ear muscle reflex threshold (1–4 kHz elicitors). We also conducted extended high-frequency (EHF) audiometry (10 and 14 kHz), suggested as a marker for synaptopathy in lower frequency regions. SPiN performance was assessed using the coordinate response measure with spatial maskers. None of the physiological measures of synaptopathy correlated significantly with SPiN. There was a significant correlation between EHF thresholds and SPiN, although it is unclear whether this is due to a direct relation between EHF hearing and SPiN, or whether elevated EHF thresholds are a marker for hidden damage at lower frequencies.

2:10

4pPP5. Olivocochlear system plasticity in response to chronic noise. Dillan F. Villavisanis, Katrina M. Schrode, Hamad A. Javaid (Dept. of Otolaryngol. - Head & Neck Surgery, Ctr. for Hearing and Balance, Johns Hopkins Univ. School of Medicine, 720 Rutland Ave., Baltimore, MD 21218, dvillav1@jhu.edu), Michael Muniak (Div. of Neurosci., Garvan Inst. of Medical Res., Sydney, NSW, Australia), Omobolade Odedoyin (Dept. of Otolaryngol. - Head & Neck Surgery, Ctr. for Hearing and Balance, Johns Hopkins Univ. School of Medicine, Baltimore, MD), Victoria Gellatly, Matthew A. Xu-Friedman (Dept. of Biological Sci., Univ. at Buffalo, Buffalo, NY), and Amanda Lauer (Dept. of Otolaryngol. - Head & Neck Surgery, Ctr. for Hearing and Balance, Johns Hopkins Univ. School of Medicine, Baltimore, MD)

Plasticity in the medial and lateral olivocochlear (MOC and LOC) systems in response to chronic exposure to background noise was investigated. Three different age-groups of young mice were exposed to the same chronic moderate noise conditions with either immediate tissue harvest or harvest following restoration to quiet conditions, with age matched controls. Cochleae were dissected and standard immunohistochemistry protocols were used to label hair cells with antibodies against myosin 6 and olivocochlear synaptic terminals with synaptic vesicle protein 2 (SV2). Specimens were imaged using confocal microscopy, and density of SV2 labeling was quantified. There was no statistically significant difference in MOC SV2 density between mice raised in noise and age matched controls for any group. However, there was a statistically significant increase in LOC SV2 density for adult mice raised in noise, particularly at higher frequency regions. This could suggest a protective upregulation of the efferent system against chronic moderate noise exposure. The increase in LOC innervation persisted for juvenile mice raised in noise with subsequent restoration to quiet conditions. These data suggest that the LOC system demonstrates sound-dependent plasticity, but that synaptic morphology may be altered for a substantial time period after exposure to noise ceases.

2:25

4pPP6. Reliability and interrelations of seven proxy measures of cochlear synaptopathy. Christopher J. Plack (Manchester Ctr. for Audiol. and Deafness, Univ. of Manchester, Ellen Wilkinson Bldg., Oxford Rd., Manchester M13 9PL, United Kingdom, chris.plack@manchester.ac.uk), Hannah Guest, and Kevin Munro (Manchester Ctr. for Audiol. and Deafness, Univ. of Manchester, Manchester, Greater Manchester, United Kingdom)

Investigations of cochlear synaptopathy in living humans rely on proxy measures of auditory nerve function. Numerous procedures have been developed, typically based on the auditory brainstem response (ABR), envelope-following response (EFR), or middle-ear muscle reflex (MEMR). Some metrics correlate with synaptic survival in animal models, but translation between species is not straightforward; measurements in humans likely reflect greater error and greater variability from non-synaptopathic sources. The present study assessed the reliability of seven measures, as well as testing for correlations between them. Thirty-one normally hearing young women underwent repeated measurements of ABR wave I amplitude, ABR wave I growth with level, ABR wave V latency shift in noise, EFR amplitude, EFR growth with stimulus modulation depth, MEMR threshold, and an MEMR difference measure. Intraclass correlation coefficients indicated good-to-excellent reliability for the raw ABR and EFR amplitudes, and for both MEMR measures. The ABR and EFR difference measures exhibited poor-to-moderate reliability. No significant correlations, nor any consistent trends, were observed between measures, providing no indication that the between-subject variability in responses are due to the same underlying physiological processes. Findings suggest that proxy measures of cochlear synaptopathy should be regarded with caution, at least when employed in young, normally hearing adults.

2:40

4pPP7. Behavioral and physiological measures of auditory processing in individuals with autism spectrum disorder. Bonnie K. Lau (Univ. of Washington, 1715 NE Columbia Rd., Box 357988, Seattle, WA 98195, blau@uw.edu), Ross K. Maddox (Univ. of Rochester, Rochester, NY), and Adrian K. C. Lee (Univ. of Washington, Seattle, WA)

Sensory processing abnormalities are a hallmark of autism spectrum disorder (ASD). In the auditory domain, hyper- and hypo-sensitivity to sound, reduced orientation to speech, and difficulty listening in noise are commonly reported. However, the etiology of these auditory processing abnormalities is poorly understood. In this preliminary study, multi-talker speech perception thresholds, otoacoustic emissions, electrophysiological responses, as well as standardized measures of cognition, language, and adaptive function were measured in each individual subject with ASD and their age-matched controls. Speech perception was assessed by estimating target-to-masker ratios at 50% correct for speech targets (0° azimuth) presented with two spatially separated (±45° azimuth) simultaneous speech maskers. The electrophysiological battery used to characterize the transmission and representation of sound included: (1) a click-evoked suprathreshold auditory brainstem response, (2) an envelope following response recorded to a 400-ms-long 4 kHz pure tone carrier amplitude modulated at 100 Hz at two modulations depths (0 and -6 dB) and (3) a binaural potential evoked by an interaural phase difference embedded in an amplitude-modulated carrier tone. Together, these measures provide a behavioral and physiological assay of auditory function and show the potential to further our understanding of the mechanisms underlying auditory abnormalities in individuals with ASD.

THURSDAY AFTERNOON, 8 NOVEMBER 2018 — SALON A (VCC), 1:15 P.M. TO 2:35 P.M.

Session 4pSCa

Speech Communication: Phonetics of Under-Documented Languages I

Benjamin V. Tucker, Cochair
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Richard Wright, Cochair
Linguistics, University of Washington, Box 352425, Seattle, WA 98195-2425

Chair's Introduction—1:15

Invited Papers

1:20

4pSCa1. Phonetic documentation in the literature: Coverage rates for topics and languages. D. H. Whalen (Haskins Labs., 300 George St. Ste. 900, New Haven, CT 06511, whalen@haskins.yale.edu), Christian DiCanio (Univ. at Buffalo, Buffalo, NY), and Rikker Dockum (Yale Univ., New Haven, CT)

With the introduction of automatic methods for examining both the acoustic and articulatory aspects of speech, our knowledge of the phonetic details of well-studied languages has grown substantially. Research on the phonetics of under-documented languages has expanded, but the breadth of coverage remains a challenge. What topics within the sound systems of these languages should be documented? With an ultimate aim of a reasonably complete typological assessment, should researchers focus on particular language families or phenomena and exclude others? Here, a survey of several hundred articles was undertaken, examining three major venues in the phonetics literature: *Journal of Phonetics*, *Journal of the International Phonetic Association*, and publications from the UCLA Languages of the World project (which appeared in various venues). 20 features were found to be addressed in many studies, and a rough guide to the coverage for a language can be seen in the total number of features reported in a study. Coverage has increased over time, but most studies still address a small percentage of the phonetic substance of the target language. Coverage across language family and region also increased but is still skewed toward Indo-European languages. Future directions and recommendations will be presented.

1:35

4pSCa2. Intervocalic consonant voicing vs. lengthening in two Athabaskan languages. Sharon L. Hargus (Dept. of Linguist, Univ. of Washington Seattle, Box 352425, Seattle, WA 98195-2425, sharon@uw.edu)

For over a century, various authors have described intervocalic consonants in Athabaskan languages as being long (e.g., Sapir 1914 on Chasta Costa, Cook 2004 on Dene Sųłiné (Chipewyan), McDonough and Ladefoged 1993 on Navajo). Such descriptions, even if instrumental, have generally not been very detailed, and have not adequately investigated the factors which may influence lengthening, such as stress and position in word. In this presentation, I first present results of an investigation of effects of position and stress on the duration of consonants with internal cues to duration (fricatives and consonantal sonorants) in two Athabaskan languages, Deg Xinag and Kwadacha Tsek'ene. Results show that intervocalic consonants are longer before stressed than unstressed syllables. Next, I present results of an instrumental study of [t] which shows that a competing and antagonistic process to intervocalic lengthening, intervocalic voicing, is also found in both languages. Intervocalic stops are voiced through approximately 80% of their closure duration before unstressed vowels in both languages. Intervocalic voicing of stops has been reported for Tahltan (Bob 1997) and Dene Sųłiné (McDonough and Wood 2008) but generally not for other Athabaskan languages except for those in which nasals have evolved into voiced obstruents (e.g., Jicarilla Apache, Tuttle 2000).

1:50

4pSCa3. Kuki-Chin languages in Indiana: Investigating typologically rare sounds in a developing community of collaboration.
Kelly Berkson, Samson A. Lotven, James Wamsley, Zai Sung, Peng H. Thang, Thomas Thawngza, Kenneth de Jong, Sandra Kuebler, and Steven M. Lulich (Indiana Univ., 1021 E. Third St., Dept. of Linguist - Mem 322E, Bloomington, IN 47405, kberkson@indiana.edu)

More than 20,000 Burmese refugees live in Indiana at the time of this writing. Most are originally from Chin State in western Myanmar and speak under-documented Tibeto-Burman languages from the Kuki-Chin branch. For the most widely spoken of them—Hakha Chin, also known as Laiholh or Hakha Lai—syntactic and morphological work exists but basic acoustic work is minimal. Others are completely undocumented: Zophei is spoken by about 4,000 people in Indiana and has to the best of our knowledge been mentioned but not described. In this talk, we describe a burgeoning collaboration between speech scientists and speakers of Kuki-Chin languages in Central Indiana. Our long-term communal goal is to develop resources ranging from the linguistic (e.g., documentary, analytical) to the practical (e.g., speech technology for use in emergency rooms, literacy materials). Phonetic investigation is a necessary precursor to much of this work, in part because these languages contain typologically unusual sounds (voiceless laterals, rhotics, and nasals; a 5-way coronal stop contrast involving dental and alveolar obstruents; voiced and voiceless laterally-released obstruents). Herein, we present the results of several phonetic investigations which will support future development of both theoretical and practical technological resources.

2:05

4pSCa4. A cross-linguistic study of phonetic correlates of metrical structure in under-documented languages.
Matthew K. Gordon, Ayla Applebaum (Dept. of Linguist, UC Santa Barbara, Santa Barbara, CA 93106, mgordon@linguistics.ucsb.edu), Thiago Chacon (Univ. of Brasilia, Brasilia, Brazil), Jack Martin (College of William & Mary, Williamsburg, VA), and Françoise Rose (Laboratoire Dynamique Du Langage, Université Lyon2/CNRS, Lyon, France)

Although languages with rhythmic stress have been extensively described and analyzed in the phonology literature, there has been little phonetic verification of the metrical structures suggested by these stress patterns. This paper presents results of a cross-linguistic study of the acoustic correlates of foot structure in several under-documented languages with morphologically complex words providing the necessary backdrop for the realization of rhythmic metrical structure. Various potential acoustic exponents of metrical structure are considered, including duration, F0, intensity, and formant frequencies. The targeted languages include Muskogean languages of the United States (Koasati, Muskogee, and Chickasaw), Circassian languages spoken primarily in Turkey and Russia (Kabardian and Adyghe), the Tukanoan language Kubeo of Brazil and Colombia, and the Arawak language Mojeño Trinitario of Bolivia. The languages differ in their foot templates (trochaic vs. iambic), the direction of their metrical parse (left-to-right vs. right-to-left), their degree of metrical rhythm, and whether they also possess lexical tone or not. Results suggest considerable diversity in the acoustic manifestations of metrical structure (including the possibility of lack of rhythmic feet even in long words) and in the relationship between the word-level metrical system and other prosodic features, including intonation and (in tonal languages) lexical tone.

2:20

4pSCa5. Obstruent and rhotic contrasts in Adnyamathanha, a language of South Australia.
Andrew Butcher (Speech Pathol. & Audiol., Flinders Univ. of South Australia, Flinders University, GPO Box 2100, Adelaide, SA 5001, Australia, andy.butcher@flinders.edu.au) and John McEntee (Independent Researcher, Glenelg South, SA, Australia)

Adnyamathanha is one of the Thura-Yura languages, spoken in the northern Flinders Ranges of South Australia. It has a fairly complex consonant system with six places of articulation (including four coronals). Through a combination of traditional phonological analysis and acoustic phonetic measurement, we attempt to throw some light on one aspect of this complexity. We show that there is a contrast between two series of obstruents in intervocalic position, which sets Adnyamathanha apart from most other languages of the region—and of the Pama-Nyungan language family as a whole. The main phonetic correlates of the contrast are closure duration and presence versus absence of glottal pulsing during the closure. Voiceless obstruents are consistently realised as long stops, whereas their voiced counterparts, though always shorter, vary in duration and manner of articulation, depending on place of articulation. We analyse the voiced labial fricative as the voiced equivalent of the voiceless (bi)labial stop and we analyse the alveolar tap and the retroflex flap as voiced cognates of the voiceless alveolar and retroflex stops respectively. Thus, although we recognise the existence of four phonetically distinct rhotic sounds, we assign only the alveolar trill and the retroflex glide to the phonological category of rhotics.

Session 4pSCb

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

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

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

All posters will be on display from 3:00 p.m. to 5:00 p.m. To allow authors an opportunity to view other posters in their session, authors of odd-numbered papers will be at their posters from 3:00 p.m. to 4:00 p.m. and authors of even-numbered papers will be at their posters from 4:00 p.m. to 5:00 p.m.

Contributed Papers

4pSCb1. The rhythm of Hul'q'umi'num'. Mackenzie Marshall and Sonya Bird (Dept. of Linguist, Univ. of Victoria, 3800 Finnerty Rd, Victoria, BC V8P 5C2, Canada, mackenzieomarshall@gmail.com)

This research explored the linguistic rhythm of Hul'q'umi'num', a dialect of the Coast Salish Hul'q'umi'num'-Halq'eméylem-hən̓q̓əmin̓əm̓ language, and initiated phonetic documentation of it. Rhythm was analyzed from an audio file of an Elder telling a story. The story was segmented and phonetically transcribed using acoustic analysis software (Praat), and rhythm was measured based on the segmentation. Rhythm metrics demonstrated that consonantal intervals of Hul'q'umi'num' patterned like no other documented language (according to ΔC and VarcoC). In terms of vocalic intervals (%V, ΔV, and VarcoV), Hul'q'umi'num' patterned in the same rhythmic category as English ("stress-timed"). Interestingly, several differences emerged between the segmentation-based phonetic transcription and the transcription provided by language experts, such as consonant cluster elisions, loss of glottal stops, and vowel alternations. Further investigation of their systematicity and effects on Hul'q'umi'num' rhythm as a whole is needed to understand what components of the language give it its unique rhythm.

4pSCb2. Pitch realization of post-focus components in Chongming Chinese. Yike Yang, Si Chen (Dept. of Chinese and Bilingual Studies, The Hong Kong Polytechnic Univ., AG511, Kowloon N/A, Hong Kong, yi-ke.yang@connect.polyu.hk), and Kechun Li (The Univ. of Hong Kong, Hong Kong, Hong Kong)

Prosodic focus makes use of pitch to highlight part of an utterance. It has been generally found that a focused component is realized with an expanded pitch range, and there is evidence suggesting that a post-focus component may be associated with a reduced or compressed pitch range, the phenomenon of which is further coined as post-focus compression (PFC). However, the presence or absence of PFC seems to diverge even within the same language family, and what is more interesting is the potential interaction between focus and lexical tones in tone languages. The current project thus aims to investigate whether PFC is present and how post-focus pitch is realized in Chongming Chinese, an under-documented language with eight tones. Specifically, we conducted a production experiment in which 20 target words varying in consonants, vowels and tones were selected, four focus conditions (no focus, pre-focus, on focus and post-focus) were designed, and four tonal contexts with different proceeding and following syllables were manipulated. Linear mixed-effects models were fitted to examine the effects of these variables on the realization of pitch contours in the post-focus components and also tackle the interactions among them. Our findings contribute novel data to the prosodic typology literature.

4pSCb3. Dagaare [a] is not neutral to ATR harmony. Avery Ozburn, Samuel Akinbo, Alexander Angsongna, Murray Schellenberg, and Douglas Pulleyblank (Univ. of Br. Columbia, 2613 West Mall, Vancouver, BC, Vancouver, BC V6T 1Z4, Canada, aozburn@gmail.com)

Bodomo (1997) describes Dagaare (Gur; Ghana) as having a single low vowel, [a], which is neutral to ATR harmony. This paper presents acoustic data from a study of Dagaare <a> which is inconsistent with this description. A list of sentences was elicited from five native speakers of Dagaare. Each sentence contained <a> in one of four verbal particles situated in one of four contexts: ATR _ ATR, ATR _ RTR, RTR _ ATR, and RTR _ RTR. Formants of the low vowel were measured and compared across contexts. Results showed a substantial, significant difference in F1 values and a smaller but still significant difference in F2 values in contexts where <a> is followed by an ATR word compared to when it is followed by an RTR word. All speakers and all particles showed the same pattern. We conclude that, contrary to previous claims, the Dagaare low vowel is not neutral to harmony, but rather has acoustically distinct variants in RTR versus ATR contexts. Bodomo, A. (1997). *The structure of Dagaare.* California: CSLI publications. [Funded by SSHRC.]

4pSCb4. Phonetic structure of Intonational prominence in 3 Dene/Athabasakn (ISO den) languages. Joyce McDonough (Dept. of Linguist, Univ. of Rochester, Rochester, NY 14627, joyce.mcdonough@rochester.edu)

In this paper, we examine the prosodic structure of Y/N Q's and focus constructions in three related Dene/Athabaskan tone languages spoken in North American: Navajo, Dene Suɬ̨íné, and North Slavey (Deline), known for their complex polysynthesis, representing a geographically widespread language family, from the American Southwest (Navajo), to northern Alberta (Dene Suɬ̨ine) to the arctic (North Slavey). These languages share a strikingly similar phonetic inventory and morphological structure. One interesting feature is the phonotactic structure of the morphology—the monosyllabic verb stem is the location of phonemic contrasts: outside the stem the contrasts are severely reduced, making the stems phonemically and phonetically prominent. The stems are the rightmost elements in the verbal complex and in a sentence (verb final). Navajo has been argued to lack any kind of intonational prominence, but the northern languages show more typical prominence marking. In this study we lay out the difference associated to the demarcation of boundaries and prominence: duration and the acoustic correlates of pitch accent events associated with focus and YN/Q's and boundary marking. Data is taken from conversation games models on map tasks. Findings indicate while phonetically and morphologically similar great variety is found in the prosody.

4pSCb5. Neutralization of underlying vs. derived "unnatural" palatals in Xhosa. Aaron Braver (Texas Tech Univ., P.O. Box 43091, Lubbock, TX 79409-3091, aaron@aaronbraver.com)

Xhosa (Bantu) has an "unnatural" process of palatalization—contrary to typological expectations, it is triggered by [w], but not [i] or [j], and applies only to labials. For example, [m]→[ɲ] as in uku-lum-à to bite' ~ uku-luɲ-w-a 'to be bitten', and [mb]→[ⁿd͡ʒ] as in uku-ɬamb-a 'to wash' ~ uku-ɬaⁿd͡ʒ-w-a 'to be washed'. This paper compares underlying and derived palatals to determine whether the palatal/non-palatal contrast is neutralized (in)completely in the derivational context. 40 Xhosa nonce verbs, ending in segments which undergo palatalization, were created to be read by 6 native speakers. Speakers read these forms and were then asked to produce forms with the passive -w suffix, thus triggering palatalization. F2 was measured at boundaries and ten ms. into the vowels preceding and following the palatalized segment. F2 at ten ms into the following vowel differed on average by 167.99 Hz between underlying and derived palatals, trending towards, but not reaching significance ($t = 1.94$, $p = 0.054$), while F2 slope showed no clear trend. While Zsiga (1995) proposes that English palatalization can be complete or gradient, this is the first study of palatalization in the context of incomplete neutralization, and the first study to look at incomplete neutralization in Bantu.

4pSCb6. The special nature of Australian phonologies: Why auditory constraints on the sound systems of human languages are not universal. Andrew Butcher (Speech Pathol. & Audiol., Flinders Univ., GPO Box 2100, Adelaide, SA 5001, Australia, andy.butcher@flinders.edu.au)

The phonological systems of human languages are constrained by what are often assumed to be universal properties of human auditory perception. However, the atypical phonologies found in many hearing-impaired speakers indicate that such constraints also operate at an individual level. The phonology of a child with chronic *otitis media*, for example, may lack a voicing distinction or sometimes have no fricatives. So, if a large group of speakers in a speech community operates with an atypical auditory system over many generations, then the phonology of the language(s) spoken by such a community might also over time be influenced by the particular properties of that common auditory system. Over half of the Australian Aboriginal population develop chronic *otitis media* with effusion in infancy and 50–70% of Aboriginal children have a significant hearing loss at both ends of the frequency range. Most Australian languages have phonologies which are atypical in world terms, having no voicing distinction and no fricatives or affricates, but an unusually large number of places of articulation. Acoustically, the sound systems of Australian languages appear to be a very good match for the hearing profiles of large numbers of their speakers. This paper reviews the evidence for a connection.

4pSCb7. Acoustic properties of singleton and geminate ejective stops in Tsova-Tush. Bryn G. Hauk (Linguist, Univ. of Hawaii at Manoa, 713 Hausten St. #2, Honolulu, HI 96826, bhauk@hawaii.edu)

Tsova-Tush, also known as Batsbi, is an underdescribed Northeast Caucasian language spoken by a few hundred people in Zemo Alvani, Georgia. The Tsova-Tush phoneme inventory includes four geminate stops that contrast with singletons at the same place of articulation: /tʰː t'ː qʰː q'ː/. The existence of geminate ejective stops is particularly interesting, as such phonemes are cross-linguistically rare, having been reported in only 13 of the 2,155 phoneme inventories sampled by PHOIBLE (http://phoible.org/). The present study characterizes these stops in Tsova-Tush based on high-quality audio recordings of four Tsova-Tush speakers producing a list of 65 target words in a carrier sentence. The following measures were statistically compared: closure duration, VOT, duration of the preceding vowel, and f_0 and H1*-H2* in the following vowel. Preliminary results from three speakers suggest that ejectives are characterized by shorter VOT, a shorter preceding vowel, and a difference in both f_0 and H1*-H2* in the following vowel, with marked interspeaker variation in the latter measures. This study provides the first detailed description of this typologically interesting two-by-two contrast (singleton aspirates, geminate aspirates, singleton ejectives, geminate ejectives) at two places of articulation (dental and uvular) in an underdescribed language.

4pSCb8. An acoustic comparison of /θ/ and /s/ in Comox-Sliammon. Gloria Mellesmoen (Univ. of Br. Columbia, Box #346 - 6335 Thunderbird Crescent, Vancouver, BC V6T 2G9, Canada, gloria.mellesmoen@alumni.ubc.ca)

Though some Coast Salish languages have innovated /θ/, a typologically rare segment, the only study of Salish fricatives describes Montana Salish, an Interior language without /θ/ [Gordon *et al.* (2002). A cross-linguistic acoustic study of voiceless fricatives. *JIPA, 32*(2), 141-174.]. In addition to being typologically noteworthy, /θ/ is also interesting within Salish, as impressionistic descriptions suggest articulatory similarity and perceptual ambiguity between /θ/ and /s/. This is found in Halkomelem, Northern Straits, and Comox-Sliammon. Motivated by a gap in documentation and reported ambiguity, this paper is an acoustic study of /θ/ and /s/ in Comox-Sliammon. Following the methodology of Reidy [(2016). Spectral dynamics of sibilant fricatives are contrastive and language specific. *JASA, 140*(4), 2518–2529.], PeakERB$_N$ trajectories are compared for /θ/ and /s/ across four fluent speakers of Comox-Sliammon. The results suggest that the fricatives are acoustically distinct, though there is considerable inter-speaker and intra-speaker variability for /θ/. Lack of overlap for three of four speakers suggests that the source of the reported ambiguity may be L1 English speaker perception, rather than the realization of Mainland Comox fricatives. The high level of variation suggests that /θ/ may be a recent and unstable innovation, supporting the reconstruction of a Proto-Comox [s]-like form.

4pSCb9. Pausing as a prosodic correlate of speech units in St'át'imcets (Lillooet Salish). Marion Caldecott, Ewa Czaykowska-Higgins, Janet Leonard, and John Lyon (Office of Linguist, Univ. of Victoria, PO Box 1700 STN CSC, Victoria, BC V8W 2Y2, Canada, mariongc@uvic.ca)

The reliability of pausing as a correlate to prosodic, syntactic and discourse units has been debated in commonly-studied languages (outlined, for instance, in Krivokapic 2007). Preliminary research in Nxaʔamxčín (Interior Salish) (Caldecott & Czaykowska-Higgins 2012) has shown that pausing could be a more reliable acoustic correlate of these structures than pitch, in line with previous research indicating that Salish languages do not exhibit a strong reliance on pitch to mark information structure (Caldecott 2017; Caldecott & Czaykowska-Higgins 2012; Davis 2012; Koch 2008, 2011). The current study further tests this hypothesis by examining the occurrence and duration of pauses in a 9.5-minute spontaneous narrative by a fluent speaker of St'át'imcets (Interior Salish). Pause distribution with respect to syntactic and prosodic boundaries is analysed. Preliminary analysis suggests that St'át'imcets follows previous research on spontaneous narratives in some respects but not others. Very long pauses (<5.5 s) mark major thematic shifts (Oliveira 2000) and most pauses (70%) occur clause-finally (Henderson, Goldman-Eisler & Skarbek 1966). Most clause-medial pauses occurred between a determiner and noun, following Goldman-Eisler (1968), but contra Gee & Grosjean (1983). A t-test indicates that clause-final vs. clause medial pauses are not significantly different in duration ($p>.05$) (contra Goldman-Eisler 1972).

4pSCb10. An acoustic and articulatory investigation of the Mauritian vowel inventory. Samantha Myers (Lingustics, Indiana Univ., Bloomington, IN), Fabiola Henri (Univ. of Kentucky, Lexington, KY), and Kelly Berkson (Lingustics, Indiana Univ., 1021 E. Third St., Dept. of Linguist - Mem 322E, Bloomington, IN 47405, kberkson@indiana.edu)

While there have been a decent amount of morphosyntactic and sociolinguistic studies on Mauritian, a "French-based" creole spoken by most of the population of Mauritius (e.g. Baker 1972, Alleesaib 2012, Henri 2010, Miller 2015, Syea 2013), phonetic or phonological description on the language hardly exists. Pudaruth (1993) proposes a phoneme inventory consisting of 19 consonants and 8 vowels, intuitively noting that rhotics preceding vowels are weakened, almost unpronounced. In the present work, we empirically investigate the vowel inventory of Mauritian using acoustic and articulatory data. In addition to presenting basic acoustic measures (e.g. vowel spaces), we investigate the rhotic and report acoustic measures which suggest strong variation between the orthographic "a" that precedes "r" and that which precedes all other written consonants. Ultrasound imaging confirms that the pre-rhotic "a" differs from other "a"s articulatorily.

4pSCb11. Acoustics of Tatar vowels: Articulation and vowel-to-vowel coarticulation. Jenna Conklin and Olga Dmitrieva (Linguist, Purdue Univ., 610 Purdue Mall, West Lafayette, IN 47907, jconkli@purdue.edu)

Volga Tatar is a Turkic language spoken by 5 million people in Central Russia for which instrumental acoustic descriptions are lacking. This study uses formant analysis of acoustic recordings from 27 native speakers of Volga Tatar to describe the vowels of Tatar and evaluate the accuracy of previous impressionistic phonetic descriptions. In addition to describing the acoustic vowel space of Tatar, the study uses carefully chosen target words to evaluate vowel-to-vowel coarticulation in height and backness among the Tatar vowels /i, æ, ɑ/. Examining vowel-to-vowel coarticulation in Tatar is of particular theoretical interest due to the presence of vowel harmony in the language. While the majority of native Tatar words are subject to backness, and possibly rounding, harmony, a large class of disharmonic lexical items, mostly from borrowings, provides insight into the coexistence of long-distance phonological vowel assimilation (vowel harmony) and long-distance phonetic vowel assimilation (coarticulation) in the same language. Previous research (Banerjee, Dutta, & S., 2017; Beddor & Yavuz, 1995) suggests that vowel harmony may suppress coarticulation proceeding in the same direction. However, the current results indicate that direction of coarticulation in Tatar is mediated primarily by other factors, such as target and trigger vowel identity.

4pSCb12. Variation in Scottish Gaelic preaspirated stops: Across speech styles and in connected speech. Maya Klein (Linguist and Anthropology, Univ. of Arizona, 1103 e University Blvd., Tucson, AZ 85721, mayaklein@email.arizona.edu)

This paper examines variation in preaspiration in Scottish Gaelic, an endangered language spoken in Scotland. Previous studies (Clayton 2011; Nance & Stuart-Smith 2013) have examined Gaelic preaspiration in careful speech. The present study investigates whether 1. Preaspiration patterns similarly in casual speech as previously attested in careful speech and 2. If variation in preaspiration resembles processes of stop reduction, or sound change in progress. Preaspiration is examined in casual speech across speakers, and across different speech styles within one speaker. Preaspiration is measured for duration, as well as band-pass filtered zero crossing rates, developed by Gordeeva & Scobbie (2010) and adapted for Scottish Gaelic by Nance & Stuart-Smith (2013). Results confirm that variation in preaspiration pattern similarly to previous studies: preaspiration is longer and has higher change in zero crossings for /k/ and in words with a preceding short vowel. This study contributes to the existing literature in that it shows that preaspiration is shortest in words that are phrase medially. This study shows that preaspirated stops are not heavily reduced, and that variation in preaspiration patterns like an ongoing sound change. This study highlights the importance of carefully investigating variation in typologically rare phenomena in endangered languages.

4pSCb13. Perception of focus in Sümi (Tibeto-Burman, India). Amos Teo and Melissa M. Baese-Berk (Linguist, Univ. of Oregon, 1505 Orchard St., Unit 28, Eugene, OR 97403, ateo@uoregon.edu)

This paper investigates the homophony/polysemy between a morphological agentive marker and a contrastive focus marker in Sümi, a Tibeto-Burman language of Northeast India. Both are realized by an enclitic =*no* that attaches to grammatical subjects, but the interpretation of the enclitic varies by clause type: textual analysis shows that =*no* with subjects of transitive clauses is more commonly interpreted as an agentive marker without focus; while =*no* with subjects of verbless and intransitive clauses is more commonly interpreted as a contrastive focus marker (Teo 2018). The present study examines whether transitive subjects in contrastive focus receive any special prosodic marking that is recognizable to native listeners. We answer this question using a ratings task in which listeners were presented with sentences that were produced under different pragmatic conditions to have broad focus over the entire clause or narrow contrastive focus on the subject, i.e., in responses to a question like "Who (out of a set of potential participants) did the action?" The study has implications for understanding the development of focus markers in other languages of the Himalayas, as well as in New Guinea and Australia where homophony/polysemy between agentive or ergative markers and focus markers has been found.

4pSCb14. A vowel space comparison of Tlawngrang Zophei and Lawngtlang Zophei. Samson A. Lotven and Kelly Berkson (Dept. of Linguist, Indiana Univ., 1020 E. Kirkwood Ave., Bloomington, IN 47405-7005, slotven@indiana.edu)

Zophei refers to an undescribed group of Tibeto-Burman languages within the Kuki-Chin family. Originally spoken in the Chin Hills of Western Myanmar, approximately 4,000 Zophei-speaking refugees now live in Central Indiana. No previous research on Zophei exists. The speakers located in Indiana who identify as ethnically Zophei hail from 14 distinct villages, and it is not yet known how many dialects or languages are represented. As part of a larger effort to kick-start a research program on Zophei, the current study compares and contrasts the vowel spaces of two speakers, one from Tlawngrang and one from Lawngtlang. The vowel spaces created for each speaker show some clear differences, especially with regards to the number and distribution of high vowels and diphthongs, indicating that these two areas speak different varieties with markedly different phonologies. For example, where one speaker has an /ui/ diphthong the other speaker r consistently has the front rounded monophthong /y/. This research contributes to our ultimate goal, which is to determining the dialectal make-up of Zophei and to develop a description of the language or languages spoken by the ethnic Zophei population in Indiana.

4pSCb15. F0 of the verbal nasal prefix in Medumba. Yadong Liu (Linguist, Univ. of Br. Columbia, Run Run Shaw Tower, The University of Hong Kong, Hong Kong 000000, Hong Kong, yadong@connect.hku.hk)

Both tone-bearing unit (TBU) and non-TBU syllables can exist in a tonal language [Roberts 2003, *Cahiers voltaïques/Gur Papers* 6, 95]. The verbal nasal prefix bears a high tone in many Bamileke languages [Voorhoeve 1974, *Stud. Afr. Ling.* 5: 205]; in Medumba, however, anecdotal evidence suggests that the nasal prefix may be a non-TBU [Keupdjio 2018, *PC*] in which case it cannot bear a high tone. The present study investigates the F0 contour of the verbal N-prefix and its surrounding TBUs (e.g., vowels in the preceding auxiliary verb and the following main verb) in various contexts in Medumba. Acoustic results show that the F0 contour of the nasal prefix itself is largely determined by the F0 trend of the preceding vowel, and no F0 change is observed at the onset of the nasal prefix, confirming that the verbal N-prefix is a non-TBU in Medumba. However, the nasal prefix appears to induce a raised F0 at the onset of the following TBU, indicating that, although the N-prefix itself does not express tone, it nevertheless "bears" a high tone. Implications for theories of TBUs will be discussed.

4pSCb16. Enriching the understanding of glottalic consonant production: Vertical larynx movement in Hausa ejectives and implosives. Miran Oh, Dani Byrd, Louis Goldstein (Linguist, Univ. of Southern California, 3601 Watt Way, Grace Ford Salvatori 301, Los Angeles, CA 90089, miranoh@usc.edu), and Shrikanth S. Narayanan (Elec. Eng., Univ. of Southern California, Los Angeles, CA)

Ejectives and implosives are understood to have laryngeal activity distinct from their pulmonic counterparts. However, while correlations of vertical larynx movement (VLM) with tone and intraoral pressure are known, little articulatory data exists regarding VLM in glottalic consonants. This study examines the degree and timing of VLM in a Hausa speaker's productions, using real-time MRI of the vocal tract. We use a novel centroid tracking technique developed to quantify VLM. Findings indicate that the onset of VLM is tightly coordinated with oral closure achievement for glottalic stops. For ejectives, larynx raising is faster than in plosives and is especially large for ejective fricatives, which require maintenance of sufficient air flow for turbulence. Additionally, contrary to previous descriptions, implosives and voiced plosives do not differ in degree of larynx lowering. Instead, they contrast in timing of the VLM with the oral closure; larynx lowering starts early, with initiation of the oral constriction gesture, for voiced plosives; whereas lowering begins later, at oral closure achievement, for implosives. Overall these patterns suggest that VLM may be coupled in an anti-phase mode with the oral constriction gesture in both glottalic stops and coupled in-phase with the oral gesture in pulmonic stops. [Work supported by NIH.]

4pSCb17. The production and perception of Ikema geminates. Catherine Ford and Benjamin V. Tucker (Univ. of AB, Unit 204 7111 80 Ave. NW, Edmonton, AB T6B0C7, Canada, cford1@ualberta.ca)

Gemination is largely assumed to be cued through duration. While production and perception studies consistently confirm the importance of duration as a cue, certain phones are not as easily distinguished using duration alone. The current study analyzes geminate and singleton minimal pair production and perception in Ikema, an endangered Japonic language, to determine to what extent other cues may contribute to this distinction. Productions of stop, fricative, affricate, nasal, and rhotic geminates and singletons were used to determine acoustic differences and possible cues to gemination. We analyzed duration for all phones, and spectral moment and vocalic measures for fricatives and affricates. Results indicate that while duration is an important cue, spectral moment measures and vocalic measures are better predictors of affricate geminates and contribute significantly to the acoustic distinction among fricatives. Word-initial stops may be difficult to identify as closure duration is a more significant predictor of gemination than voice onset time. Listener perception was analyzed using a lexical decision task. Results indicate that while Ikema speakers distinguish geminates from singletons in novel situations, non-word perception may not be consistent cross-linguistically due to varying exposure to the concept, and community-internal acceptability of pronunciation variation.

4pSCb18. Comparative analysis of South Korean and North Korean vowels: A pilot study. Jungah Lee, Kaori Idemaru (East Asian Lang. and Literatures, Univ. of Oregon, 1455 Moss St. Unit 306, Graduate Village Apartment, Eugene, OR 97403, jlee27@uoregon.edu), and Charlotte Vaughn (Linguist, Univ. of Oregon, Eugene, OR)

This study investigates cardinal vowels of standard North Korean and South Korean. Prior reports have suggested that North and South Korean vowels have undergone changes after decades of relative isolation. This poster reports a pilot study investigating the ways in which the language standards of North and South Korea are similar and different by examining the speech of newscasters from each country. Acoustic analysis of the speech data suggested that North Korean vowels [ɛ] and [æ] were produced in the higher position relative to the South Korean counterparts, and the back vowels [ʌ] and [o] showed overlapping formant values unlike the South Korean counterparts. The perception experiment suggested that South Korean listeners could not accurately identify the North Korean [ʌ] and [o]. These results indicate that there may be interesting differences across North Korean and South Korean vowels.

4pSCb19. On the non-universality of intonation: Evidence from Triqui. Christian DiCanio and Richard Hatcher (Linguist, Univ. at Buffalo, 601 Baldy Hall, Buffalo, NY 14260, dicanio@haskins.yale.edu)

Languages with large lexical tone inventories typically involve less freedom for suprasegmental properties to be manipulated for pragmatic meaning or phrasal constituency (Connell 2017). However, such languages may still use F_0 to a limited degree for marking information structure or utterance finality (DiCanio *et al.* 2018, Xu 1999). We present results from three field experiments with 11 speakers where we investigated information structure and prosody in Itunyoso Triqui, an Otomanguean language (Mexico). Itunyoso Triqui possesses nine lexical tones (/4, 3, 2, 1, 43, 32, 31, 13, 45/), fixed final stress, and contrastive phonation type. In experiment 1, we examined tone production in words in broad and narrow focus contexts. Words under narrow focus were lengthened slightly (13-14%) but no general effect of focus on F0 levels or contours was found. In experiment 2, we examined tones in utterance non-final and final contexts. Words were lengthened in utterance-final position relative to non-final position, but no F0 differences were found. In experiment 3, we investigated F0 declination in sentences consisting of only level tones and found no F0 change across utterances. The results from these experiments suggest that Itunyoso Triqui does not use F_0 to encode information structure or prosodic boundaries.

4pSCb20. Vowel length distinctions in Plains Cree. Angeliki Athanasopoulou and Darin Flynn (School of Lang., Linguist, Literatures, and Cultures, Univ. of Calgary, Craigie Hall D310, 2500 University Dr. N.W., Calgary, AB T2N1N4, Canada, angeliki@udel.edu)

Plains Cree is a widely-spoken Indigenous language in Canada. Its vowels are traditionally described as contrasting short /i, o, a/ vs. long /iː, oː, aː (eː)/ (Wolfart 1973). Muehlbauer's (2012) acoustic study of speakers born in the early 20th century confirms that duration is a significant cue, but differences in vowel quality were just as significant. We investigate the durational and quality differences in a younger L1 speaker's productions of short and long vowels. Preliminary analysis of 195 vowels shows that the durational difference between the long (102ms) and short vowels (66ms) remains significant on average, but there is greater overlap between the two length categories than previously thought: the range is 40-186ms for long vowels and 15-201ms for short ones. Vowel quality remains significant, too. In addition, we report on the results of a logistic regression analysis that tests the relative importance of duration vs. quality (and their interaction) as acoustic cues to the phonological contrast in the current generation of Plains Cree speakers. Our findings dovetail with descriptions of Eastern varieties, e.g.: "the old long/short distinction in Proto-Cree became (or is still becoming) a tense/lax distinction in East Cree" (Dyck 2011).

4pSCb21. Stop, approximant, and timing slot: The changing faces of the velar stop in Iwaidja. Jason Shaw (Yale Univ., Dow Hall 305, New Haven, CT, underlying.representation@gmail.com), Christopher Carignan (LMU Munich, Munich, Germany), Tonya Agostini (Univ. of Newcastle, Wollongong, NSW, Australia), Robert Mailhammer (Humanities and Commun. Arts / The MARCS Inst., Western Sydney Univ., Penrith, NSW, Australia), Mark Harvey (Univ. of Newcastle, Newcastle, NSW, Australia), and Donald Derrick (Univ. of Canterbury, Christchurch, New Zealand)

Limited access to speakers and incomplete lexical knowledge are common challenges facing phonetic description of under-documented languages. We address these challenges by taking a multi-dimensional approach, seeking to constrain our phonetic description by covariation across acoustic and articulatory parameters. We demonstrate the approach through an analysis of velar consonants in the Australian Aboriginal language Iwaidja. Existing accounts contrast a velar stop /k/ with a velar approximant /ɰ/ in word-medial position (Evans 2009). Converging evidence from ultrasound images of the tongue body and acoustic analysis of intensity data reveal that the posited opposition is not consistent across speakers (N = 4) and lexical items. Unsupervised categorization of the phonetic data indicates two phonetic categories, appropriately labelled as [a] and [ɰ], which do not map consistently to dictionary labels in existing descriptions. We conclude that speaker-specific allophonic variation is the result of an ongoing process of lenition of /k/ between sonorant segments which has not yet phonologized. More broadly, integrating phonetic dimensions revealed categories that were ill-defined on the basis of just acoustic or articulatory measures alone. Depth of analysis, characterized by phonetic multi-dimensionality, may support robust generalization where broad analysis (multiple speakers, large corpora) are impractical or impossible.

4pSCb22. Laryngeal contrasts in first and second language speakers of Hul'q'umi'num'. Maida Percival (Linguist, Univ. of Toronto, 100 St. George St., 4th Fl., Toronto, ON M5S 3G3, Canada, maida.percival@mail.utoronto.ca), Sonya Bird (Linguist, Univ. of Victoria, Victoria, BC, Canada), and Donna Gerdts (Linguist, Simon Fraser Univ., Burnaby, BC, Canada)

This paper investigates laryngeal contrasts in first (L1) and second (L2) language speakers of Hul'q'umi'num', a dialect of Halkomelem (Salish). The language, which lacks a thorough acoustic description of its consonants, is highly endangered with 50-100 native speakers, but has a growing number of L2 speakers. Since the learners come from an English background and are therefore unfamiliar with glottalized consonants, the sounds can prove challenging. This study addresses this issue by examining how L1 and L2 speakers pronounce plain and ejective stops in terms of what acoustic correlates they're using. Tokens of words read in isolation will be

analyzed from L1 and L2 speakers participating in a language course in Duncan, BC. Tokens of both speaker groups will be classified by phoneme and acoustic measures of duration (e.g., voice onset time, closure duration), centre of gravity, and vowel coarticulation (e.g., spectral tilt, F0, rise time) will be made. The results of the two groups will be compared with statistical analysis, and the findings will be used to create guidelines to assess future learners' and teach pronunciation. The findings will also contribute to a broader understanding of how Hul'q'umi'num' consonants fit into voicing and ejective typology.

4pSCb23. The Suzhou Chinese "apical" and "fricative" vowels: Uniformity and idiosyncrasy. Matthew Faytak (Linguist, Univ. of California, Berkeley, 2632 San Pablo Ave., Apt. A, Berkeley, CA 94702, mf@berkeley.edu)

"Fricative" and "apical" vowels have been variously described as syllabic voiced fricatives, syllabic approximants, or high vowels, often based on impression. To better characterize their articulatory and acoustic nature, ultrasound video and acoustic data on the production of the "fricative" and "apical" vowels were collected from 43 speakers of Suzhou Chinese, along with the vowels [i, y, æ, u] and the fricatives [s, ɕ]. Analysis reveals substantial interspeaker variation, largely idiosyncratic, which may contribute to the range of descriptions given to "fricative" and "apical" vowels. This variation, however, obscures the fact that the "fricative" vowels are curiously uniform in tongue posture with the alveolopalatal fricatives for most speakers. The "apical" vowels are consistently produced with a tongue posture similar to the alveolar fricatives and affricates, such as [s], that they obligatorily follow. The "fricative" vowels, on the other hand, are phonotactically freer, and speakers exhibit a wider range of articulatory strategies. Individuals' favored articulations can be situated on a continuum between lamino-postalveolar, or [ɕ]-like, and dorso-postalveolar, akin to an advanced [i]; most speakers favor the lamino-postalveolar strategy. Acoustic analysis suggests that speakers also vary in whether they favor production of fricative noise during both the "fricative" and "apical" vowels.

THURSDAY AFTERNOON, 8 NOVEMBER 2018 RATTENBURY A/B (FE), 1:00 P.M. TO 4:25 P.M.

Session 4pSP

Signal Processing in Acoustics, Underwater Acoustics, Engineering Acoustics, and Physical Acoustics: Detection and Tracking of Mobile Targets II

Siu Kit Lau, Cochair
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Kainam T. Wong, Cochair
Dept. of Electronic & Information Engineering, Hong Kong Polytechnic University, 11 Yuk Choi Road, Hung Hom KLN, Hong Kong

Contributed Papers

1:00

4pSP1. Post-tracking range-rate estimation with up/down linear-frequency-modulated pulses. Douglas Abraham (CausaSci LLC, PO Box 627, Ellicott City, MD 21041, abrahad@ieee.org)

Range-rate (RR) estimation in active sonar is most often accomplished using continuous-wave (CW) pulses. An alternative that is more suited for use with slowly moving objects in reverberation-limited conditions is presented and evaluated. As described by A. Rihaczek [Principles of High Resolution Radar, McGraw Hill, 1969], the approach entails simultaneously transmitting up- and down-sweeping linear frequency modulated pulses (LFM-U&D). The key enabler of the LFM-U&D is the complementary range-bias information relative to using a single LFM sweep type (LFM-U|D). Novel in the proposed LFM-U&D approach is the use of multiple pings after tracking to form improved range and RR estimates. Cramer-Rao lower bounds (CRLBs) on estimator variance as a function of the number of pings illustrate that LFM-U&D is a compromise between the CW and the standard LFM-U|D. Estimates of the RR of clutter tracks from the Target and Reverberation Experiment (TREX13) were used to evaluate the approach. An order of magnitude reduction in the inter-quartile range (IQR) of the RR estimate was observed for LFM-U&D relative to LFM-U|D, which allows a significant reduction in the number of false tracks. The full reduction in IQR predicted by the CRLB was seen to be limited by the effects of surface motion.

1:15

4pSP2. Acoustic tracking and localization using computationally efficient algorithms. Geoffrey H. Goldman (U.S. Army Res. Lab., 2800 Powder Mill Rd., Adelphi, MD 20783-1197, geoffrey.h.goldman.civ@mail.mil)

U.S. Army Research Laboratory (ARL) is developing low size, weight, power, and cost (SWAP-C) solutions for acoustic detection, tracking, and localization of moving targets. Many standard signal processing algorithms cannot be implemented on low SWAP-C commercial off-the-shelf (COTS) hardware for unattended ground sensor applications due to power and computational requirements. To overcome this issue, ARL is developing lower complexity beamforming, tracking and localization algorithms. For example, beamformer algorithms are based upon 1-dimensional scans, multi hypothesis target trackers are based on alpha beta filters and localization algorithms are based upon linear least squares algorithms. The performance of these algorithms is slight poorer than other standard approaches, but their computational complexity can be lower by an order of magnitude.

Invited Papers

1:30

4pSP3. DOA estimation in heteroscedastic noise. Peter Gerstoft, Kay L. Gemba (Noise Lab, Univ. of California, San Diego, 9500 Gillman Dr., La Jolla, CA 92093-0238, gerstoft@ucsd.edu), and Santosh Nannuru (Noise Lab, Univ. of California, San Diego, San Diego, CA)

The paper considers direction of arrival (DOA) estimation from long-term observations in a noisy environment. In such an environment the noise source might evolve, causing the stationary models to fail. Therefore a heteroscedastic Gaussian noise model is introduced where the variance can vary across observations and sensors. The source amplitudes are assumed independent zero-mean complex Gaussian distributed with unknown variances (i.e. the source powers), inspiring stochastic maximum likelihood DOA estimation. The DOAs of plane waves are estimated from multi-snapshot sensor array data using sparse Bayesian learning (SBL) where the noise is estimated across both sensors and snapshots. This SBL approach is more flexible and performs better than high-resolution methods since they cannot estimate the heteroscedastic noise process. An alternative to SBL is simple data normalization, whereby only the phase across the array is utilized. Simulations demonstrate that taking the heteroscedastic noise into account improves DOA estimation.

1:50

4pSP4. Multi-frequency sparse Bayesian learning for matched field processing in non-stationary noise. Kay L. Gemba, Santosh Nannuru, and Peter Gerstoft (MPL/SIO, UCSD, University of California, San Diego, 8820 Shellback Way, Spiess Hall, Rm. 446, La Jolla, CA 92037, gemba@ucsd.edu)

Using simulations and data, we localize a quiet source in the presence of an interferer. The SWellEx-96 Event S59 consists of a submerged source towed along an isobath over a 65 min duration with an interferer traversing the source track. This range independent, multi-frequency scenario includes mismatch, non-stationary noise, and operational uncertainty. Mismatch is defined as a misalignment between the actual source field observed at the array and the modeled replica vector. The noise process changes likely with time. This is modelled as a heteroscedastic Gaussian process, meaning that the noise variance is non-stationary across snapshots. Sparse Bayesian learning (SBL) has been applied previously to the matched field processing application [Gemba *et al*, J. Acoust. Soc. Am., 141:3411-3420, 2017]. Results demonstrate that SBL exhibits desirable robustness properties and improved localization performance when compared to the white noise constraint and Bartlett processors.

2:10

4pSP5. Passive acoustic detection of surface ships at ranges exceeding 100 kilometers and mechanisms for ship noise generation. Matthew E. Schinault (Elec. and Comput. Eng., Northeastern Univ., 360 Huntington Ave., Boston, MA 02115, schinault.m@husky.neu.edu), Heriberto A. Garcia (Elec. and Comput. Eng., Northeastern Univ., Arlington, MA), Chenyang Zhu, Anna Kaplan, and Purnima Ratilal (Elec. and Comput. Eng., Northeastern Univ., Boston, MA)

The passive ocean acoustic waveguide remote sensing technique is employed to detect diesel-electric vehicles at ranges exceeding 100 kilometers. The underwater sounds radiated from these vessels are received at long ranges on a large-aperture densely-sampled horizontal coherent hydrophone array. The source levels of these signals are estimated by correcting the received pressure levels for transmission losses modeled using a calibrated parabolic equation-based acoustic propagation model for random range-dependent ocean waveguides. Here we find spectra of ship-radiated sound that is extremely dynamic containing both broadband signals and narrowband tonals at discrete frequencies with source levels that vary depending on ship conditions. We track a vessel with increasing range to find range dependence on broadband signals at close range and tonal signals at long range. Machinery noise generated from engines, propellers, flow noise and other cavitation sources are found to vary depending on ship conditions and are unique to each vessel. Our analysis indicates these vessels can be instantaneously tracked over wide areas spanning more than 300 kilometers in diameter.

Contributed Papers

2:30

4pSP6. Comparison of adaptive and compressed sensing beamformers. Paul Hursky (Sonar-synesthetics, 4274 Pilon Point, San Diego, CA 92130, paul.hursky@gmail.com)

We will review the capabilities and requirements of the Minimum Variance Distortionless Response (MVDR) and sparsity-seeking beamformers, from a system design perspective. We will focus on the problem of detecting a quiet source in the presence of loud interferers. These methods are solutions to particular and different optimization problems, requiring particular combinations of inputs. Thus, for example, MVDR does not require us to model the interference, whereas the sparsity-seeking methods need both interferer and quiet source models to be put into their dictionary. MVDR is notorious for requiring more precise calibration than is needed for conventional beamformers. We will present demonstrations on simulated and experiment data, illustrating key differences between the two methods.

2:45–3:05 Break

3:05

4pSP7. Extraction of shaft frequency based on the DEMON line spectrum. Mingyu Song, JiangQiao Li (Systems Eng. Res. Inst. of CSSC, Systems Eng. Res. Inst. of CSSC, Beijing CuiWei St., Beijing, China, 56608812@qq.com), and Juan Hui (Underwater Acoust., Harbin Eng. Univ., Harbin, Heilongjiang, China)

Target-identification is an important technique in modern sea war. As the core device to identify targets, negative sonar analyzes noise underwater and abstracts their properties and determines its source. Shaft frequency which barely lies in the properties of the propeller differs among different kinds of ships, so this property can be regarded as a key to identify targets. This paper studied the theory and math model of the noise produced by ships. And Detection of Envelope Modulation on Noise (DEMON) is an effective technique to abstract low-frequency properties from high-frequency signals via demodulation. There are two methods to demodulate the noise, namely absolute low pass demodulation and square low pass demodulation. To improve the performance of DEMON spectrum estimation, adaptive line spectrum enhancement technology is introduced. An adaptive threshold is then set to eliminate the continuous spectrum. Then the largest common divisor algorithm is presented to calculate to shaft frequency out of the array consist of the centre frequency of these line spectrum. Simulation is conducted as described above. Two methods of demodulation is compared in this part. Brief sea experiment data processing shows that this algorithm is able to abstract the shaft frequency among the ship noise.

3:20

4pSP8. Research on suppression for tow-ship interference. Juan Hui (Underwater Acoust., Harbin Eng. Univ., Harbin, Heilongjiang 150001, China, huijuan@hrbeu.edu.cn), Mingyu Song, and JiangQiao Li (Systems Eng. Res. Inst. of CSSC, Beijing, China)

Towing line array underwater detection equipment detection performance will be tug of radiation noise interference, in order to enhance the performance of sonar detection, the key from the array spacing and follow-up signal processing to proceed. The purpose of using the towing array underwater probing device is to target the tail of the tug, but in actual use the towed array of half-wave lengths is affected by the strong tug noise The detection performance of the ship's tail direction is greatly restricted, and it is imperative to improve it. Otherwise, it will not be able to detect the threat target from the wake of the wake, and the target can not be found in time. This paper focuses on the reception directivity of the quarter-wave spacing matrix under the single-frequency signal and the wideband signal. By comparing the results of the two directional simulations, it is known that, unlike the half-wavelength interval matrix, the end-to-receive directionality of the quarter-wave spacing matrix is asymmetric, which can significantly suppress the interference in the direction of the drag Detection performance. The tandem noise can be suppressed by the combination of the received signal and the zeroing of the tampon, and the quarter-wave spacing matrix formed by the permutation-weighted combination array can further suppress the tug noise.

Invited Paper

3:35

4pSP9. Estimating the velocity of a moving acoustic source based on chirplet transform. Ningning Liang, Yixin Yang, Xijing Guo, and Boxuan Zhang (School of Marine Sci. and Technol., Northwestern PolyTech. Univ., 127 West Youyi Rd., Beilin District, Xi'an, Shaanxi 710072, China, liangningning96@gmail.com)

The instantaneous frequency (IF) of a pure tone emitted by a transiting acoustical source varies due to the acoustical Doppler effect, which is usually used to estimate the velocity of the source. Hence, the common practice is first to estimate the IFs by the time-frequency analysis (TFA) of the signals and then to derive the velocity from the IFs. In this paper, a TFA method based on the chirplet transform is proposed where the IFs of the chirplets are formulated by the nonlinear kinetic function with respect to the source velocity. By iteratively fitting the chirplets to the signals it directly renders the velocity of the source. The efficiency of the proposed method is validated by the microphone recorded noise data due to a flying helicopter during an experiment carried out in May, 2018, northeast to the coast of Sanya, Hainan province, China.

Contributed Papers

3:55

4pSP10. The frequency choice of continuous-wave signal in low frequency active detection. Yiming Gu, Yanjun Zhang, ZaiXiao Gong, and Zhenglin Li (State Key Lab. of Acoust., Inst. of Acoust., Chinese Acad. of Sci., No. 21 North 4th Ring West Rd., Haidian District, Beijing, Beijing 100190, China, guyiming16@mails.ucas.ac.cn)

In active sonar, continuous-wave (CW) pulses are used to evaluate the velocity of a target because of its high-resolution of velocity. Unfortunately, due to multipath interface in shallow water, the received CW signals are fluctuating. This phenomenon degrades the performance of velocity measurement. The interference structure of low frequency sound field in shallow water is relatively stable and the intensity striations are regular in range-frequency graph, which makes it possible to reduce the influence of amplitude fluctuations by reasonable frequency design. In this paper, we obtain the separations of striations caused by different modes interfering. According to the separations, appropriate frequencies of multi-frequency CW pulses were selected to compensate the fluctuations. The simulation and experimental results show that this method can mitigate the influence of frequency selective fading in shallow water channel.

4:10

4pSP11. Development of the application of the digital microphone array system for real time source localization. Gee-Pinn J. Too and Ke-Han Liao (Systems and Naval Mechatronic Eng., National Cheng Kung Univ., No. 1 University Rd. NCKU, Tainan 70101, Taiwan, toojames@yahoo.com)

This research is to develop a digital microphone array system to build a real-time sound source localization system and to produce acoustical holography. Acoustical holography can be understood as taking a photo of sound which can be used to source localization. In the situation of free field with an unknown signal, source signal is restored by time reversal mirror(TRM) where a free-field impulse response function(IRF) is used to achieve the goal of real time source localization. During the processing, the unknown source location is identified by scanning over a possible space for source locations to find a maximum restored signal. In the present study, simulations and experiments of source localization are carried out in 2D and 3D space. In the experiment of source localization with single source, the estimation errors are below 10% in certain distances between source and microphone array. For two sources situation, the distance between two sources will limit the application of the present approach.

Session 4pUWa

Underwater Acoustics, Signal Processing in Acoustics, and Structural Acoustics and Vibration: Acoustic Vector Field Studies II

Kevin B. Smith, Cochair
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Robert Barton, Cochair
NUWC, 1176 Howell St, Newport, RI 02841

Invited Papers

1:00

4pUWa1. Review of directional sensors for low frequency underwater acoustic applications. James A. McConnell and Timothy P. Rorick (Ultra Electronics - Undersea Sensor Systems, Inc., 4868 East Park 30 Dr., Columbia City, IN 46725, james.mcconnell@ultra-ussi.com)

In this review, we present some of the more common embodiments directional sensors take on for use in low-frequency underwater acoustic applications. Sensors that fit within this paradigm are typically substantially smaller than an acoustic wavelength and can form first- or second-order cardioid beams at frequencies well below 10 kHz. A mathematical treatment of scalar, vector, and dyadic acoustic fields is presented as it pertains to understanding the phenomenology the sensors are required to measure. Cardioid beam-forming and directivity index are explained to provide context of using directional sensors in aperture constrained situations. The electro-acoustic performance of various sensor concepts is presented using lumped parameter models. Packaging concepts for high-fidelity operation, low self-noise, and low electronic noise in the seawater environment are shown. Special problems such as using directional sensors in deep water or near impedance boundaries is also covered.

1:20

4pUWa2. The rate of energy transport and direct measurements of group velocity. Peter H. Dahl (Appl. Phys. Lab. and Mech. Eng., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, dahl@apl.washington.edu) and David R. Dall'Osto (Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

The Intensity Vector Autonomous Recorder (IVAR) is a bottom deployed system measuring both particle velocity and pressure (combined sensor). Results using IVAR in the Sediment Characterization Experiment (SCE17) conducted off New England (spring 2017), are presented. Data originate both from ship noise involving the closest point of approach (CPA) of a large transport vessel, and experimental SCE17 signals relating to Signal Underwater Sound (SUS) MK-64 explosive sources. Several vector and scaler metrics emerge based on different combinations of second-order acoustic fields; one is the rate of energy transport (U) to be emphasized here. For the CPA data, estimates of U versus frequency represent the speed of the net transport of acoustic energy [D'Spain *et al.*, J. Acoust. Soc. Am. 89, 1991], and additional interpretations are presented. For the SUS data, arrivals from individual modes are resolved in time-frequency analysis. Vertical intensity is shown to be much less than horizontal, and estimates of U are identified with modal group velocity with some caveats to be discussed. For example, the group velocity associated with the Airy-phase for mode-2 (frequency ~29 Hz) is U ~ 1370 m/s. This estimate emerges directly from the analysis without recourse to range divided by travel time.

1:40

4pUWa3. Near field scattering measurements using acoustic vector sensors. Georges Dossot, Jessica A. Barker, Daniel Perez, and Robert Barton (NUWC, 1176 Howell St., Bldg 1320, Code 1524, Rm. 260, Newport, RI 02841, georges.dossot@navy.mil)

The objective of this research is to measure and characterize acoustic energy flow surrounding underwater objects in the near field condition. To accomplish this, we employ prototype acoustic vector sensors which simultaneously measure acoustic pressure and three-dimensional acoustic particle acceleration at a singular point, allowing for precise vector field measurements. In March of 2018, a test at the U.S. Navy's Dodge Pond Test Facility examined the intensity field surrounding hollow spheres at low *ka* (wavenumber × radius) values. Phase differences between varying acoustic path lengths from reflected, scattered, and creeping waves result in interference patterns around the object. Various intensity processing techniques are used to reconstruct the acoustic field. Instantaneous intensity describes the time-dependent energy flux of the field and can be divided into active (real) and reactive (imaginary) components, which represent physically real elements. Time-averaged intensity shows energy transport and can be visualized in the form of acoustic streamlines.

2:00

4pUWa4. Ambient vector field measurements in the Northwest Providence Channel. Thomas J. Deal, Derrick Custodio, and Benjamin Cray (Naval Undersea Warfare Ctr., 1176 Howell St., Newport, RI 02841, thomas.deal@navy.mil)

Data collected by a low-frequency acoustic vector sensor in shallow water west of the Berry Islands, Bahamas, will be presented. The frequency- and direction-dependence of the ambient noise was used to identify potential sources, such as distant shipping, wind-driven surface noise and subsurface currents. Noise sources were validated using recorded Automatic Identification System (AIS) tracks, weather observations and Acoustic Doppler Current Profiler (ADCP) measurements. A model of wind-driven surface noise that accounts for propagation effects due to range-dependent bathymetry, sound speed profiles, and surface waves will also be presented and compared to measured data.

2:20–2:35 Break

Contributed Papers

2:35

4pUWa5. Vector field observations near the continental slope off Big Sur, California. Kevin B. Smith (Dept. of Phys., Naval Postgrad. School, 833 Dyer Rd., Monterey, CA 93943, kbsmith@nps.edu), Thomas J. Deal (Naval Undersea Warfare Ctr. Div. Newport, Newport, RI), Paul Leary (Dept. of Phys., Naval Postgrad. School, Monterey, CA), Steven Seda, Benjamin Carpenter (U.S. Navy, Monterey, CA), and Fahmi Laksana (Indonesian Navy, Monterey, CA)

In this work, data collected approximately 3 km off the coast of Big Sur, California, by a low frequency vector sensor system will be analyzed. Directional and temporal variations in the ambient noise field will be evaluated, and causes will be considered including flow noise due to currents, surface (wind) noise, distant shipping, and marine mammals. Bearing estimation results for shipping will be compared with AIS tracks recorded during the period of deployment. Local sound speed measurements and bathymetry in the vicinity of the deployment area will be used as inputs to a two-dimensional propagation model that properly invokes reciprocity of the acoustic vector field. Surface wave spectra collected by a nearby buoy will be used to generate rough surface realizations in order to evaluate the impact of these perturbations on low frequency vector field structure. These results will be combined with acoustic data observations in attempts to determine the impact of propagation on the structure of the field. Such an approach may allow us to infer acoustic source levels of AIS traffic off the coast.

2:50

4pUWa6. Embedded real-time processing of acoustic vector sensors using a lightweight open-source microcontroller. Paul Leary (Phys., Naval Postgrad. School, 833 Dyer Rd., Spanagel Hall, Rm. 203, Monterey, CA 93943, pleary@nps.edu), Vladimir Dobrokhodov, Kevin Jones (Mech. and Aerosp. Eng., Naval Postgrad. School, Monterey, CA), and Kevin B. Smith (Phys., Naval Postgrad. School, Monterey, CA)

This abstract is intended for the special session "Acoustic Vector Field Studies." In this work, we will show our progress towards a low-power, lightweight, embedded data acquisition system and real-time signal processor for acoustic vector sensors, using an open-source microcontroller. Modern advances in low-cost, low-power "systems-on-a-chip" (SoC) provide unprecedented computational power, and some state-of-the-art devices are particularly well designed for acoustic signal processing applications. Here, we present the results of performance experiments, demonstrating the speed and accuracy of sampling and processing audio-frequency signals for a wide variety of signals and sampling regimes. We demonstrate trade-offs between sampling rates and resolutions for accuracy and speed at desired frequencies, and how these may be dynamically managed by autonomous systems. Finally, we will share some of our work towards real-time acoustic beamforming for use on autonomous vehicles.

3:05

4pUWa7. Effect of acoustic horizontal refraction on DOA estimation with a single vector hydrophone. Jun Tang and Shengchun Piao (College of Underwater Acoust., Harbin Eng. Univ., Nantong St. 145, Harbin 150001, China, tangjun@hrbeu.edu.cn)

The effect of horizontal refraction (HR) on DOA estimation with a single vector hydrophone is studied. It has been demonstrated in a previous study that HR may bring about a significant deviation between the true bearing of the source and the direction of time-averaged sound flux at the receiver point, i.e., DOA estimation error. [J. Tang *et al*., 2018, 43(2), ACTA ACUSTICA (in Chinese)]. In the present work, the DOA estimation error in a more realistic scenario is studied, more precisely, source signals are considered to be wide-band instead of time-harmonic, and meanwhile ambient noise is added to the received signal. The ocean waveguide used in simulations is a 3D version of the standard ASA wedge, which implies that the HR considered in this work is raised from multiple reflections between a horizontal sea surface and a sloping sea bottom. The results of this work shall offer some reference to DOA estimation in circumstances with strong HR effects.

3:20

4pUWa8. Parabolic equation modeling of a seismic airgun array. Kevin D. Heaney (OASIS Inc., 11006 Clara Barton Dr., Fairfax Station, VA 22039, oceansound04@yahoo.com) and Richard L. Campbell (OASIS Inc., Lexington, MA)

The numerical modelling of seismic airgun sources, with a specific emphasis on the impact of sound on the marine ecosystem, is a challenging problem. In this paper, the simplified range-independent iso-velocity problem set up as part if the International Airgun Modelling Workshop (IAMW), is solved using the Parabolic Equation (PE). The acoustic pressure and particle acceleration are computed including the arrival time series and source energy level in deci-decade bands for ranges spanning from 30m to 30 km. The particle acceleration is computed by taking the spatial gradient of the pressure on the PE computational grid. The two significant challenges this problem poses to the PE solution are the very large bandwidth (5-4500 Hz) and the computational accuracy required for time-series and particle acceleration computations. Each of the metrics outlined in the statement of the problem are computed and presented. In addition to computing the required fields, the 17-gun volumetric array is placed in a 3D ocean environment in deep water in the Gulf of Mexico, with peak pressure computed out to 30km and the band integrated Signal Energy Level (SEL) computed to 400 km.

Session 4pUWb

Underwater Acoustics, Acoustical Oceanography, Signal Processing in Acoustics, and Physical Acoustics: Sediment Acoustics–Inferences from Forward Modeling, Direct, and Statistical Inversion Methods II

Charles W. Holland, Cochair
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Stan E. Dosso, Cochair
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Chair's Introduction—1:00

Invited Papers

1:05

4pUWb1. Statistical inference approach applied to simultaneous vertical particle velocity and acoustic pressure measurements from SUS explosive sources made in the New England Mudpatch. David P. Knobles (KSA LLC, PO Box 27200, Austin, TX 78755, dpknobles@yahoo.com), Peter H. Dahl, David R. Dall'Osto (Appl. Phys. Lab., The Univ. of Washington, Seattle, WA), and Preston S. Wilson (Mech. Eng. Dept. and the Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

Pressure and particle velocity time series were measured in the New England "Mudpatch" in the Spring of 2017. The time series were generated from Signal Underwater Sound (SUS) MK-64 explosive sources and recorded by a bottom-mounted L-shaped array of hydrophones and a vector sensor system that recorded both pressure and particle velocity (1.3 m off the bottom) in a separate location. A maximum entropy approach was utilized to infer the marginal probability distributions of geo-acoustic parameter values of the seabed using both the particle velocity and the pressure. The seabed consists of a surface layer of mud over sand whose thicknesses are range- and azimuth dependent. The modeled vertical component of particle velocity is computed by a three-point derivative of the pressure fields generated by RAM-PE along the vertical axes centered at the depth of the vector sensor. The two-way travel time CHIRP data for the multiple range and azimuth-dependent sediment layers provide a constraint for the sediment thickness for each sound speed hypothesis of the model parameterization space. Results of the statistical inference of the geo-acoustic structure derived from the particle velocity measurements are compared to those derived from pressure field measurements. [Work supported by ONR.]

1:25

4pUWb2. Inferring low-frequency compressional sound speed and attenuation in muddy sediments from long-range broadband sound propagation. Lin Wan, Mohsen Badiey (Univ. of Delaware, Newark, DE 19716, wan@udel.edu), David P. Knobles (Knobles Sci. and Anal., LLC, Austin, TX), and Preston S. Wilson (Mech. Eng. Dept. and Appl. Res. Labs, The Univ. of Texas at Austin, Austin, TX)

While researchers have successfully applied many geo-acoustic inversion methods, involving normal mode analysis or matched field processing, to invert seabed parameters in sandy bottoms, inferring low-frequency compressional sound speed and attenuation in mud is still a challenging problem, especially for a marine sediment where a mud layer overlies a sandy bottom. Recent studies show that there is an ambiguity between the low-frequency attenuation of mud and sand in three different inversion algorithms based on (1) modal amplitude, (2) transmission loss, and (3) spatial coherence measurements [JASA **143**, 1798 (2018)]. An increase (decrease) of mud attenuation can be compensated by decreasing (increasing) the attenuation in the sandy basement. In this paper, an inversion method combining these three inversion algorithms and statistical inference techniques (e.g., Bayesian-Maximum Entropy, [JASA **138**, 3563–3575 (2015)]) is utilized to analyze the long-range broadband acoustic signals measured by several vertical and horizontal line arrays during the Seabed Characterization Experiment 2017 conducted in the New England Mud Patch. Reliable estimates of the low-frequency mud properties are obtained by removing the ambiguity. Finally, the inverted results are compared with historical experimental data and theoretical predictions. [Work supported by ONR Ocean Acoustics.]

1:45

4pUWb3. In situ observation of sediment sound speed and attenuation from coarse to fine-grained sediments. Jie Yang (Acoust. Dept., APL-UW, 1013 NE 40th St., Seattle, WA 98105, jieyang@apl.washington.edu)

The long-term objective is to address the impact of spatial variability of sediment properties on sound propagation and reverberation from a few hundred hertz to 10 kHz. In this band, the frequency dependences of sound speed and attenuation are important for applications, but cannot be inferred from high-frequency data, hence direct measurement is necessary. In this talk, a summary of *in situ* measurements of sediment sound speed and attenuation collected from three major field experiments, i.e., Shallow Water 2006 (SW06),

Target and Reverberation Experiment 2013 (TREX13), and Seabed Characterization Experiment 2017 (SCE17), are presented. Sediment sound speed and attenuation within the surficial 3 m of sediments were obtained through penetration, using the Sediment Acoustic-speed Measurement System (SAMS). In TREX13, *in situ* measurements were carried out at five sites along a 5-km track, with sediment types ranging from coarse sand to a mixture of soft mud over sand. SAMS was deployed at 18 sites within the 10×30 km SCE17 study area. Significant variation of sediment geoacoustic properties was observed in range (TREX13) and depth (SCE17), respectively. Preliminary sediment acoustic modeling work is presented, using SAMS data taken from the wide range of sediment types. [Work supported by ONR.]

2:05

4pUWb4. Bottom scattering by a moving, narrowband source. Brian T. Hefner (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, hefner@apl.washington.edu) and William S. Hodgkiss (UC San Diego, La Jolla, CA)

During the Target and Reverberation Experiment in 2013 (TREX13), two vertical line arrays were deployed and a mid-water source was towed past these arrays while transmitting 15-s long, mid-frequency CW tones. While the received signals exhibited the expected positive and negative Doppler shifts as the source moved toward and away from the arrays, a significant amount of energy was always present in the band between these two frequencies. This counterintuitive result is due to sound scattering from the seafloor in the vicinity of the source. This sound will experience a different Doppler shift and although it may be incident above the critical angle, it can scatter into propagating modes that can be received on the arrays. As opposed to wide-band reverberation where the energy received at a given time can be associated with an elliptical scattering patch, in this case the energy received at a given frequency can be associated with scattering from a hyperbolic scattering patch. A normal-mode reverberation model for this effect has been developed and is used to examine whether this data can be used to invert for the seafloor scattering strength at the TREX13 site. [Work supported by ONR.]

2:25–2:40 Break

Contributed Papers

2:40

4pUWb5. An experimental test of end-fire synthetic aperture sonar for sediment acoustics studies. Shannon-Morgan M. Steele and Anthony P. Lyons (Ctr. for Coastal and Ocean Mapping/Oceanogr., Univ. of New Hampshire, 24 Colovos Rd., Durham, NH 03824, ssteele@ccom.unh.edu)

Seafloor sediment acoustic returns are comprised of scattering from both the interface and sediment volume. At low-frequencies, volume scattering is often the dominant mechanism; however, direct measurements of this component have rarely been made, due to interface roughness biasing caused by large beamwidths. End-fire synthetic aperture sonar (SAS) can achieve narrower beamwidths by coherently combining multiple acoustic returns as a vertically oriented transmitter and/or receiver is moved towards the seafloor. Beam pattern simulations suggest end-fire SAS can reduce the beamwidth of a sonar by a factor of 6 with an array length of 100 wavelengths. Achieving these gains is dependent on the ability to resolve relative sonar motion to at least an eighth of a wavelength. This talk will present results from an end-fire SAS field trial. Results will include an analysis of beamwidth gains achieved during the end-fire SAS field test and methods to improve these gains by using the scattered field to refine sonar positions.

2:55

4pUWb6. Studies on water column nutrient distribution and sound speed profile variations along coastal region of India. Sakthivel M. Santhanam (Electronics and Commun., SSN College of Eng., Rajiv Gandhi Salai, OMR Kalavakkam, Chennai, Tamil Nadu 603110, India, sakthivels@ssn.edu.in)

This study aim is to measure the water column nutrient and sound velocity variation along the coastal region of Poompuhar, Cuddalore, Mahabalipuram, Kalpakkam, Chennai and Pondicherry. Sound speed variation in the environment of the particular ocean depends on temperature, salinity, and pressure. Since temperature and salinity variations are large compared to pressure. In study locations the sound speed varies from 1445 to 1540m/s, temperature 27 to 29^0c and salinity 29 to 33psu. Water column nutrients like Nitrate, Nitrite, Phosphate, Silicate and Urea were analyzed to understand the physical and chemical concentration of study location. This nutrient has an inverse relationship with the temperature. Location-based nutrients analysis has been carried to identify the concentration each nutrients level in the particular location. The maximum concentration of nutrients observed in the study locations are silicate in Pondicherry (84%), Nitrite in Kalpakkam (4%), Nitrate in Cuddalore (16%), Phosphate in Chennai (6%), and Urea Poompuhar-2 (24%). The total suspended solids level in the water column is measured to identify the impact on the signal transmission in underwater. The higher value of TSS is observed in Cuddalore (0.15 mg/L).

3:10

4pUWb7. Measurements of modal attenuation using broadband sources in the New England Mud Patch. Kerry Unrein, Gopu R. Potty, James H. Miller (The Univ. of Rhode Island, Narragansett, RI 02882, kerrycutler@my.uri.edu), Dag Tollefsen (Norwegian Defense Res. Establishment, Horten, Norway), David P. Knobles (KSA, LLC, Austin, TX), Preston S. Wilson (Mech. Eng. and Appl. Res. Labs, The Univ. of Texas at Austin, Austin, TX), and Ying-Tsong Lin (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., Woods Hole, MA)

Measurements of underwater acoustic signals were made on a bottom-mounted horizontal line array during the Seabed Characterization Experiment (SCEx) in the New England Mud Patch south of Martha's Vineyard in about 70 meters of water. The signals were generated by SUS (Signals, Underwater Sound) charges detonated at various locations in the experimental area at a depth of 18 m, during nearly-isovelocity conditions. The broadband signals were analyzed for modal arrival time and amplitude using time-frequency techniques. Ratios of modal amplitudes at the different hydrophones were used to estimate the modal attenuation coefficients. Hence, these estimates are independent of any uncertainty in the frequency-dependent source level of the SUS charges. These coefficients are directly related to the depth-dependent sediment attenuation profile. A sensitivity study was performed to understand how the modes sample the different layers and provides estimates of resolution kernels. Posteriori error analysis provides averages and standard deviations for the estimate of sediment attenuation as function of depth. The frequency bands of interest range from 10 Hz to 200 Hz for modes 1 to 4. We will compare our estimates of sediment attenuation with historical measurements. [Work supported by Office of Naval Research.]

3:25

4pUWb8. On the use of conventional, adaptive beamforming and the Fourier-Transform-based algorithm for estimating seabed properties.
Lanfranco Muzi, Martin Siderius (Elec. and Comput. Eng., Portland State Univ., 1900 SW 4th Ave., Ste. 160, Portland, OR 97201, muzi@pdx.edu), and John Gebbie (Metron, Inc, Portland, OR)

Knowledge of the bottom reflection coefficient as a function of frequency and grazing angle can be used to infer seabed properties. A known passive technique for estimating the bottom reflection coefficient is based on conventional beamforming of natural marine ambient noise over a vertical line array. In this study, the extension of the technique to adaptive beamforming is formally investigated, and compared to a more recent Fourier-Transform-based algorithm. This is done by developing both a mathematical formalization of the high-resolution algorithm in the discrete space-wavenumber domain, and a mathematical proof of the original technique (based on conventional beamforming). It is shown that replacing conventional beamforming with adaptive beamforming cannot be guaranteed to provide an estimate of the bottom reflection coefficient. The conclusions are demonstrated on simulated and measured data. [Work supported by the Office of Naval Research, Ocean Acoustics Program.]

OPEN MEETINGS OF TECHNICAL COMMITTEES/SPECIALTY GROUPS

The Technical Committees of the Acoustical Society of America will hold open meetings on Tuesday, Wednesday, and Thursday. See the list below for the exact schedule.

These are working, collegial meetings. Much of the work of the Society is accomplished by actions that originate and are taken in these meetings including proposals for special sessions, workshops, and technical initiatives. All meeting participants are cordially invited to attend these meetings and to participate actively in the discussion.

Committees meeting on Tuesday, 6 November

Committee	Start Time	Room
Engineering Acoustics	4:30 p.m.	Rattenbury A/B (FE)
Acoustical Oceanography	7:30 p.m.	Esquimalt (VCC)
Animal Bioacoustics	7:30 p.m.	Oak Bay 1/2 (VCC)
Architectural Acoustics	7:30 p.m.	Theater (VCC)
Physical Acoustics	7:30 p.m.	Colwood 1/2 (VCC)
Psychological and Physiological Acoustics	7:30 p.m.	Salon B (VCC)
Speech Communication	7:30 p.m.	Salon A (VCC)
Structural Acoustics and Vibration	7:30 p.m.	Saanich 1/2 (VCC)

Committees meeting on Wednesday, 7 November

Committee	Start Time	Room
Biomedical Acoustics	7:30 p.m.	Sidney 1/2 (FE)
Signal Processing in Acoustics	7:30 p.m.	Colwood 1/2 (FE)

Committees meeting on Thursday, 8 November

Committee	Start Time	Room
Computational Acoustics	4:30 p.m.	Esquimalt (VCC)
Musical Acoustics	7:30 p.m.	Crystal Ballroom (FE)
Noise	7:30 p.m.	Shaughnessy (FE)
Underwater Acoustics	7:30 p.m.	Rattenbury A/B (FE)