Session 5aAA

Architectural Acoustics: Topics in Architectural Acoustics: Measurements, Modeling, and Isolation

Ian B. Hoffman, Cochair
Peabody Institute, Johns Hopkins Univ., 1 E Mount Vernon Place, Baltimore, MD 21202

Shane J. Kanter, Cochair
Threshold Acoustics, 53 W. Jackson Blvd., Suite 815, Chicago, IL 60604

Chair's Introduction—8:30

Contributed Papers

8:35

5aAA1. Effect of acoustic computer model precision on auralization perception. Brandon M. Westergaard and Timothy Hsu (School of Music, Georgia Inst. of Technol., 840 McMillan St. NW, Atlanta, GA 30318, brandon.westergaard@gatech.edu)

Most modern acoustic simulation programs use geometric acoustics to auralize the sound within a space, providing acceptable results in a relatively quick manner. Still, because of the limitations of geometric acoustics in computer simulations, it is of particular interest to investigate the relationship between model geometry, simulation parameters, and the listener's perception of auralizations created from iterations of an acoustic model. In the context of this study, differences between auralizations were investigated from a perceptual standpoint, using the subjective judgment of listeners. A modified version of the Multiple Stimulus test with Hidden Reference and Anchor methodology was used to define a threshold at which an increase in model precision no longer results in perceived differences between auralizations, aiming to inform efficient modeling practices in practical applications. Acoustic simulations were performed on a set of speech and music venues modeled at incremental geometrical precision regarding their real-life counterparts, ultimately resulting in a series of auralizations that reflected changes in modeling precision. Single factor analysis of variance and a series of two-sample t-tests indicate that increasing acoustic model precision may not always result in perceptually relevant differences.

8:50

5aAA2. Room impulse response synthesis with device diffraction via image source method and finite element analysis. Aidan Meacham and Andrew Unruh (Knowles Electronics, 331 Fairchild Dr., Mountain View, CA 94043, aidan.meacham@knowles.com)

A method to create an accurate reproduction of the soundfield on the surface of a device was developed through the combination of the image source method and anechoic finite element analysis simulations. Typically, the image source method is unable to model diffraction of sound pressure around a device, and it would be impractical to model the wave equation in an entire room with finite element analysis due to computational constraints. By combining the two techniques, however, individual impulse responses from a precomputed finite element dataset can be assembled like image sources to derive transfer functions from sources in space to receivers on the surface of a device. The finite element dataset is comprised of anechoic simulations of plane waves incident on the device, where the number and direction of plane waves are chosen to maximize angular resolution. The direction of arrival from a given image source is matched with a simulated plane wave, and the corresponding anechoic impulse response is delayed and attenuated according to the image source's time of travel and reflection path. Each reflection is constructed in this manner, and all are summed to create a synthesized reverberant impulse response that accurately portrays diffraction around the device.

9:05

5aAA3. Acoustic characterization of Penn State's recording studio above and below the Schroeder frequency. Andrew D. Kinzie and Daniel A. Russell (Acoustics, Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, azk5643@psu.edu)

In most performance spaces, rooms are large enough that most of the audible frequency range falls above the Schroeder frequency, whereas smaller spaces have a significant audible range below the Schroeder frequency. The goal of this study was to characterize the acoustic performance of the control room and live room of Penn State's recording studio. Both rooms have important roles to play in the operation of a studio which depend on their acoustic character. COMSOL models for each room were created to understand the modal behavior below the Schroeder frequency and measured modes were visualized from frequency response measurement points in one plane at a one foot spacing. For frequencies above the Schroeder frequency, an ODEON model was created and validated against measured impulse responses, where metrics such as T60 and EDT have been considered. By comparing the computer simulations to the measurements, one is able to make recommendations for improvements and provide audio engineers with insight on how they can adapt to the room and refine their craft.

9:20

5aAA4. Assessment of a variable acoustic system proposal in a real case. Merve Esmebasi, Ali Murat Tanyer (Architecture Dept., Middle East Tech. Univ., Ankara 06800, Turkey, merveesmebasi@gmail.com), and Mehmet Caliskan (Mech. Eng., Middle East Tech. Univ., Ankara, Turkey)

This paper is about testing a variable acoustic system proposal under development in a real case. The variable acoustic system is designed as alternating slot panels and prototyped via using Arduino and Processing. Digital interface to control the system is designed in Processing and via Arduino board digital commands are converted to physical movements. After the possibility of construction of variable perforated panel system is simply clarified, perforated panels of the system are produced via CNC routers. Panel positions are defined regarding perforation characteristics in terms of slot width and distances between slots. Changing these parameters is linked to perforation ratio of the system, while air gap and porous material behind are kept unchanged. Sound absorption capabilities of variable perforations are examined with Impedance tube measurements. Measurement results of variable slot panels, as a promising variable acoustic system, are transferred to ODEON room acoustics software, to assess the system in terms of room acoustics. The amphitheater at METU Faculty of Architecture Building, used for different functions from lectures to musical performances, is

selected to demonstrate the capability of the proposed variable acoustics system to improve acoustical conditions in an amphitheater.

9:35

5aAA5. Experimental characterization of the Green's function in a room using sparse reconstruction principles. Efren Fernandez-Grande (Acoust. Technol., Tech. Univ. of Denmark, Ørsteds Plads, B. 352, DTU, Kgs. Lyngby DK-2800, Denmark, efg@elektro.dtu.dk) and Rasmus Ellebæk Christiansen (Mech. Eng., Tech. Univ. of Denmark, Kgs. Lyngby, Denmark)

Measuring the Green's function over the entire volume of a room would typically require an unfeasible number of measurements, due to requirements on spatial sampling. To alleviate the need for excessive measurements, sparse reconstruction methods can be employed, as they make it possible to reconstruct a seemingly undersampled signal. The present study proposes a method for acquiring experimentally the Green's function in a room by measuring directly the mode shapes of the room, based on the conception that any mode can be expanded into a number of propagating waves. If the modes are described in the wavenumber domain (as a plane-wave expansion), sparse reconstruction methods can be employed, under the implicit assumption that each mode shape is represented as the superposition of a small number of plane waves. In addition, it is assumed that the medium is source-free and homogeneous. The methodology is examined numerically and verified experimentally, based on measurements in a lightly damped rectangular room.

9:50

5aAA6. A concert hall database of US and European halls: Preliminary measurements and results. 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)

Subjective perception in the realm of concert hall acoustics, specifically overall impression, is a difficult problem to approach. Ideally, this type of work should be done using realistic concert hall auralizations, allowing direct comparison of a wide variety of rooms from around the world. Currently, measurements are being taken in concert halls across the United States and in Europe. Halls to be measured were selected from an online survey of researchers and consultants around the United States and Europe. Final selections were made to ensure that a wide variety of hall shapes, sizes, and reverberation times were included in the database, given the available travel resources. Measurements have been made using a 32-element spherical microphone array, a three-way omnidirectional sound source, and a directional sound source. The directional sound source is a 20-element compact array that can reconstruct the frequency-dependent radiation patterns of different orchestral instruments. An overview of the halls included in the database and measurement results will be presented to illustrate the variety within the database. As well, plans for upcoming subjective testing to piece-apart the perception of preference in concert hall acoustics, considering both the hall and individual taste, will be discussed. [Work supported by NSF Award 1302741.]

10:05–10:20 Break

10:20

5aAA7. Absorption: The misunderstandings about what can be measured and the newest ideas of how to perform these measurements. Ronald Sauro (NWAA Labs, Inc., 90 Tower Blvd., Elma, WA 98541, audio_ron@msn.com)

We will present the latest research into the measurement of absorption, the pitfalls, and some possible ways out of the these pits. We will also present new information on methods derived from the measuring of diffusion that can be applied to measuring absorption.

10:35

5aAA8. Overcoming field impact isolation performance failures due to faulty flooring installation practices. Marlund E. Hale (Adv. Eng. Acoust., 663 Bristol Ave., Simi Valley, CA 93065, mehale@aol.com)

There is very strong demand by new owners of luxury condominiums for hard-surfaced flooring. In some cases, there is a medical need. In most cases, the reasons are personal preference. However, most high-end condominium homeowner associations (HOA) have enacted regulations requiring that upper level dwelling unit owners, making flooring upgrades from any existing flooring to hard-surfaced flooring, meet higher minimum field impact insulation class (FIIC) or impact sound rating (ISR) performance standards than is typically required by building codes. Failure to comply usually causes the HOA to require restoration to prior flooring conditions within a certain time period, with fines and potential litigation for restoration delays. This paper presents field performance test results for several renovated condominiums in different multi-family residential complexes that had successfully passed HOA impact performance tests by using customized sample underlayments and the owners' desired hard-surfaced flooring, but failed to pass the HOA impact performance requirements after final installation due to faulty installation practices, in spite of explicit verbal and written installation instructions provided to the owners and their flooring contractors. Examples of methods verifying faulty installations are discussed and successful remedies are presented, along with respective passing test results.

10:50

5aAA9. Sound isolation of large transformers mounted on raised floors. Angelo J. Campanella (Campanella Assoc., 3201 Ridgewood Dr., Hilliard, OH 43026, a.campanella@att.net)

Transformers for substation service mounted on elevated floors often cause spurious sound to be transmitted to the spaces below it. Magnetostriction vibration of the massive iron core at 120 Hz and its harmonics drives the supporting floor to emit sound into the spaces below. Contemporary vibration isolation design fails to prescribe the proper isolation method since the mass of that transformer considerably exceeds that of the participating mass of the concrete floor. A vibration isolation paradigm is proposed of a two-body system in inertial space; the participating mass of the concrete floor being the second active body; the virtual mass of the pair along with the spring stiffness determines the resonance frequency. The ratio of that resonant frequency to that to be isolated is a measure of the expected isolation. For good isolation, the resonant frequency should be 1/10th that of the vibration to be isolated. The floor participating mass is approximated as a square one-half wavelength of the thin concrete slab bending wave. Recent case history supporting this model is described.

11:05

5aAA10. Archaeoacoustics: Testing and evaluation methodology. Vincent C. Paladino (PO Box 29 Lodi, Lodi, NJ 07644, bioenginearing@gmail.com)

Testing and measurement procedures of acoustic properties in caves, caverns, and buildings have been conducted along with EEG measurements in order to determine the effects of acoustic resonance on the human brain. The results have given rise to questions and speculative models concerning the role of acoustic phenomena in the development of human culture. Conceptualizations of the role of sound in sociocultural evolution add needed dimensions to archaeology, and challenge the science of acoustics. The problem of methodology needs to be addressed, and standards must be established. The relationship of testing techniques and the relevance of measurement parameters to the spaces and material culture found in these locations is central to the successful establishment of archaeoacoustics as a valid scientific study.

11:20

5aAA11. VampireVerb: A surreal simulation of the acoustics of Dracula's Castle. Alex Chechile, Constantin Basica, Elliot K. Canfield-Dafilou, and Jonathan S. Abel (Ctr. for Comput. Res. in Music and Acoust. (CCRMA), Stanford Univ., 660 Lomita Ct., Stanford, CA 94305, chechile@ccrma.stanford.edu)

This work introduces VampireVerb, a parameterized modal reverberator created from acoustics measurements recorded at Bran Castle in Transylvania, Romania, popularly called "Dracula's Castle." With limited evening access to the space, balloon pop sources were recorded using an A-format microphone and handheld recorder in several acoustically interesting rooms, including the armory. Spatial impulse responses were estimated from the balloon pop recordings, and a modal reverberator was designed to simulate the measured acoustics. In the main rooms, reverberation times T30 in the range [0.7 s, 1.2 s] were noted. A long, narrow stairwell and the torture chamber were acoustically dry, and did not produce the sound one would typically associate with Dracula's Castle. To bridge this gap between the actual and imagined acoustics, the mode frequencies and dampings were parametrized so as to interactively vary the size of the space and materials present, transforming the recorded rooms into large, stone spaces with the long reverberation times one would expect in Dracula's Castle. Finally, a nonlinear process was embedded in the reverberator architecture, producing a reverberant noise process modulated according to the evolving input signal spectral envelope, thus making VampireVerb both a traditional reverberator, and an instrument for compositional and performance applications.

FRIDAY MORNING, 8 DECEMBER 2017 SALON A/B/C, 9:00 A.M. TO 10:15 A.M.

Session 5aAO

Acoustical Oceanography: Topics in Acoustical Oceanography

Carolyn M. Binder, Cochair
Defence R&D Canada, P.O. Box 1012, Dartmouth, NS B2Y 3Z7, Canada

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

Contributed Papers

9:00

5aAO1. Confounding effects of environment-dependent propagation on automated classification of cetaceans. Carolyn M. Binder (Defence R&D Canada, P.O. Box 1012, Dartmouth, NS B2Y 3Z7, Canada, carolyn.binder@drdc-rddc.gc.ca) and Paul C. Hines (Dept. of Elec. and Comput. Eng., Dalhousie Univ., Halifax, NS, Canada)

Significant effort has been made over the last few decades to develop automated passive acoustic monitoring (PAM) systems capable of classifying cetaceans at the species level. The utility of such systems depends on their ability to operate across a wide range of ocean acoustic environments; however, anecdotal evidence suggests that site-specific acoustic propagation characteristics impact the performance of PAM systems. This is because properties of the ocean acoustic environment can be markedly different between regions and seasons in which PAM is used to observe cetaceans. Variability in propagation characteristics leads to differences in how each cetacean vocalization is distorted as it propagates along the source-receiver path. Unless these differences are accounted for, the acoustic environment will bias estimates of a PAM system's performance. The impact of environment-dependent propagation on an aural classifier was assessed using a pulse propagation model. Simulations were conducted that showed the sensitivity of classifier performance to sound speed profile type. Furthermore, it was found that classifier performance was range-dependent, largely due to changes in signal-to-noise ratio (SNR); however, in some environments only 60% of the performance reduction was attributed to decreasing SNR, indicating that signal distortion—resulting from environment-dependent propagation—had a large impact on performance.

9:15

5aAO2. Calculating a sensitivity kernel for the final cutoff of received energy in a broadband channel impulse response. Matthew Dzieciuch and Bruce Cornuelle (SIO/UCSD, 9500 Gilman Dr., IGPP-0225, La Jolla, CA 92093-0225, mad@ucsd.edu)

The final cutoff of a deep-water ocean acoustic tomography reception is often interpreted through a modal analysis. An alternative approach is presented here. First, a stable travel-time is generated by using an estimator-correlator to account for the statistical variability of the final low-angle arrival with the assumption of an appropriate coherent bandwidth and depth coherence. A time-series of these travel-times can then be tracked using the Viterbi algorithm. Then, the resolvable, trackable peaks are interpreted using the travel-time sensitivity kernel (TSK). The TSK has been previously used to account for the full-wave effects of ray-like arrivals. But the TSK machinery can be adapted to any processing strategy with the application of the chain-rule. The question of how to interpret an arrival produced by an Estimator-Correlator vs. an arrival produced by standard processing is examined. The feasibility of this approach is first analytically determined with the Pekeris waveguide and then an example from a recent ocean acoustics experiment is presented.

9:30

5aAO3. Whispering gallery waves in horizontal plane in vicinity of curved isobaths in shallow water. Boris Katsnelson (Marine GeoSci., Univ. of Haifa, Mt. Carmel, Haifa 31905, Israel, bkatsnels@univ.haifa.ac.il) and Pavel S. Petrov (V.I.Il'ichev Pacific Oceanological Inst., Vladivostok, Russian Federation)

Interference structure of the sound field in horizontal plane in shallow water with parameters variable in horizontal plane is studied within the framework of the so-called 3D problem. Formally, this problem can be solved using separation of the vertical coordinate (depth) in the wave equation in supposition that depth dependence of the sound field is described by adiabatic waveguide modes. Remaining part is two-dimensional dispersive wave equation which can be solved by different methods (PE, ray approximation, and modal decomposition). It is shown in the paper that in a shallow water waveguide with variable bathymetry with curvilinear isobaths (Laguna, lake) there exist specific solutions of this equation, concentrated in horizontal plane approximately along isobath lines (whispering gallery waves), up to formation of the waveguide modes in the horizontal plane. Number of modes and their shape depend on position of the source, frequency and radius of curvature. Remark that this whispering gallery modes can exist in real conditions, for example, for frequency about a few hundreds Hz and radius of curvature about 5–10 km. Analytical expression and results of modeling using ray approximation and PE are presented for real shallow water conditions. [Work was supported by ISF, grant 565/15.]

9:45

5aAO4. Data-driven decomposition of long-term echosounder time series from ocean observatories. Wu-Jung Lee (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, wjlee@apl.washington.edu), Valentina Staneva, Bernease Herman (eScience Inst., Univ. of Washington, Seattle, WA), and Aleksandr Aravkin (Dept. of Appl. Mathematics, Univ. of Washington, Seattle, WA)

Recent advances in technology have produced a deluge of acoustic data that offer opportunities to study ecological processes at scales that are not possible previously. A prominent example is the continuous data flow from the fleet of upward-looking echosounders installed by the Ocean Observatories Initiatives (OOI) at diverse global locations. However, it is unclear if conventional echosounder analysis routines are effective in analyzing these data sets, due to the generally scarce ground-truth resources and limited empirical knowledge at many locations. In this study, we explored the use of signal decomposition methods in discovering daily patterns of marine organism activities in long-term echosounder time series. Using non-negative matrix factorization (NMF), we show that the echograms can be decomposed into a weighted combination of discrete components, each with acoustic features that can be exploited for inferring the underlying biological assemblage. In addition, the component weights provide an avenue to capture echogram variations in a significantly reduced dimensional space, which can be utilized to describe changes in the ecosystem. Building on these results, we are developing new formulations to incorporate continuity in both time and space to construct a fully scalable framework for data-driven discovery using long-term echosounder data.

10:00

5aAO5. Cross frequency cross mode coherences using transport theory. Sivaselvi P, Tarun K. Chandrayadula (Ocean Eng. Dept., IIT Madras, Chennai, Tamil Nadu 600036, India, oe15d008@smail.iitm.ac.in), and John A. Colosi (Dept. of Oceanogr., Naval Postgrad. School, Monterey, CA)

Internal waves induced acoustic scattering cause mode coupling. Previous studies on mode scattering effects used narrowband approaches to predict mode energies and time coherences. However, these single-frequency predictions do not explain statistics such as time-wander and multipath spread. In order to address this issue, this paper develops a broadband scattering model that uses the frequency coherences of the modes. For the frequency coherence predictions, this paper uses the physics based transport theory approach. The predictions are setup for an environment similar to the Philippine Sea deep water experiment, which used frequencies around 250 Hz, bandwidth of 100 Hz, and propagation ranges from ~150 km to ~450 km. The physics based predictions are compared with complementary coupled mode based Monte-Carlo simulations. The coherences are used to predict two types of statistics. The first models the time spread for each respective mode. The second uses the cross-frequency cross-mode coherences to predict the stability of the ray time-fronts. The model uses up to mode 75 to predict the multipath time spread in the finale region and a few preceding ray-like arrivals. For the finale region, this paper uses the first 20 modes and the preceding rays 75 modes.

FRIDAY MORNING, 8 DECEMBER 2017 BALCONY M, 9:00 A.M. TO 11:45 A.M.

Session 5aBA

Biomedical Acoustics: Therapeutic Ultrasound

Douglas Miller, Chair
Radiology, Univ. Michigan, 3240A Medical Science I, 1301 Katherine Street, Ann Arbor, MI 48109-5667

Contributed Papers

9:00

5aBA1. Ultrasonic cavitation-enabled treatment for therapy of hypertrophic cardiomyopathy: Proof-of-principle. Douglas Miller, Xiaofang Lu, Chunyan Dou, Yiying I. Zhu, Mario L. Fabiilli (Radiology, Univ. Michigan, 3240A Medical Science I, 1301 Katherine St., Ann Arbor, MI 48109-5667, douglm@umich.edu), Gabe E. Owens (Pediatrics Cardiology, Univ. of Michigan, Ann Arbor, MI), and Oliver D. Kripfgans (Radiology, Univ. Michigan, Ann Arbor, MI)

Ultrasound myocardial cavitation enabled treatment (MCET) creates scattered microlesions in the myocardium, which can be accumulated to produce a desired macrolesion. MCET was applied to the SS-16^{BN} rat model of hypertrophic cardiomyopathy (HCM) for proof-of-principle as a means for myocardial reduction. A focused ultrasound transducer was targeted using 10 MHz imaging (10S, GE Vivid 7) to the left ventricular wall of anesthetized rats in a warmed water bath. Pulse bursts of 4 MPa peak rarefactional pressure amplitude were intermittently triggered 1:8 heartbeats during 10 min infusion of a microbubble suspension. Methylprednisolone was given to reduce initial inflammation and Losartan was given to improve healing. MCET significantly reduced the targeted wall thickness ($n = 11$) at 28 d post treatment by 16.2% ($P < 0.01$) relative to shams ($n = 8$), with myocardial strain rate and endocardial border displacement reduced by 34% and 29%, respectively. This demonstrates sufficient effect for a therapeutic outcome similar to surgical myectomy or alcohol ablation. Premature ECG complexes and plasma troponin measurements at the time of treatment were found to be useful to gauge optimal and suboptimal treatments, and thus aid in achieving a desired impact. With clinical translation, MCET therapy should fill the need for a new non-invasive HCM therapy option.

9:15

5aBA2. An efficient boundary element solver for trans-abdominal high-intensity focused ultrasound treatment planning. Pierre Gélat, Seyyed Reza Haqshenas (UCL Mech. Eng., London, United Kingdom), Timo Betcke (UCL Dept. of Mathematics, London, United Kingdom), Elwin van 't Wout (Pontificia Universidad Católica de Chile, Santiago de Chile, Chile), and Nader Saffari (UCL Mech. Eng., London, United Kingdom, n.saffari@ucl.ac.uk)

High-intensity focused ultrasound (HIFU) is a promising treatment modality for the non-invasive ablation of pathological tissue in many organs, including the liver. Since many patients are not suitable candidates for liver surgery, the possibility to locally deposit thermal energy in a non-invasive way would bear significant clinical impact. Optimal treatment planning strategies based on high-performance computing numerical methods are expected to form a vital component of a successful clinical outcome in which healthy tissue is preserved and optimal focusing achieved, thus compensating for soft tissue heterogeneity and the presence of ribs. The boundary element method (BEM) is an effective approach for this purpose because only the boundaries of the ribs and soft tissue regions require discretization, as opposed to standard approaches which require the entire volume around the ribcage to be meshed. A Galerkin discretized Burton-Miller formulation used in combination with preconditioning and matrix compression techniques was applied to compute the acoustic field generated by a focused array transducer in the presence of a layer of abdominal fat, a human ribcage model and liver tissue. The results demonstrate the effectiveness of this dedicated BEM algorithm for trans-abdominal HIFU treatment planning.

9:30

5aBA3. Pulse inversion therapy for improved monitoring of blood-brain barrier opening. Antonios Pouliopoulos, Mark Burgess, and Elisa Konofagou (Biomedical Eng., Columbia Univ., 630 West 168th St., Physicians & Surgeons 19-418, Columbia University Medical Campus, New York City, NY 10032, a.pouliopoulos@columbia.edu)

Microbubble-based ultrasound therapy has enabled non-invasive and reversible opening of the blood-brain barrier (BBB). However, the skull limits our ability to monitor microbubble activity due to high attenuation and beam aberrations. In ultrasound imaging, pulse inversion is used to cancel echoes from linear scatterers by summing the signal obtained from consecutive positive (phase: 0^o degrees) and negative (phase: 180^o degrees) pulses, thus facilitating imaging of non-linear scatterers such as microbubbles. Here, we adapt the pulse inversion technique to improve monitoring of BBB opening, by transmitting consecutive therapeutic pulses of inverse polarity. Pulse inversion therapy (PIT) was achieved by synchronizing the emission of inverted short pulses (pulse length: 2–3 cycles, PRF: 2 kHz, and pressure: 400 kPa) through a focused 0.5 MHz therapeutic transducer driven by two function generators. A concentric P12-5 linear array was used to passively capture the microbubble emissions. Absolute time-of-flight information was introduced in the beamforming since emission and reception were synchronous. PIT suppressed the signals from linear scatterers within the focal region by up to 6 dB in a gelatine phantom containing microbubbles and in mice *in vivo*, compared to positive-only pulses. Ongoing *in vivo* work aims at correlating the BBB opening with the microbubble signal identified by PIT.

9:45

5aBA4. Investigation of molecular mechanisms induced by a combination of high-intensity focused ultrasound and chemical anti-cancer agents. Heng Yu, Hakm Y. Murad, Daishen Luo (Biomedical Eng., Tulane Univ., 6823 St. Charles Ave., 440 Lindy Boggs Bldg., New Orleans, LA 70118, hyu3@tulane.edu), Andrew Sholl (Dept. of Pathol., TulaneUniv., New Orleans, LA), and Damir Khismatullin (Biomedical Eng., Tulane Univ., New Orleans, LA)

Tumor exposure to a combination of chemical treatment and physical forces causes chemical modifications and structural changes in cancer cells which inhibit their ability to survive and proliferate. High-Intensity focused ultrasound (HIFU), a unique non-invasive and non-ionizing physical force modality, has great potential as an adjuvant for chemical anti-cancer therapy. *In vitro* and *in vivo* studies conducted in our laboratory showed that HIFU synergistically enhanced tumor destruction by ethanol injection or chemotherapeutic drugs and reduce the potential for tumor metastasis. The objective of this study was to investigate the molecular mechanisms behind adjuvant anti-cancer effects of HIFU. To do this, we employed a combination of flow cytometry and western-blot analysis as well as *in vitro* and *in vivo* assays of tumor growth and cell adhesion. We showed HIFU drastically reduced the metastatic potential of chemically treated cancer cells via overproduction of heat-shock protein 70 (HSP70), death receptor Fas, its ligand FasL and TNF-α receptor. Although Hsp70 plays a key role in cancer initiation and progression, its overproduction induced by HIFU interferes with NF-κB signaling, thereby causing apoptosis and reduced expression of adhesion molecules required for metastasis. All these factors lead to phenotypic changes in surviving cancer cells that reduce their aggressiveness.

10:00

5aBA5. Ultrasound-enhanced molecular therapy for axon neurogenesis. Asis Lopez, Adrian Jones, Bridget Daugherty, and Damir Khismatullin (Biomedical Engineering/Bioinnovation, Tulane Univ., 6823 St. Charles Ave., New Orleans, LA 70118, alopez12@tulane.edu)

An estimated 55M individuals experience spinal cord injuries (SCIs), 230K in the United States. When a nerve is injured, the environment prevents healing of neurons and myelin. This is due to the presence of myelin inhibitors, growth factor not re-expressing, and glial tissue scarring rapidly. In order to bypass this mechanism nerves need to regenerate quickly, and unfortunately, no effective method currently exists to stimulate neurogenesis to the central nervous system. We propose an innovative non-invasive method to stimulate neurogenesis through low-intensity focused or unfocused ultrasound irradiation. We developed an *in vitro* system integrating unfocused ultrasound (UFUS) with neuron axon dynamics using a 3D microenvironment for neurogenesis. Dorsal root ganglion (DRG) neurons are used for our studies and microsurgically removed at day 15 and E18 Sprague Dawley cortical neurons from BrainBits, LLC. To determine if stimulation of axon growth occurs, we measure density and distance using phase contrast imaging and perform Scholl Method. In parallel, we began an *in vivo* protocol exposing the spinal cord of mice and using two-photon microscopy. We tested and demonstrated a variety of operating parameters to identify optimal conditions that stimulate axon DRGs by UFUS.

10:15–10:30 Break

10:30

5aBA6. *In vivo* investigation of high-intensity focused ultrasound combined with thermally triggered chemotherapy for liver cancer treatment. Gray Halliburton, Hakm Y. Murad, Monica Kala (Biomedical Eng., Tulane Univ., 31 Mcalister Dr., New Orleans, LA 70118, ghallibu@tulane.edu), Yueheng Zhang (Chemical and Biomolecular Eng., Tulane Univ., New Orleans, LA), Andrew Sholl (Pathol. and Lab. Medicine, Tulane Univ. School of Medicine, New Orleans, LA), Vijay John (Chemical and Biomolecular Eng., Tulane Univ., New Orleans, LA), and Damir Khismatullin (Biomedical Eng., Tulane Univ., New Orleans, LA)

High-Intensity Focused Ultrasound (HIFU) induces rapid tissue heating and mechanical disruption and also provides noninvasive release of chemotherapy from temperature-sensitive-liposomes (TSLs). We evaluated the combination of Sorafenib-loaded TSLs (SfTSLs) and HIFU via analysis of cell viability, tumor growth, and long-term survival. Liposomes were encapsulated with Sorafenib at 10μM with a glass transition temperature of 43°C. Hep3B human liver cancer cells were placed in PCR tubes to mimic dense tumor aggregates. Aggregates were treated with HIFU alone, SfTSLs alone, or SfTSLs + HIFU. HIFU exposure was applied for 30 seconds at acoustic output powers of 8.7 and 12W. Cell viability and proliferation were measured over 4 days post-treatment via AnnexinV/PI and WST-8 staining. Xenografted tumors were created via injection of 1.0×10^6 Hep3B cells within Matrigel into flanks of athymic-nude mice. Tumors at 8–10 mm were separated into the following groups: Control (sham), HIFU, SfTSLs, and SfTSLs + HIFU. Tumors were measured daily with endpoints at 5 days, 14 days, or long-term survival. Dissected tumors were H&E stained and evaluated by a blinded pathologist. The *in vitro* data indicates that Hep3B cells exposed to SfTSLs + HIFU have a significantly lower viability and

proliferation rate. Similarly, *in vivo* results show SfTSLs + HIFU reduces tumor growth and increases survival.

10:45

5aBA7. Focused-ultrasound mediated anti-alpha-synuclein antibody delivery for the treatment of Parkinson's disease. Hairong Zhang, Carlos S. Sanchez, Nancy Kwon (Dept. of Biomedical Eng., Columbia Univ., 630 West 168th St., Physicians & Surgeons 19-418, New York, NY 10032, hz2440@columbia.edu), Vernice R. Jackson-Lewis, Serge Przedborski (Dept. of Neurology, Columbia Univ., New York, NY), and Elisa Konofagou (Dept. of Biomedical Eng., Columbia Univ., New York, NY)

Parkinson's disease (PD) is associated with the selective death of dopaminergic (DA) neurons in the substantia nigra pars compacta (SNpc). While the specific cause of the neuronal loss remains elusive, the abnormal accumulation of alpha synuclein (α-syn), a major constituent of Lewy bodies, is considered to play a central role in the pathology of PD. Previous studies have shown the potential of immunotherapy with antibodies against α-syn, but such treatments remain ineffective due to the presence of the blood-brain barrier (BBB), which hinders most therapeutic agents to diffuse to the brain parenchyma. Therefore, in this study, we used focused ultrasound (FUS) in conjunction with microbubbles to transiently and noninvasively open the BBB and deliver anti-α-syn monoclonal antibodies (mAb) to the brains of transgenic PD mouse models. Preliminary histological findings demonstrate that the FUS promotes the delivery of anti-α-syn mAb to the transgenic mice overexpressing the human A53T or A30P α-synuclein. We hypothesize that weekly FUS treatments with mAb will help ameliorate histopathological deficits such as alpha-synuclein and Lewy bodies represented in these mouse models of PD. Ongoing work aims to explore the potential of FUS-mediated drug delivery to achieve both neuroprotection and neurorestoration for the treatment of Parkinson's disease.

11:00

5aBA8. Variation of high intensity therapeutic ultrasound pressure field characterization: Effects of hydrophone choice, nonlinearity, spatial averaging, and complex deconvolution. Yunbo Liu, Keith A. Wear (FDA, 10903 New Hampshire Ave., WO62RM2126, Silver Spring, MD 20993, yunbo.liu@fda.hhs.gov), and Gerald Harris (FDA, Rockville, MD)

Reliable acoustic characterization is fundamental for patient safety and clinical efficacy during high intensity therapeutic ultrasound (HITU) treatment. Technical challenges, such as measurement variation and signal analysis, still exist for HITU exposimetry using ultrasound hydrophones. In this work, four hydrophones were compared for pressure measurement: a robust needle hydrophone, a small PVDF capsule hydrophone, and two different fiber-optic hydrophones. The focal waveform and beam distribution of a single element HITU transducer (1.05 MHz and 3.3 MHz) were evaluated. Complex deconvolution between the hydrophone voltage signal and frequency-dependent complex sensitivity was performed to obtain pressure waveforms. Compressional pressure (p_+), rarefactional pressure (p_-), and focal beam distribution were compared up to 10.6/−6.0 MPa (p_+/p_-) (1.05 MHz) and 20.65/−7.20 MPa (3.3 MHz). The effects of spatial averaging, local nonlinear distortion, complex deconvolution, and hydrophone damage thresholds were investigated. This study showed a variation of at least 10-15% between different hydrophones during HITU pressure characterization.

11:15

5aBA9. Development of a treatment simulation platform—Application to probe optimization and tissues characterization. Raphaël Loyet (Univ. Lyon, Université Lyon 1, INSERM, LabTau, 151 Cours Albert Thomas, Lyon 69424, France, raphael.loyet@inserm.fr), Sylvain Chatillon (CEA, LIST, Gif sur yvette, France), Françoise Chavrier (Univ. Lyon, Université Lyon 1, INSERM, LabTau, Lyon, France), loic sifferlen (CEA, LIST, Gif sur yvette, France), Ayache Bouakaz (INSERM U930, Tours, France), Stéphane Leberre, and Pierre Calmon (CEA, LIST, Gif sur yvette, France)

In order to ease the development and optimization of new transducers and therapeutic protocols, a unified tool to easily simulate 3D pressure fields created by HIFU in human tissues has been developed jointly by INSERM and CEA-LIST. This includes three propagation algorithms: 1/ a GPU implementation of Rayleigh integral for fast linear propagation in human soft tissues, 2/ a dynamic ray tracing algorithm for the simulation of propagation in heterogeneous or inhomogeneous media (i.e., bones, skull, heated tissues, etc.), 3/ a Westervelt solver for non-linear propagation. These algorithms have been validated and implemented in a unified graphical user interface and can be easily interchanged to compare the results or to target a specific research topic. A finite volume difference solver for Bio Heat Transfer Equation (BHTE) has also been implemented. It can be used with all the HIFU propagation algorithms. It is used to simulate thermal effects of HIFUs and thermal dose in tissues and phantoms. Parametric studies targeting probe optimization and better characterization of tissues will be presented. [Work supported by French Nation Research Agency (ANR SATURN -15-CE19-0016).]

11:30

5aBA10. High-intensity focused ultrasound is synergistic with anti-neoplastic drugs that target endoplasmic reticulum stress. Hakm Y. Murad, Emma P. Bortz (Biomedical Eng., Tulane Univ., 6823 St. Charles, New Orleans, LA 70118, hmurad@tulane.edu), Partha Chandra, Debasis Mondal (Pharmacology, Tulane Univ., New Orleans, LA), and Damir Khismatullin (Biomedical Eng., Tulane Univ., New Orleans, LA)

Previous studies from our laboratory show that combination of High-Intensity Focused Ultrasound (HIFU) and anti-neoplastic agents not only reduces cellular viability and proliferation, but also shifts cellular populations to late apoptotic/necrotic stages. Nelfinavir, a well-tolerated HIV protease-inhibitor, is known to increase endoplasmic reticulum (ER) stress, and is thus being repurposed as an anti-neoplastic agent. This led us to combine Nelfinavir with HIFU in order to create a safe and more effective treatment for patients with prostate cancer. 2.7 million cells/ml suspensions of DU145 and C4-2B prostate cancer cells were placed in PCR tubes and split into 4 treatment groups: Control, HIFU alone, Nelfinavir(2μM) alone, and Nelfinavir(2μM) + HIFU. HIFU treatment was applied for 30 seconds at an acoustic output power of 8.7 W. Apoptosis and necrosis of cells was measured via AnnexinV/PI staining, cell proliferation was measured by a WST-8 assay, and ROS was stained using cellular ROS detection assay kit. The population percentage of Late Apoptotic/Necrotic cells and ROS production was found to be significantly greater, while proliferation rate was significantly reduced post exposure to Nelfinavir + HIFU 24 and 72h post treatment. HIFU and Nelfinavir have a synergistic effect on both increasing cellular death via apoptosis/necrosis and decreasing proliferation *in vitro*.

Session 5aPA

Physical Acoustics, Structural Acoustics and Vibration: Nonlinear Elasticity in Geomaterials

Marcel Remillieux, Cochair
Geophysics Group (EES-17), Los Alamos National Laboratory, Mail Stop D446, Los Alamos, NM 87545

Pierre-yves Le Bas, Cochair
Geophysics Group (EES-17), Los Alamos National Laboratory, Mail Stop D446, Los Alamos, NM 87545

Invited Paper

8:00

5aPA1. Stress corrosion crack imaging in stainless steel using the time reversed elastic nonlinearity diagnostic and nonlinear resonant ultrasound spectroscopy. Brian E. Anderson, Sarah M. Young, Stephen M. Hogg, and Joshua F. Gregg (Dept. of Phys. and Astronomy, Brigham Young Univ., N283 ESC, Provo, UT 84602, bea@byu.edu)

Spent nuclear fuel is often stored in stainless steel canisters in the United States. Stainless steel is susceptible to Stress Corrosion Cracking (SCC). This presentation will discuss progress on the use of the Time Reversed Elastic Nonlinearity Diagnostic (TREND) and Nonlinear Resonant Ultrasound Spectroscopy (NRUS) to determine whether SCC is present and attempt to quantify the depth of the cracking. NRUS is the measurement of the amplitude dependence of a sample's resonance frequency, which occurs because of a softening of the elastic modulus in damaged media. NRUS provides a global indication of damage in a sample. TREND employs time reversal acoustics, which focuses wave energy at various points of interest to excite localized high amplitude. The amplitude dependence of this localized energy allows pointwise inspection of a sample. [This work was funded by the Nuclear Energy University Program of the U.S. Department of Energy through a subcontract from Los Alamos National Laboratory.]

Contributed Papers

8:20

5aPA2. Nonuniform electromechanical efficiency and point defects in lithium niobate. Chandrima Chatterjee and Igor Ostrovskii (Dept. of Phys. and Astronomy, Univ. of Mississippi, 108 Lewis Hall, PO Box 1848, University, MS 38677-1848, cchatter@go.olemiss.edu)

The electromechanical efficiency (EME) and distribution of point defects (DPD) are measured from $LiNbO_3$ samples at room temperature. The EME is estimated by reading an electric potential (V) generated while applying a mechanical force to a local point. The DPD is measured by Photoluminescence (PL) taken from samples. PL is excited by 310-nm light and optical spectra are registered from 350 to 900 nm. The defects are identified by their characteristic peaks. The applied mechanical force and intensity of 310-nm light are kept constant while scanning V and PL along Z-axis or normal to Z. The F-center and Fe^{+} have been shown to be sensitive to local electrical polarization. Thus, they may influence EME in a local point of crystal. The experimental data from bulk and wafer samples show a nonuniform distribution of the F-center and Fe^{+} defects. For example, a 1.9-mm-thick Y-cut wafer demonstrates periodicity in defect concentration of 1 to 2 mm along the Z-axis. Similar periodicity is detected in the potential V along the same crystallographic axis. The revealed correlation between defects and EME along with their nonuniform distribution may be a physical basis of nonlinear phenomena involving piezoelectricity including nonlinear elasticity, acoustical memory, etc.

8:35

5aPA3. Three-dimensional description of nonlinear elasticity in rocks: Modeling and experiments. Marcel Remillieux (Los Alamos National Lab., Geophys. Group (EES-17), Mail Stop D446, Los Alamos, NM 87545, mcr1@lanl.gov), Martin Lott, Vincent Garnier (LMA, CNRS, UPR 7051, Aix-Marseille Université, Centrale Marseille, Marseille, France), Pierre-yves Le Bas, Timothy J. Ulrich (Los Alamos National Lab., Los Alamos, NM), and Cedric Payan (LMA, CNRS, UPR 7051, Aix-Marseille Université, Centrale Marseille, Marseille, France)

We study theoretically and experimentally the mechanisms of nonlinear and nonequilibrium dynamics in geomaterials through dynamic acousto-elasticity testing. In the proposed theoretical formulation, the classical theory of nonlinear elasticity is extended to include the effects of conditioning. This formulation is adapted to the context of dynamic acousto-elasticity testing in which a low-frequency "pump"-wave induces a strain field in the sample and modulates the propagation of a high-frequency "probe"-wave. Experiments are conducted to validate the formulation in a long thin bar of Berea sandstone. Several configurations of the pump and probe are examined—the pump successively consists of the first longitudinal and first torsional mode of vibration of the sample while the probe is successively based on (pressure) P- and (shear) S-waves. The theoretical predictions reproduce many features of the elastic response observed experimentally, in particular, the coupling between nonlinear and nonequilibrium dynamics and the three-dimensional effects resulting from the tensorial nature of elasticity. [This work was supported by the French National Research Agency through the ENDE program (Grant No. ANR-11 RSNR 0009) and the U.S. Department of Energy through the Fossil Energy program (Grant No. FE-634-15-FY15).]

8:50

5aPA4. Nonlinear shear wave resonator consisting of a relaxing material. John M. Cormack and Mark F. Hamilton (Appl. Res. Labs., Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, jcormack@utexas.edu)

Soft materials such as rubbers, polymers, and tissue exhibit low shear wave speeds, facilitating the generation of shear waves with large acoustic Mach numbers. In addition to finite-amplitude effects that result from cubic nonlinearity, plane shear wave propagation in these materials is subject to frequency-dependent attenuation and dispersion that result from viscoelastic effects. A wave equation for plane shear waves in a relaxing material is obtained from a nonlinear Zener constitutive model that accounts for cubic nonlinearity as well as the attenuation and dispersion associated with relaxation. The wave equation is used to analyze a one-dimensional shear wave resonator comprised of a nonlinear relaxing material that is shaken at one end and free at the other. For excitation of the lowest mode the wave equation is approximated by an augmented Duffing equation, and the resulting frequency-response equation is compared with numerical finite-difference solutions of the original wave equation. Frequency responses are presented as functions of both drive amplitude and relaxation time. The model predicts behavior resembling that reported by Andreev *et al.* [*Acoust. Phys.* **57**, 779 (2011)] for experiments on a similar resonator comprised of plastisol and a head mass. [Work supported by the ARL:UT McKinney Fellowship in Acoustics.]

Invited Papers

9:05

5aPA5. Experimental measurements and bristle friction modeling of nonlinear hysteresis loops and harmonic generation in rock fractures. Seth Saltiel, Brian Bonner (Earth and Environmental Sci., Lawrence Berkeley National Lab, 1 Cyclotron Rd., Berkeley, CA 94720, ssaltiel@lbl.gov), Tushar Mittal, Brent Delbridge (Earth and Planetary Sci., UC Berkeley, Berkeley, CA), and Jonathan Ajo-Franklin (Earth and Enivironmental Sci., Lawrence Berkeley National Lab, Berkeley, CA)

Frictional properties affect the propagation of high-amplitude seismic waves across rock fractures. Laboratory evidence suggests that these properties can be measured in active seismic surveys, potentially offering a route to characterizing friction *in situ*. We present experimental results from a subresonance torsional modulus and attenuation apparatus that utilizes micron-scale sinusoidal oscillations to probe the nonlinear stress-strain relation at a range of strain amplitudes and rates. Nonlinear effects are further quantified using harmonic distortion; however, time series data best illuminate underlying physical processes. The low-frequency stress-strain hysteretic loops show stiffening at the sinusoid's static ends, but stiffening is reduced above a threshold frequency. This shape is determined by harmonic generation in the strain; the stress signal has no harmonics, confirming that the fractured sample is the source of the nonlinearity. These qualitative observations suggest the presence of rate-dependent friction and are consistent between fractures in three different rock types. We propose that static friction at the low strain rate part of the cycle, when given sufficient "healing" time at low oscillation frequencies, causes this stiffening cusp shape in the hysteresis loop. While rate-and-state friction is commonly used to represent dynamic friction, it cannot capture static friction or negative slip velocities. So, we implement a bristle friction model, which describes this process and produces similar results.

9:25

5aPA6. Nonlinear acoustics evaluation of CO_2 exposed sandstone. James Bittner (Univ. of Illinois, 205 N. Mathews St. MC-250, Urbana, IL 61801, jbittn2@illinois.edu), Pierre-yves Le Bas (Los Alamos National Lab., Los Alamos, NM), John Popovics (Univ. of Illinois, Urbana, IL), and Paul A. Johnson (Los Alamos National Lab., Los Alamos, NM)

In an effort to better understand the hygro-thermo-mechanical behavior of geologic CO_2 reservoir material, we investigate the nonlinear behavior of elastic wave propagation in Berea sandstone samples, which are used as a standard for reservoir rock formations. Nonlinear characterization methods, including resonant ultrasound spectroscopy (RUS), dynamic acousto-elasticity (DAET), and single-impact nonlinear resonance techniques, are applied to pristine, damaged (distributed microfractures), and CO_2 injected Berea samples; conventional linear vibrational and wave propagation measurements are also applied to the samples. The results of these sensitive test methods are compared to reveal the characteristics of geologic reservoir materials that are most affected by varying microstructural and environmental conditions. An analysis of the work also leads to potential bases for test methods that could be deployed in the field in the future to monitor the condition of reservoir formations and lead to better understanding of CO_2 injection-induced seismic events. This work is done within the framework of the $GSCO_2$ center for geologic storage of CO_2 from the U.S. department of Energy whose purpose is to better understand CO_2 sequestration to make it safer and more efficient. As such the results obtained by elastic waves measurement will also be compared to other testing, providing an insight on the physical origin of the nonlinear behavior of geomaterials.

9:45

5aPA7. Dynamic acousto-elastic testing with an ultrasonic tomography probe: Application to fractured rock characterization. Jacques Riviere and Philippe Roux (ISTerre, Inst. of Earth Sci., Grenoble Alpes Univ., ISTerre Université Grenoble Alpes CS 40700, Grenoble Cedex 9 38058, France, jacques.riviere@univ-grenoble-alpes.fr)

We study nonlinear elastic phenomena at the laboratory scale to help interpret the subtle velocity changes observed in the Earth's crust, for instance, during strong ground motion, earthquake slip processes or Earth tides. Dynamic Acousto-Elastic Testing (DAET) provides unprecedented details on the nonlinear elastic response of consolidated granular media (e.g., rocks, concrete), including tension/compression asymmetry, hysteretic behaviors as well as conditioning and relaxation effects. Such technique uses a pump-probe scheme where a high frequency, low amplitude wave probes the state of a sample that is dynamically disturbed by a low frequency, large amplitude pump wave. While previous work typically involved a single pair of probing transducers, here we use two dense arrays of ultrasonic transducers to image a sample of Westerly granite with a complex fracture. We apply double beamforming to disentangle complex arrivals and conduct ray-based and finite-frequency tomography using both travel time and amplitude information. By comparing images obtained before, during, and after the pump wave disturbance, we are able to locate and characterize nonlinear sources within the sample. We discuss their locations with regard to low velocity/high attenuation zones and relate our observations to large-scale data.

10:05–10:25 Break

Contributed Paper

10:25

5aPA8. Flaw detection of wellbore systems by combining time reversal methods and nonlinear acoustic measurements. Carly M. Donahue and Pierre-yves Le Bas (Los Alamos National Lab., PO Box 1663, Los Alamos, NM 87545, cmd@lanl.gov)

A wellbore system consisting of a steel pipe cased in cement is susceptible to cracking and delamination. Such flaws greatly reduce the integrity of the wellbore and can lead to significant leakage. To resolve the location and extent of the defects, we have combined a unique time reversal method with nonlinear acoustic evaluation. The time reversal method generates a compact high-amplitude wave only in a localized area that can be used to probe the acoustic properties spot by spot of a three dimensional region. Additionally, the nonlinear acoustic signal an object exhibits is far more sensitive to damage and defects than conventional linear analysis. The nonlinear behavior can be locally determined by observing the changes in elastic properties as the amplitude of the time-reversed focus wave is increased. The experimental setup consist of an array of piezoelectric transducers for generating the focused energy and a laser vibrometer mounted on a scanning frame for focusing and measuring nonlinearity on the interior of the steel tubing.

Invited Papers

10:40

5aPA9. Nonlinear softening of unconsolidated granular earth materials. Charles Lieou (Los Alamos National Lab., MS D446, Los Alamos, NM 87545, clieou@lanl.gov), Eric Daub (Ctr. for Earthquake Res. and Information, Univ. of Memphis, Memphis, TN), Robert Guyer (Physics, Univ. of Nevada Reno, Reno, NV), and Paul A. Johnson (Los Alamos National Lab., Los Alamos, NM)

Unconsolidated granular earth materials exhibit softening behavior due to external perturbations such as seismic waves, namely, the wave speed and elastic modulus decrease upon increasing the strain amplitude. In this letter, we describe a theoretical model for such behavior. The model is based on the idea that shear transformation zones (STZs)—clusters of grains that are loose and susceptible to contact changes and rearrangement—are responsible for plastic deformation and softening of the material. We apply the theory to experiments on simulated fault gouge composed of glass beads, and demonstrate that the theory predicts nonlinear resonance shifts and reduction of the P-wave modulus, in agreement with experiments. The theory thus offers insights on the nature of the critical state prior to failure on earthquake faults.

11:00

5aPA10. The role of nonlinear ultrasound in the diagnosis of early-stage damage in heterogeneous materials. Gun Kim (Carle Illinois College of Medicine, The Univ. of Illinois at Urbana-Champaign, 405 N. Mathews Ave., MC-251, Urbana, IL 61801-2325, mcguncall@gmail.com), Tae Sup Yun (School of Civil and Environ. Eng., Yonsei Univ., Seoul, South Korea), Jin-Yeon Kim, Kimberly Kurtis, and Laurence Jacobs (School of Civil and Environ. Eng. / G. W. Woodruff School of Mech. Eng., Georgia Inst. of Technol., Atlanta, GA)

Predictable noninvasive evaluation of engineering materials requires a more reliable sensing technique capable of providing quantitative information of early stage damage. Nonlinear ultrasound (NLU) is a promising candidate because it provides a direct measure of the nonlinear elastic behavior of materials. NLU excels in the direction and quantification of damage that originates at or beneath the material's microscale. This talk will present a procedure for the second harmonic generation (SHG) measurements using nonlinear Rayleigh surface waves. This technique quantifies material nonlinearity through the acoustic nonlinearity parameter, β. Specifically, microscale material characterization of physical/chemical phenomena in heterogeneous materials will be reviewed by means of the acoustic nonlinearity parameter, β. The results reveal how the SHG technique can provide the quantitative relationship between the acoustic nonlinearity parameter and the damage state of these materials. Last, new strategies for the application of the SHG technique will be discussed with an emphasis on bio-engineering materials and rocks.

Contributed Papers

11:20

5aPA11. The soil plate oscillator: Nonlinear mesoscopic elasticity of layers of glass beads in flexural vibration. Emily V. Santos and Murray S. Korman (Phys. Dept., U.S. Naval Acad., 572 C Holloway Rd., Annapolis, MD 21402, santosemily08@gmail.com)

The soil plate oscillator (SPO) apparatus consists of two circular flanges sandwiching and clamping a thin circular elastic plate. In this presentation, uniform spherical glass beads—representing a nonlinear mesoscopic elastic material—are supported at the bottom by the plate and stiff cylindrical sidewalls of the upper flange. A small magnetic disk centered and fastened below the plate is driven by an AC coil placed below the magnet. Nonlinear tuning curves of the magnet's acceleration are measured by driving the coil with a swept sinusoidal signal applied to a constant current amplifier. Experiments are first performed to measure the peak resonant frequency vs. mass loading as glass beads are added to make a level column above the plate. Experiments are repeated using 1,2,3,…,10 mm diameter beads to demonstrate how the frequency first decreases and then eventually increases for each bead diameter. Next, ten (separate) tuning curve experiments are performed using a fixed column of 350 grams of beads for each diameter. The backbone curves (peak acceleration vs. corresponding resonant frequency) exhibit a linear region with comparable slopes. While the detailed curvature vs. bead diameter reveals more structure, a bilinear hysteresis model fits the results.

11:35

5aPA12. Broadband acoustic wave experiments in borehole configurations. Huajun Fan (College of Geophys. and Information Eng., China Univ. of Petroleum (Beijing), 18 Fuxue Rd., Changping, Beijing 102249, China, hj.fan@hotmail.com) and David Smeulders (Dept. of Mech. Eng., Eindhoven Univ. of Technol., Eindhoven, Netherlands)

A borehole in the ground may penetrate rocks which are porous, permeable, and fractured. Pressure transients in the borehole will therefore cause viscous fluid to flow into and out of the wall of the borehole. This forced flow consumes some energy and affects the phase velocity of the waves

traveling along the fluid column. These two effects can be evaluated to extract information about the rocks adjacent to the borehole, which is of paramount importance for the oil and gas industry. This work discusses laboratory wave experiments in a 7.5 m long vertical shock tube. Rock samples containing a vertical borehole and pre-fabricated fracture configurations are installed in the test section and filled with water. The shock tube generates broadband pressure transients in the borehole. These are measured in the borehole at variable depth by means of a sliding pressure probe. In this way borehole microseismograms are constructed from repetitive wave experiments. Borehole wave modes are identified using coherence analysis. It is found that the presence of fractures significantly influences the transmission and reflection of borehole waves over the fracture zone. Moreover, there is a strong correlation between fracture aperture and borehole acoustic properties.

FRIDAY MORNING, 8 DECEMBER 2017 STUDIOS FOYER, 9:00 A.M. TO 12:00 NOON

Session 5aSC

Speech Communication: Bilingual and Non-Native Speech Perception and Production (Poster Session)

Charlotte Vaughn, Chair
Linguistics, University of Oregon, 1290 University of Oregon, Eugene, OR 97403-1290

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

5aSC1. Context effects on accentedness ratings. Charlotte Vaughn and Melissa M. Baese-Berk (Linguist, Univ. of Oregon, 1290 University of Oregon, Eugene, OR 97403-1290, cvaughn@uoregon.edu)

Intelligibility and accentedness are largely independent judgments (Derwing & Munro, 1995); a speaker rated as highly accented can be quite intelligible. The context of listeners' exposure to accented speakers has been shown to affect adaptation to speakers in terms of intelligibility (e.g., Tzeng *et al*., 2016), suggesting that implicit comparison between examples facilitates learning of systematic properties of accented speech. However, it is unknown whether context affects accentedness ratings, which are often assumed to be stable properties of speakers. To better understand the susceptibility of accent ratings to context effects, the present study examines listeners' accentedness judgments of native and non-native speech taken from the ALLSSTAR Corpus (Bradlow *et al*., 2010). A target set of the same accented speech samples was embedded in a variety of task contexts, varying whether the stimuli were randomized among speakers or blocked by speaker, or presented at the beginning or end of the experiment. Results will shed light on how stable accentedness judgments are, and begin to test the extent to which these judgments are tied to particular items, speakers, accents, or listeners. Determining whether accentedness ratings are affected by context is a step toward understanding what factors listeners use when making accentedness ratings.

5aSC2. The perception of English Vowels by Korean Native Speakers: Concerning the influence of the manner of articulation for consonants following vowels. Heesun Han (Graduate School of Lang. and Culture, Osaka Univ., 1-8 Machikaneyama, Toyonaka, Osaka 560-0043, Japan, kenkyuhhs@gmail.com) and Takeshi Nozawa (Graduate School of Lang. and Culture, Osaka Univ., Kusatsu, Japan)

The purpose of this study is to explore the perception of English vowels by Korean native speakers. It focuses on difficulty levels of perception for English vowels in relation to the sound environment. The sound environment is determined by differences in the manner of articulation of consonants following vowels. The experiment consists of 24 test words formed with six American English vowels(/i, ɪ, ɛ, æ, ɑ, ʌ/) embedded in four different sound environments (/hVt/, /pVt/, /pVn/, /pVl/). The experiment compares two words paired for each sound environment, examining judgment accuracy for each combination and the influence of sound environments. Twenty Korean native speakers in their 20s or 30s participated in the experiment. The results found that participants could rarely differentiate /i/ and /ɪ/, and /ɛ /and /æ/ in all sound environments. Moreover, judgment accuracy was lower overall for the /pVn/ and /pVl/ environments. For /pVl/, in particular, Korean native speakers misjudged /ʌ/ as /ɑ/, confirming that difficulty levels differed according to the sound environments.

5aSC3. Second language English learners' perception of foreign-directed speech. Kathrin Rothermich (Speech, Lang. and Hearing, Univ. of Connecticut, 850 Bolton Rd., Unit 1085, Storrs, CT 06269-1085, kathrin.rothermich@uconn.edu), Kristin Mello, Emily Turco, Larissa Lemes, Erika Fernandez, and Susan Bobb (Psychology, Gordon College, Wenham, MA)

Previous research has shown that people adapt the way they speak depending on the perceived comprehension level of the listener (Uther *et al*., 2007) including when they speak to foreigners. While the acoustic properties of accommodations such as foreign speech are well documented, few studies have investigated how second language learners in particular interpret them. The present study investigated English language learners' (ELLs) perception of different speech types. Participants heard four types of auditory stimuli (casual, clear, infant-directed, and foreign-directed) spoken by four different speakers (two males, two females) and evaluated the extent to which the speaker was easy to understand, competent, condescending, friendly, and respectful. Perceptual ratings showed an interaction between speech type and question type ($p < 0.001$). Casual speech was least intelligible, least competent, least friendly, and least respectful compared to all other speech types. No effects were found for condescension. The implications of these results suggest that ELLs perceive speech accommodation of any type as positive.

5aSC4. Non-native perception of Vietnamese tone contrasts. Madeleine Oakley (Linguist, Georgetown Univ., 3700 O St. NW, Washington, DC 20057, mo643@georgetown.edu)

A listener's ability to discriminate non-native sound contrasts has been shown to be largely influenced by the listener's native phonological system (Best *et al*., 2003; Tyler *et al*., 2014; Tskuda, 2012). Looking specifically at

suprasegmental contrasts, experimental results suggest that the degree to which pitch is used to distinguish lexical items in a speaker's L1 influences how well they are able to detect non-native tone contrasts (Schaefer & Darcy, 2014). However, these results are based only on Thai tones. The present study shows that native speakers of English (a stress language), and Mandarin (a tone language) do not perform significantly differently in their ability to perceive Vietnamese tone contrasts. Results from an ABX categorization task show that native speakers of English are not more likely to make errors in categorizing Vietnamese tones than native speakers of Mandarin, and both groups have difficulty perceiving the difference between the low falling tone and the low falling-rising tone. These results suggest that the acoustic properties of a tone, such as the register and the contour, contributes more to how well non-native speakers can discriminate a contrast than does the L1 of the listener.

5aSC5. Training new second language category formation. Anna M. Schmidt and Nora Hassan (Speech Pathol. & Aud., Kent State Univ., A104 MSP, Kent, OH 444242, aschmidt@kent.edu)

The Speech Learning Model (Flege, 1995) hypothesizes that, in second language (L2) learning, the greater the difference the learner perceives between in a sound from a first language (L1) category and a sound from an L2 category, the more likely it is that a new L2 category will be formed. Finding an L1 category that is not similar in some way to an L2 category is difficult. However, a L2 sound, that is similar to an L1 sound, could be consciously taught as a new allophone of that sound in a way that focuses attention on phonetic differences. In this study, Chinese L1 speakers, from a region of China where /n/ and /l/ in word initial position in their dialect (and in Mandarin) are produced as /n/, will be taught a dark Arabic /l/ as a new position specific allophone to be used in word initial position in English and Mandarin Chinese. Results from perception and production of /l/ and /n/ before and after training will be reported.

5aSC6. Phonological neighborhood density and speech production in L1 and L2 English speakers. Veera S. Vasandani (Univ. of Minnesota, Minneapolis, MN), Melissa M. Baese-Berk (Linguist, Univ. of Oregon, Eugene, OR), Jennifer S. Kim, and Benjamin Munson (Univ. of Minnesota, 115 Shevlin Hall, Minneapolis, MN, munso005@umn.edu)

Phonological Neighborhood Density (PND) affects speech production latencies, word durations, and acoustic-phonetic detail (e.g., Baese-Berk & Goldrick, 2009; Buz & Jaeger, 2016; Fox, Reilly, & Blumstein, 2015; Munson, 2013; Munson & Solomon, 2004; Wright, 2004). The nature of these effects has been debated actively. Some have argued that these effects reflect intentional articulatory modifications to increase intelligibility, while others have suggested that they are the consequence of planning processes. The purpose of this presentation is to examine whether these effects are present in second-language speakers of English whose first language is Korean. Because the mechanisms underlying these effects are unclear, examining whether non-native speakers also show sensitivity to PND will inform our understanding of the mechanisms that drive them. Moreover, studying non-native speakers of varying proficiency allows us to infer whether PND-driven variation in production emerges over the course of second-language acquisition. The production of multiple repetitions of high- and low-PND CVC words was collected from 17 native adult speakers of English and 19 adult L2 speakers of English whose L1 was Korean. Estimates of vocabulary size were also collected for both groups. Analyses of production latencies and vowel acoustics are ongoing. [Funding provided by a University of Minnesota Multicultural Summer Research Opportunity Program award to the first author.]

5aSC7. Effects of clear speech and language background on multimodal perception of English fricatives. Sylvia Cho (Linguist, Simon Fraser Univ., 8888 University Dr., Burnaby, BC V5A 1S6, Canada, sylvia_cho@sfu.ca), Allard Jongman (Linguist, The Univ. of Kansas, Lawrence, KS), Yue Wang (Linguist, Simon Fraser Univ., Burnaby, BC, Canada), and Joan A. Sereno (Linguist, The Univ. of Kansas, Lawrence, KS)

Research shows that acoustic modifications in clearly enunciated fricative consonants (relative to the plain, conversational productions) facilitate auditory fricative perception. However, clear-speech effects on visual fricative perception have received less attention. A comparison of auditory and visual (facial) clear-fricative perception is particularly interesting since sibilant fricatives in English are more auditorily salient while non-sibilants are more visually salient. This study thus examines clear-speech effects on audio-visual perception of English sibilant and non-sibilant fricatives. Native English perceivers and non-native perceivers with different L1 fricative inventories (Mandarin, Korean) identified clear and plain fricative-vowel syllables in audio-only (AO), visual-only (VO), and audio-visual (AV) modes. The results across perceiver groups and speech styles showed an overall visual benefit and auditory dominance (AV>AO>VO). Comparisons of styles revealed clear-speech benefits in AO across fricatives and groups, but different patterns were noted in the visual conditions. In VO and AV, clear speech helped the more visually salient non-sibilant identification for native perceivers; however, clear-speech benefits were less prominent in non-natives' perception of the non-sibilants, which are non-existent in their L1s. These findings are discussed in terms of the relative audio-visual weighting that benefits perception in clear speech as a function of input saliency and perceiver experience.

5aSC8. L2 duration characteristics of Mandarin tones by Japanese learners. Yue Sun (Waseda Univ., Room 702, 3 Chome 14-9, Okubo, Shinjuku-ku, Tokyo 169-0072, Japan, yue.cherry.sun@gmail.com), Jinsong Zhang (Beijing Lang. and Culture Univ., Beijing, China), and Yoshinori Sagisaka (Waseda Univ., Tokyo, Japan)

In order to get a better understanding of the acquisition of Mandarin Chinese prosody by L2 learners, we investigate the temporal aspects of Mandarin tones in running speech for native Chinese speakers and Japanese L2 learners. Significant differences are found between natives and learners when generating tones with varying syllable durations depending on their position in the utterance as well as on specific syntax constraints associated with them. Among the tone types, the shortening of tone 3 pronouns at the beginning of utterances and the lengthening of tone 4 syllables at the end of utterances remain difficult for learners to generate, even at a higher overall proficiency level. Throughout the duration analysis, we could speculate the existence of stress control in Chinese syllable timing.

5aSC9. A comparative acoustic analysis of Korean vowels by native and non-native speakers. Na-Young Ryu (Linguist, Univ. of Toronto, 30 Charles St. West, Unit 903, Toronto, ON M4Y1R5, Canada, nayoung.ryu@mail.utoronto.ca)

The primary purpose of this study is to examine L1 transfer in L2 vowel acquisition (Flege 1995,1996) by comparing L1 vowels with L2 Korean vowels produced by Korean native speakers and both Mandarin and English learners of Korean. It would predict which L2 Korean vowels are relatively difficulty or easy to produce for the non-native speakers based on acoustic similarities and dissimilarities between their L1 vowels and L2 Korean vowels. A total of 68 female speakers participated in a word-list reading task. For acoustic analysis, the formant frequencies (F1 and F2) and vowel duration were measured. Results demonstrated that there were cross-language differences in both vowel quality and duration. Both Mandarin and English learners of Korean perform well when producing Korean [a, i] vowels, but have difficulty producing Korean vowel contrasts [ʌ]-[o], [o]-[u], [ɨ]-[u]. In terms of vowel duration, Korean vowels were the shortest, English vowels were the longest, and Mandarin vowels were intermediate between the two. Overall, the L2 Korean vowel duration of both Mandarin and English speakers was too short, compared to their L1 vowel productions, to have a Korean native-like performance of vowels.

5aSC10. Effect of frequency shifts on talker recognition in native and foreign-accented speech. Michelle R. Kapolowicz, Daniel R. Guest, Vahid Montazeri, and Peter F. Assmann (Behavioral and Brain Sci., The Univ. of Texas at Dallas, 800 West Campbell GR 4.1, Richardson, TX 75080, mrk092020@utdallas.edu)

The present study examined the effect of frequency shifts on perceived talker recognition in foreign-accented speech compared to native-accented speech. Sentences were processed using the STRAIGHT vocoder. The

spectral envelope and the fundamental frequency were shifted up or down in seven steps (3 up, 3 down plus unshifted) using scale factors of 8% and 30%, respectively, at each step. Listeners heard pairs of sentences and were asked to judge whether the identity of the talker was the same or different. Frequency shifts had similar effects for native- and foreign-accent conditions, in that listeners perceived the shifted versions as different talkers when, in fact, the talkers were the same. However, listeners were more likely to judge native-accented sentence pairs as the same talker regardless of whether or not they were the same; foreign-accented sentence pairs were more likely to be heard as different talkers. Overall, these results indicate that patterns of frequency-shifted foreign-accented speech are similar to previously reported patterns for frequency-shifted native speech; however, the small differences in patterns between the accent conditions might be attributed to listeners being less familiar with non-native speech patterns.

5aSC11. Towards automated detection of similarities and differences in bilingual speakers. Marisha Speights (Commun. Disord., Auburn Univ., 1199 Haley Ctr., Auburn, AL 36849, mls0096@auburn.edu), Noah H. Silbert (Commun. Sci. and Disord., Univ. of Cincinnati, Cincinnati, OH), Joel MacAuslan (Speech Technol. & Appl. Res., Bedford, MA), Rachel Blades, Donaldson Maya, Kara Swanson, Sarah Tuohy, JoHannah Ungruhe, and Karla Washington (Commun. Sci. and Disord., Univ. of Cincinnati, Cincinnati, OH)

Bilingualism is increasingly the norm in the United States with at least 20% of Americans being bilingual. This change in linguistic demography has created a new challenge to accurately diagnose speech disorders among bilingual children. This study explores Acoustic Landmark Detection (ALD) system as an objective approach to characterizing similarities and differences in speech production in child speakers of Standard English and Jamaican Creole (JC). Eight JC-English bilingual children were recorded speaking eleven words three times in each language. Words were transcribed and entered into PROPH+ to provide: (1) Phonological Mean Length of Utterance [pMLU], (2) phonotactic structure, and (3) Percent Consonants Correct (PCC). Landmarks were hand-marked to determine probable landmark sequences based on canonical word production. Canonical landmark sequences were aligned with detected landmark sequences in both languages using the Needleman-Wunsch global alignment algorithm. Analysis revealed that if PCC indicates JC and English words are different, the mismatch between landmark tends to be lower (higher production accuracy); if phonotactics indicate JC and English words are different, mismatch tends to be slightly higher (lower accuracy); and if pMLU indicates word differences, mismatch tends to be higher (lower accuracy). Our finding support traditional linguistic expectations regarding bilingual speakers' similarities and differences.

5aSC12. Acoustic correlates of accentedness and comprehensibility in L2 Spanish judgments: A mixed-effects modeling approach. Ziwei Zhou and Charles Nagle (Iowa State Univ., 2603 Kent Ave. # 209, Ames, IA 50010, ziweizh@iastate.edu)

Two constructs—comprehensibility and accentedness—figure themselves prominently in listeners' judgments of L2 (second language) speech. Correlational analyses have shown that they make separate contributions to such judgments (Derwing and Munro, 1995, 1997, 2005). As it is important to set realistic goals for adult L2 learners by prioritizing understanding over nativelikeness (Levis, 2005), recent studies began to examine the relative contribution of linguistic aspects, especially acoustic features, to comprehensibility and accentedness (Munro & Derwing, 1999; Kang, Rubin, & Pickering, 2010; Trofimovich & Issac, 2012). Consistent with this agenda, this study first examines the reliability of scores (produced by the 9-point Lickert scale) under modern measurement framework (e.g. multi-facet Rasch measurement model). Then, the relationship between acoustic features, including the confidence values (for each utterance) obtained using Google Cloud Platform speech recognition engine as well as suprasegmental features extracted using Prosogram (Mertens, 2014), and ratings are established using mixed-effects modeling techniques. Results indicate that, even though it can be shown that comprehensibility and accentedness are statistically distinct constructs, both ratings share a common set of acoustic correlates as significant predictors—speech time, standard deviation of pitch values, and normalized pairwise variability index (nPVI).

5aSC13. Audio-visual perception of mandarin tone in clear speech. Yuyu Zeng (Linguist, Univ. of Kansas, 1541 Lilac Ln., Lawrence, KS 66045, yzengae@ku.edu), Keith Leung, Yue Wang (Linguist, Simon Fraser Univ., Burnaby, BC, Canada), Allard Jongman, and Joan A. Sereno (Linguist, Univ. of Kansas, Lawrence, KS)

Clearly enunciated speech (relative to conversational, plain speech) involves articulatory and acoustic modifications that have been shown to enhance segmental intelligibility. However, little research has explored clear-speech effects on the perception of suprasegmental properties such as lexical tone, particularly involving visual (facial) perception. Moreover, research has not determined if visual tonal cues are linguistically significant, presumably because tone production does not rely on vocal tract configurations. Investigating clear-speech effects may help validate the contribution of visual information, since improved visual tone intelligibility in clear speech would indicate the linguistic relevance of visual cues. The present study tested this hypothesis by examining the intelligibility of clear and plain Mandarin tones by native (Mandarin) and non-native (English) perceivers with auditory/visual input modalities (AO: audio-only; VO: visual-only; and AV: audio-visual). Results showed small but significant clear-speech benefits in each modality for both natives and non-natives. However, while the natives revealed an overall visual gain and auditory dominance (AV>AO>VO), visual gain appeared to be less prominent in non-native perception (AV<AO>VO in plain, AV=AO>VO in clear). These results demonstrate clear-speech facilitation in visual as well as auditory perception, particularly with native perceivers, suggesting the existence of linguistically relevant visual tonal cues.

5aSC14. Neutralization of Taiwanese tone sandhi: An acoustic study. Yu-Fu Chien (Chinese Lang. and Lit., Fudan Univ., Rm. 701, West Main Guanghua Bldg., Number 220, Handan Rd., Yangpu District, Shanghai, China, whouselefthand@gmail.com) and Allard Jongman (Linguist, The Univ. of Kansas, Lawrence, KS)

Taiwanese tonal alternation is realized in a circular chain shift fashion for both smooth and checked syllables. Debate regarding the processes of Taiwanese tonal alternation has centered on whether a surface tone is derived from an underlying tone, or whether a surface tone is selected without undergoing any derivation. The current study investigates this controversial issue by examining Taiwanese checked tone and smooth tone neutralization in production. In particular, we analyzed whether smooth citation and sandhi tone 55 (T51→T55) are completely neutralized in F0 contour, F0 height, and duration. We further extended the Taiwanese neutralization literature by also comparing smooth citation and sandhi tone 21 (T33→T21), checked citation and sandhi tone 53 (CT21→CT53), and checked citation and sandhi tone 21 (CT53→CT21). Non-sandhi exceptions were also included to evaluate the effect of position-in-word on F0 height and duration given that citation tones always appear in phrase-final position. Complete neutralization would indicate a surface-tone-storage-and-access point of view, whereas incomplete neutralization would suggest a derivational account for the production of Taiwanese tonal alternation. Results showed complete neutralization for checked tones after factoring out the positional effect. Results for smooth tones and theoretical implications will be also be discussed.

Session 5aSP

Signal Processing in Acoustics: Topics in Acoustic Signal Processing

Juliette W. Ioup, Chair
Dept. of Physics, Univ. of New Orleans, New Orleans, LA 70148

Contributed Papers

8:00

5aSP1. Direction of arrival estimation for conformal arrays on real-world impulsive acoustic signals. Emily Gorman (Elec. and Comput. Eng., Mississippi State Univ., 406 Hardy Rd., MS State, MS 39762, eeg87@msstate.edu), Steven L. Bunkley (USACE ERDC, Vicksburg, MS), John Ball (Elec. and Comput. Eng., Mississippi State Univ., MS State, MS), and Anton Netchaev (USACE ERDC, Vicksburg, MS)

Current methods for direction of arrival (DOA) estimation are disproportionately represented in the literature by microphone array geometry and sound source properties. A wide array of implemented methods and publications are available for uniformly spaced 2D arrays such as uniform linear arrays (ULA), uniform circular arrays (UCA), and uniform rectangular arrays (URA). Further, implemented DOA estimators are specifically designed for narrowband, continuous signals. Methods applicable to wideband signals on arbitrarily-shaped arrays are limited; alternative approaches that partition the array into sub arrays expand the number of applicable methods. For a realistic military application of a single impulse localization on a conformal microphone array, methods must be able to estimate the DOA of wideband, static, acoustic sources. DOA estimator methods' performances, capabilities, and limitations are explored on various real-world sound sources and configurations of a five-microphone conformal array.

8:15

5aSP2. Direction of arrival estimation for broadband array based on conjugate gradient. Hai X. Sun, Dandan Hai, Yongchun Miao, and Huiming Guo (School of Information Sci. and Eng. .Xiamen Univ., Xiamen, FuJian 361005, China, hxsun@xmu.edu.cn)

In Broadband array, the direction of arrival (DOA) has high computing capacity, high complexity and real time difference. To solve this issue, we have proposed fast subspace estimation based on conjugate gradient (CG) method, which used to find orthogonal vectors and span the signal subspace. Initially, training signals are not required and also avoids eigenvalue decomposition. In this paper, subspace method based on CG is combined with DOA estimation of wideband signal coherent subspace, which replaces the focus matrix and the eigenvalue decomposition of narrowband DOA estimation. Herein, we equated the computational complexity is highly reduced and also better for real-time implementation of the DOA estimation. As performance compared with old methods experimental outcomes show our method has high efficiency and low complexity. Key words: direction of arrival, conjugate gradient, subspace estimation, focus matrix, eigenvalue, etc.

8:30

5aSP3. Off-the-grid direction-of-arrival estimation with multiple measurement vectors. Yongsung Park (Seoul National Univ., Gwanak-gu, Gwanak-ro 1, Seoul National University Bldg. 36 - Rm. 212, Seoul 08826, South Korea, ysparkwin@snu.ac.kr), Youngmin Choo (Sejong Univ., Seoul, South Korea), and Woojae Seong (Seoul National Univ., Seoul, South Korea)

Acoustic signals with different direction-of-arrivals (DOAs) arrives at an array of a limited number of sensors. Compressive beamforming technique [Xenaki *et al.*, *J. Acoust. Soc. Am.* **136**(1), 260–271 (2014)] provides high-resolution results, but the conventional compressive sensing methods cause estimation degradation when basis mismatch occurs. To overcome the basis mismatch, the off the grid DOA estimation technique [Xenaki and Gerstoft, *J. Acoust. Soc. Am.* **137**(4), 1923–1935 (2015)] is suggested, but the formulation of the scheme is limited to the case of a single snapshot. In the scenario where the DOAs of the sources are stationary across multiple snapshots, the signal processing technique, which can treat the multiple snapshot data jointly, provides a more stable estimate. We extend the single snapshot grid-free DOA estimation scheme to handle the case of multiple snapshots. The multiple snapshot off the grid DOA estimation technique is demonstrated with the experimental data and this technique shows robust performance for noisy environment.

8:45

5aSP4. Eigenanalysis-based adaptive interference suppression for underwater target estimation. Suiling Ren, Lianghao Guo, and Yanli Chen (State Key Lab. of Acoust., Inst. of Acoust., Chinese Acad. of Sci., No. 21 North 4th Ring, Beijing 100190, China, rsl@mail.ioa.ac.cn)

It is generally difficult for a passive sonar system to localize a weak source due to the complexity and variability of the ocean environment, especially in the presence of strong interferences. In this paper, an eigenanalysis-based adaptive interference suppression method (EAAIS) is presented for estimating the weak source bearing. Assuming the target of interest (TOI) is in a known bearing sector, we first define a contribution ratio for each eigenvector of the cross-spectral density matrix (CSDM). The eigenvectors not dominated by the TOI are adaptively identified and removed for interference suppression. The remaining eigenvectors are then used to reconstruct the CSDM for TOI bearing estimation with adaptive beamforming methods. In comparison with the basic ECA method, the proposed method has better interference rejection capability and wider fields of application. Simulation and experimental results also show that the proposed method can achieve accurate localization estimates even in the presence of strong interferences.

9:00

5aSP5. Microphone arrays for efficient communications during long-duration space missions. Peter Achi and Andi Petculescu (Dept. of Physics, Univ. of Louisiana at Lafayette, Lafayette, LA 70504, pya0168@louisiana.edu)

The astronauts' ability to communicate easily among themselves or with the ship's computer should be a high priority for the success of the mission. Long-duration space habitats—whether spaceships or surface bases—will likely be larger than present-day Earth-to-orbit/Moon transfer ships. Hence an efficient approach would be to free the crew members from the relative burden of having to wear headsets throughout the spacecraft. This can be achieved by placing microphone arrays in all crew-accessible parts of the habitat. Processing algorithms would first localize the speaker and then perform speech enhancement. The background ''noise'' in a spacecraft is typically fan and duct noise (hum, drone), valve opening/closing (click, hiss), pumps, etc. We simulate such interfering sources by a number of loudspeakers broadcasting various sounds: real ISS sounds, a continuous radio stream, a poem read by one author, etc. To test the concept, we use a linear

30-microphone array driven by a zero-latency professional audio interface. Speaker localization is obtained by time-domain processing. To enhance the speech-to-noise ratio, a frequency-domain minimum-variance approach is used. Time-permitting, we will discuss array weight sensitivity to parameters such as frame length/overlap, windowing, (sub-)array structure, etc. [This work was supported by the Louisiana Space Consortium (LaSPACE).]

9:15

5aSP6. An improved robust adaptive beamforming based on worst-case performance optimization. Chao Yan, Weiyu Zhang, Peng Xu, and Mingyang Zhou (State Key Lab. of Acoust., Inst. of Acoust., Chinese Acad. of Sci., No. 21 North 4th Ring Rd., Beijing 100190, China, yanchao@mail.ioa.ac.cn)

Generally, the adaptive beamformer has better spatial resolution and much better interference rejection capability than the conventional data-independent beamformer. But, in practice, the performance of the traditional adaptive beamformer will degrade greatly if some assumptions or information on the propagation model, array parameters, and signal model are imprecise. In this paper, a new approach to robust adaptive beamforming based on optimization of worst-case performance by parameter auto-adjust (PAWCO) is proposed, which can estimate the power of the signal using Newton-Raphson interactions. The method can confirm the upper bound of the mismatch vector neither overly nor under estimated. Theoretical analysis and simulation results are presented to show the proposed beamformer can estimate the power of signal precisely while achieving better spatial resolution. The robustness of the PAWCO algorithm to environment mismatch is also demonstrated by the acoustical experimental data.

9:30

5aSP7. An inexpensive acoustic data acquisition system using a single-board microcomputer. Andrew B. Wright (Systems Eng., Univ. of Arkansas at Little Rock, 2801 South University Ave., EIT 522, Little Rock, AR 72204, abwright@ualr.edu)

Low-cost, single-board microcomputers have become increasingly available, driven primarily by the smart phone and tablet markets. The beaglebone black (BBK, circa $50) contains a host processor, which is normally configured to run the linux operating system, and two programmable real-time units (PRU) that can access all peripherals on the device. The host processor is used to coordinate all of the human interface tasks, such as ethernet, usb, i2c, and hdmi. The PRU microcontrollers can sample either the 12-bit, 8 channel internal analog to digital converters (ADC), an external ADC, or a USB sound card, at high, precisely controlled sample rate. In this project, a BBK was programmed to sample an inexpensive external microphone through the internal ADC at 40 kHz. The power supply, external electronics, and BBK were housed in a custom-designed, 3D printed case. A unidirectional shroud for the microphone was designed and printed.

9:45–10:00 Break

10:00

5aSP8. Building air-infiltration quantification based on sound transmission loss calculated using nearfield acoustic holography. Kanthasamy Chelliah and Ralph T. Muehleisen (Argonne National Lab., 9700 S Cass Ave., Bldg. 362, Lemont, IL 60439, kchelliah@anl.gov)

This talk will demonstrate the abilities of nearfield acoustic holography (NAH) to detect and quantify leakages in building envelopes. A tonal sound source was placed inside a building model which has known leakages and microphone array measurements were obtained from the outside. Equivalent sources model based NAH was applied on the measured data to reconstruct the sound pressure field on the wall of the building model. Present results show that the NAH method was able to successfully locate the major leakages. A single microphone was used to measure the sound pressure level inside the building model which was used as a reference for quantification calculations. The difference between the inner and outer sound pressure levels was related to the area of the leakage. Various sizes of pinholes and rectangular cracks were investigated, and the detection limits of the current method were explored.

10:15

5aSP9. Representation of the combined concurrent acoustic sources in the electrical domain. Azhar Abdulaziz and Veton Këpuska (Elec. and Comput. Eng., Florida Inst. of Technol., 220 E University Blvd. Apt#1504, Melbourne, FL 32901, aabdulaziz2013@my.fit.edu)

In the electrical domain, adding samples of individual sound sources is used to represent a combined acoustic signal formed by them. It is commonly believed that, when many concurrent acoustic sources are sampled by a single microphone, it will produce samples that are equal to the linear addition of the electrical samples of those sources. The key problem about this argument is it assumes linearity between the total sound pressure level (SPL) and each individual SPL. It also assumes that the microphone voltage varies linearly with the SPL applied on it. However, it is more likely that both of those relationships are non linear, at least in theory. We believe that this is the first study that correlates the mathematical relationship between the individual acoustic signals and their combination in both electrical and acoustic domains. A theoretical new formula is derived throughout this work showing that the acoustic source samples are added in a non-linear manner. In the electrical domain, the voltages that represent each component of a multiple acoustic sources, are added together as if they are perpendicular vectors. Experimentally, adding electrical samples using the new addition formula is more accurate than the common linear one.

10:30

5aSP10. Influence of microphone position error on generating spatial null sensitivity point by line microphone array. Akio Ando (Electric and Electronics Eng., Faculty of Eng., Univ. of Toyama, 3190 Gofuku, Toyama 930-8555, Japan, andio@eng.u-toyama.ac.jp)

We developed a method that picked up a sound of a target source located beyond the noise source [1]. It used an end-fire line microphone array whose direction was toward the noise source and created a spatial null sensitivity point at the noise source position. This method relied on the assumption that the precise distance between microphones was known so that all microphone outputs from the noise source became in phase by using the delay calculated by the distance. In this study, we analyzed the influence of the error of microphone distance on the performance of the method. As a result, if the real distance was larger than that of the system's knowledge, the dip of sensitivity at the noise position was separated along the direction normal to that of the line of the microphones. On the other hand, it the distance was smaller than that of the system's knowledge, the dip did not separate and just became shallower. Based on these results, we will propose how to compensate the microphone position error. [1] A. Ando, *et al.*, 3pSP3, Acoustics '17 Boston, 2017.

10:45

5aSP11. Analysis of a prediction error method employing orthonormal basis functions in adaptive feedback cancellation for hearing aids. Sahar Hashemgeloogerdi and Mark Bocko (Elec. and Comput. Eng., Univ. of Rochester, 72G ClintwooDC Apts, Rochester, NY 14620, shashemg@ece.rochester.edu)

Hearing aids often suffer from acoustic feedback, which limits the achievable amplification and may severely degrade sound quality by creating howling artifacts. A potential method of feedback cancellation comprises predicting the feedback path (FP) using an adaptive filter. However, a large model error, or bias, is introduced due to signal correlations. To reduce the bias in the FP estimate, a prediction-error-method (PEM) has been used, which is based on a closed-loop identification of the FP and the auto-regressive modeling of the desired signal. This approach, which represents the FP using an FIR filter, requires a large number of parameters. Furthermore, in a reverberant environment, even a high order of the FIR filter may be insufficient to fully represent the FP, which reduces the convergence rate and limits the maximum stable gain of the system. In this contribution, we introduce a PEM-based method that utilizes orthonormal basis functions to precisely predict the acoustic FP in hearing aids. Simulation results with measured data show that the proposed method outperforms the standard PEM adaptation algorithm in terms of convergence rate and maximum stable gain.

11:00

5aSP12. Applications of operator theory and symmetry arguments to sound field recording and reproduction. Jorge A. Trevino Lopez, Shuichi Sakamoto, and Yôiti Suzuki (Res. Inst. of Elec. Commun., Tohoku Univ., 2-1-1 Katahira, Aoba-ku, Sendai, Miyagi 9808577, Japan, jorge@ais.riec.tohoku.ac.jp)

The problem of recording and reproducing sound fields using microphone/loudspeaker arrays is treated by exploiting its symmetries. This approach leads to well-known results, such as high-order Ambisonics corresponding to the requirement of the error field to be invariant under rotations. More importantly, it allows us to treat some difficult problems critical to the application of sound field recording and reproduction to virtual reality, where one or more users can freely move around a realistic environment. In this presentation, we consider three of these problems. First, we present a method to extrapolate extended sound field information from conventional spherical microphone array recordings by imposing symmetry constrains based on a priori knowledge of the sound source positions. Second, we apply the same symmetry constrains to the problem of sound field reproduction, allowing us to formulate a method of presenting spatial sound to moving listeners. Finally, we consider the problem of describing and reproducing the sound field due to a moving sound source.

11:15

5aSP13. Synthesis of non-uniformly spaced circular antenna arrays using a data-driven probabilistic model. Nicholas Misiunas (ECE, Univ. of Massachusetts, Lowell, 125 Heath St., Tewksbury, MA 01854, nmisiuna@purdue.edu), Lejun Hu, Kavitha Chandra, and Charles Thompson (ECE, Univ. of Massachusetts, Lowell, Lowell, MA)

Beamforming from non-uniformly spaced elements on a circular array of fixed radius that is steered in a two-dimensional plane is investigated. Extending previous results for randomly spaced elements on a linear array, this work examines the design of a probabilistic model using element position vectors derived through a meta-heuristic optimization process. The circular array presents a benefit in that an optimal positional distribution derived for one scan angle may be applied to other scan angles through a rotational operator. The metaheuristic optimization based on the firefly algorithm is utilized to generate a dataset of viable element position vectors where the array is scanned to a specified angle. Analysis of the positional vectors shows approximately symmetrical patterns for element positions in each quadrant, with respect to the scanned angle. A robust model utilizing both the joint probability density function (pdf) of the number of elements in each quadrant along with a joint pdf of the first element's location and the total angle subtended per quadrant for activating elements along the circular array is derived and its performance demonstrated.

11:30

5aSP14. Efficient algorithm for active noise control of impulsive noise. Allina Mirza and Ayesha Zeb (NUST, Mechatronic Dept., Rawalpindi, Pakistan, ayeshazeb79@ee.ceme.edu.pk)

Active Noise Control (ANC) systems employing adaptive filters suffer from stability issues in the presence of impulsive noise. Due to its simplicity and less computational complexity, the Filtered-x Least Mean Square (FxLMS) algorithm is the most widely used ANC algorithm which minimizes the mean square error of noise, but it lacks robustness and stability in presence of high impulses. To overcome this limitation, new methods must be investigated. A robust adaptive algorithm (Bhagyashri algorithm) for impulsive noise suppression is already proposed in literature. In this paper, Bhagyashri algorithm is tested in ANC domain and thus Filtered-x Bhagyashri (FxBhag) algorithm is proposed. Computer simulations are executed to verify the enhanced performance of the FxBhag algorithm. The statistics of impulsive noise is modeled by Symmetric α-stable (SαS) distributions. The suggested solution exhibits better stability and faster convergence with almost same computational complexity as that of standard FxLMS algorithm and its variants.

FRIDAY MORNING, 8 DECEMBER 2017 SALON E, 8:30 A.M. TO 11:00 A.M.

Session 5aUW

Underwater Acoustics: Underwater Measurements and Applications

Jason D. Sagers, Cochair
Environmental Science Laboratory, Applied Research Laboratories, The University of Texas at Austin, 10000 Burnet Road, Austin, TX 78758

Michael J. Smith, Cochair
Center for Coastal and Ocean Mapping, University of New Hampshire, 664 Central Ave., A, Dover, NH 03820

Contributed Papers

8:30

5aUW1. Motion planning based on matched field processing with Bayesian filtering. Caitlin Bogdan, Kenric Nelson, Sean Andersson, and James G. McDaniel (Boston Univ., 110 Cummington Mall, Boston, MA 02215, cbogdan@bu.edu)

Matched Field Processing (MFP) has often been used with stationary arrays as a method for developing source location information, but potentially with large amounts of uncertainty. This has been overcome in some cases by using Bayesian Filtering to improve estimates over time of the source location by treating this quantity as a random variable. Here we show how these methods can be applied to an autonomous vehicle for motion planning. The vehicle moving through an acoustic field can provide spatially separated data that can be used in an MFP algorithm to generate an estimated source location, which can be integrated utilizing Bayesian Filtering. Motion planning methods are shown which aim to move the vehicle to the maxima of this probability distribution. This method is demonstrated using ocean acoustic simulations. [Work supported by Raytheon Advanced Studies Fellowship.]

8:45

5aUW2. Analysis of the radiated sound field of deep water multibeam echo sounders for return intensity calibration using an underwater hydrophone array. Michael J. Smith (Ctr. for Coastal and Ocean Mapping, Univ. of New Hampshire, 664 Central Ave., A, Dover, NH 03820, msmith@ccom.unh.edu), Thomas C. Weber, Larry Mayer (Ctr. for Coastal and Ocean Mapping, Univ. of New Hampshire, Durham, NH), David Moretti (NUWC, Wakefield, RI), Anthony P. Lyons, and Val E. Schmidt (Ctr. for Coastal and Ocean Mapping, Univ. of New Hampshire, Durham, NH)

Multibeam echo sounders (MBES) are tools used to gather geophysical information on the seafloor and watercolumn which are important for feature detection, identifying gas seeps, and characterizing the seafloor, among others. At high frequencies (>100 kHz), MBES can be calibrated for their ensonification patterns in test tanks. However, deep water MBES feature long transmit arrays and varying geometries that make tank calibration impractical. The transmit arrays can be over 8m and have a far field range in the hundreds of meters. In addition, these systems use beam steering techniques to segment the swath into multiple sectors to mitigate ship motions, which complicates the radiated pattern and return intensity. This study will better characterize the radiated sound field of deep water MBES for return intensity calibration. A MBES survey was conducted using a Kongsberg EM122 MBES on the SCORE range, a submerged broadband hydrophone array. Hydrophones were spaced ~5 km apart and were continuously recording during the survey. The EM112 is a multisector dual swath system operated at 12 kHz with CW waves. Hydrophone data were analyzed, and the resultant radiated sound field was determined at different distances and angles.

9:00

5aUW3. Spatial resolution of time-reversal processing with a virtual source array. Donghyeon Kim, Gihoon Byun (Dept. of Convergence Study on the Ocean Sci. and Technol., KIOST-KMOU Ocean Sci. and Technol. School, 727 Taejong-ro, Yeongdo-Gu, 253, Ocean Sci. and Technol., Korea Maritime and Ocean Univ., Busan Asikrks012Busan, South Korea, donghyeonkim@kmou.ac.kr), and Jea Soo Kim (Dept. of Ocean Eng., Korea Maritime and Ocean Univ., Busan, South Korea)

Time-reversal processing (TRP) can be implemented spatially and temporally to refocus incident field back to its origin. The main limitation to TRP is that it requires a probe source. This limitation was partially relaxed by the concept of a virtual source array (VSA) [S. C. Walker *et al.*, *J. Acoust. Soc. Am.* **125**, 3828–3834 (2009)] proposed for focusing time-reversed field back to the selected location without a probe source. In the case of spatial resolution of TRP, it is mathematically well documented [S. Kim *et al.*, *J. Acoust. Soc. Am.* **110**, 820–829 (2001)]. In this study, we investigate variable factors affecting spatial resolution of TRP based on a VSA. Numerical simulation results are presented and discussed.

9:15

5aUW4. Passive underwater acoustic markers for navigation and information encoding for high frequency sound navigation and ranging (SONAR) devices. Aprameya Satish, Brendan Nichols, David Trivett, and Karim G. Sabra (Mech. Eng., Georgia Inst. of Technol., Woodruff School of Mech. Eng., 801 Ferst Dr., Atlanta, GA 30313, aprameya.satish@gatech.edu)

AUV navigation requires accurate positioning information from the surrounding environment. Currently, several underwater navigation and surveying paradigms employ active transponders that assist in triangulation. These systems are expensive, require maintenance, and additional power sources. This paper presents a novel system that may be implemented to guide AUVs equipped with high frequency SONAR, using passive acoustic tags that are cost effective, and simple to deploy. The acoustic tags are constructed by layering multiple sheets of different acoustically reflective materials. When a deployed tag is ensonified by an encoded signal sent from a collocated source/receiver pair, the backscattered waveform by the tag yields a time-domain signature unique to the properties of the materials and geometry of the tag. The signature can be used to locate and identify any given tag within a known library of various tag designs. Numerical simulations and experimental results will demonstrate the feasibility of the proposed passive acoustic tag approach. The developed acoustic tags may find application in AUV navigation and docking exercises, swarm AUV vessel identification, AUV route planning, tagging undersea ecosystem boundaries, etc.

9:30–9:45 Break

9:45

5aUW5. Method for underwater impulsive sound characteristic measurement in non-anechoic pool. Junming Zhang, Rui Tang, Zirong Gao, and Xinyue Yu (Harbin Eng. Univ., Nantong Str. Nangang Dist. No.145, Harbin 150001, China, tangrui@hrbeu.edu.cn)

A method for measuring the sound power of impulsive sound sources in non-anechoic pool was proposed in this paper, which can also be applied to measure the sound power of other transient sound sources. The characteristic of sound field in enclosed space were presented based on normal-wave theory. By calculating the volume integral of square pressure within local sound field, the interference caused by boundary was eliminated and the steady sound power density of the enclosed space was obtained. The sound power of the source can then be obtained by introducing the sound field transmission relationship from enclosed space to free field. The method was extended to measure the sound power of transient sound sources in non-anechoic pool. The sound power of impulse sound sources was measured by using the method proposed in this paper. The effectiveness of the proposed method was proved by comparing the sound power measured in anechoic pool and non-anechoic pool.

10:00

5aUW6. Rapid measurement method of hydrophone in reverberation pool. Jundong Sun (Harbin Eng. Univ., Harbin, Heilongjaing Nantong St., China, sunjundong@hrbeu.edu.cn)

In this paper, a new method based on spatial averaging is proposed to measure hydrophone rapidly in reverberation pool. The method is to overcome the influence of acoustic normal mode by spatial averaging technique, get the reverberation sound RMS pressure in reverberation control area. It is according to the comparison principle to calibration of hydrophone rapidly. The tank is $15\text{m}\times9\text{m}\times6\text{m}$, in the frequency range of 400–10 kHz. The free field voltage sensitivity of the four hydrophones is measured. The measurement results are compared with the free field comparison method of measurement results, the deviation is less than 1 dB, to verify the effectiveness of the method. The method can realize the simultaneous measurement of multiple hydrophones, improve the calibration efficiency, and the lower limit of measurement frequency is much lower than the lower limit frequency of the conventional pulse comparison technique, which improves the measurement capability of the anechoic tank.

10:15

5aUW7. Methods of unwrapping phase ambiguity and selecting direct sounds in an ultra short baseline positioning system. Xuyan Liu, Nan Zou, and Yifeng Zhang (College of Underwater Acoust. Eng., Harbin Eng. Univ., 145 Nantong St., Nangang District, Harbin, Heilongjiang 150000, China, liuxuyan918@126.com)

This paper presents a novel method to unwrap phase ambiguity and an expert system to select the direct sound from several reflecting pulses in an USBL positioning system. Generally, in the crossed-shaped USBL array, unwrapping ambiguity in the long edge is based on the absence of ambiguity in the short edge, where the array spacing is less than half-wavelength. However, if the positioning system works at a high frequency, which results in the short-edge phase ambiguity, the classical method to unwrap long-edge ambiguity will fail. This paper proposes a sector searching method for this scenario. It utilizes all combinations of two edges to solve all possible target orientations, and takes the mode as the final result. Additionally, hydrophones receive multi-path reflections besides the direct sound in practical application. The expert system is particularly used for selecting the direct sound correctly, which seriously effects the positioning precision. It contains series of criteria to evaluate the quality of each pulse and considers the pulse with the best quality as the direct sound. Simulation and field

experiment results prove that the proposed methods are feasible and can achieve good accuracy. Keywords: USBL; phase ambiguity; expert system; and direct sound.

10:30

5aUW8. Underwater non-cooperative communication signal recognition with deep learning. Cheng LI (Harbin Eng. Univ., No. 145 in NanTong St., Harbin, HeiLongjiang 150001, China, lichengong1@163.com), Qiming Zhou, Xiao Han, Jingwei Yin, and Mengqi Shao (Harbin Eng. Univ., Harbin, Heilongjiang Province, China)

The channel of underwater acoustics is time-varying and space-varying, which leads to the severe interference in the underwater acoustics communication signals. That makes a big challenge to distinguish the underwater acoustics communication signals modulation types. The traditional methods of identifying the non-cooperative signal modulation rely on statistics. They regard the statistical parameters in time-domain or frequency-domain of the signals as the features. In order to avoid extracting the features artificially we utilize the deep learning method to distinguish the raw time-domain signals into different modulation types such as Binary Phase Shift Keying (BPSK), Quadrature Phase Shift Keying (QPSK), 8 Phase Shift Keying (8PSK), Direct Spread Spectrum Sequence (DSSS) and Orthogonal Frequency Division Multiplexing (OFDM). The result of recognition for the experimental data shows the validity of our method. Finally we achieve 100% accuracy rate to BPSK, QPSK, DSSS modulation types and 90% accuracy rate to 8PSK, OFDM modulation types. Comparing with the statistics method, the accuracy of our method is higher.

10:45

5aUW9. An underwater measurement and control network centralized data fusion localization algorithm based on Chan-algorithm method. Tianbai Zhao, Jin Fu, Yan Wang, and Boxuan Zhang (Underwater Acoust. Eng. College, Harbin Eng. University, Rm. 914, No. 145, Nantong St., Harbin 150001, China, 627161473@qq.com)

In order to exploit the development potential of current measurement and control equipment and to build an omnibus underwater measurement and control network with higher precision, according to the working characteristics of the system itself, an underwater measurement and control network centralized data fusion localization algorithm based on Chan-algorithm method is proposed. It is an algorithm that coarsely calculates the target position according to the time of arrival by using the weighted least squares estimation(WLS) method at first; second, constructs new error vectors on the basis of the relationship between the position and the time delay information; last, resolves the target position from the vectors above by using WLS method again. The result of research demonstrates that the algorithm realizes the data fusion method of multiple sets of underwater measurement and control equipment, which could improve the precision of localization globally. The precision of the proposed method is much better than the data fusion localization algorithm purely based on the time of arrival.