Session 1aAA

Architectural Acoustics and Noise: Impact of Entertainment Sound on Communities

David Woolworth, Cochair
Roland, Woolworth & Associates, 365 CR 102, Oxford, MS

K. Anthony Hoover, Cochair
McKay Conant Hoover, 5655 Lindero Canyon Road, Suite 325, Westlake Village, CA 91362

Chair's Introduction—8:15

Invited Papers

8:20

1aAA1. Challenges of entertainment sound in communities. Gary W. Siebein, Gary Siebein, Hyun Paek, Marylin Roa, Jennifer R. Miller, and Keely Siebein (Siebein Assoc., Inc., 625 NW 60th St., Ste. C, Gainesville, FL 32607, gsiebein@siebeinacoustic.com)

There are three basic challenges of entertainment sound in communities. First is how to measure the sound. People often clearly hear individual notes of music being played, individual beats from drums, and other lower frequency percussive instruments and sounds of words sung by performers at locations away from the facility. Equivalent sound levels taken over varying periods of time, LA Fast, LA slow, and other experimental metrics will be compared with the audibility of the sounds by people at remote listening locations. Second, it is often necessary to correct the measurements for the effects of background noise when equivalent sound levels are measured because the short term transient sounds that contain the musical information have been "averaged away" and the resulting sound levels are in the vicinity of the background sounds even though the words and music are plainly audible. Third is that methods to effectively contain the sounds from outdoor amplified entertainment facilities including operational controls, infrastructure controls, and administrative controls should be considered. All three challenges must be addressed to optimize the compatibility of the facility with the community.

8:40

1aAA2. Noise impact modeling in two residential areas from amplified music. David Manley, Benjamin Bridgewater, Ted Pitney, Ian Patrick, and Edward Logsdon (D. L. Adams Assoc., Inc, 1536 Ogden St., Denver, CO 80218, dmanley@dlaa.com)

D.L. Adams Associates has recently worked on two outdoor music venue projects, both in close proximity to residences in which venue owners were concerned about noise impact to the neighboring community. The first project involved the renovation of a city owned garden with the addition of a large stage and audience area. Residences are located within 450 feet of the proposed stage. Modeling was completed to evaluate noise impact, and mitigation recommendations were provided. The second project involved a privately owned sports facility which was leased to provide an Electronic Dance Music (EDM) festival over the course of a weekend. After reading about complaints of many such EDM festivals, the facility Owner wanted to understand the potential noise impact to the neighbors prior to the event. Environmental noise modeling of the facility and anticipated musical acts was completed to predict sound levels in the surrounding neighborhoods and demonstrate to the facility Owner the potential for complaints from the festival.

9:00

1aAA3. Interpreting conflicting noise ordinances imposed upon Merriweather Post Pavilion. Josh Curley and Scott Harvey (Phoenix Noise & Vib., 5216 Chairmans Court, Ste. 107, Frederick, MD 21703, sharvey@phoenixnv.com)

Merriweather Post Pavilion in Columbia, Maryland, has been host to numerous concerts and music festivals each year since it opened in 1967. Over the years, noise complaints regarding Merriweather have not been uncommon during musical acts. In 2013, a noise ordinance was placed on the venue that allows concert events to generate a noise level up to 95 dBA at a $^1/_4$-mile radius from the main stage between the hours of 9:00 AM and 11:00 PM, and 72.5 dBA at residential property lines. Between 11:00 PM and 11:30 PM, main stage noise levels must not exceed 55 dBA as measured both within the $^1/_4$-mile radius and at residential property lines. Currently, the residence closest to the main stage is approximately 1,400 feet away, just outside of the $^1/_4$-mile 95 dBA allowance; however, residential construction is planned in the next few years as close as 750 feet behind the main stage. Once these residential buildings are occupied, how will the noise ordinance be applied if main stage noise is greater than the 72.5 dBA allowance at residential buildings yet below the 95 dBA limit within the $^1/_4$-mile radius? And, how will the likely complaints regarding audible low frequency noise inside the residences, which are not addressed by the noise ordinance, be handled?

9:20

1aAA4. Consistency and accountability: First steps in a quest for compromise between venue and community. Eric L. Reuter (Reuter Assoc., LLC, 10 Vaughan Mall, Ste. 201A, Portsmouth, NH 03801, ereuter@reuterassociates.com)

This case study examines the first steps in reducing several years of tension between an active outdoor venue, with more than 60 performances each summer, and the surrounding community in a small coastal city in New Hampshire. Continuous sound level monitoring and realtime visual feedback provide means of achieving consistent (though somewhat arbitrary) sound levels throughout and across performances, thus eliminating a variable in the evaluation of impacts on the community. Sound level data, along with feedback from the community, will be used to develop mitigation strategies for future seasons. Details of the monitoring and feedback systems and lessons learned from the 2017 season will be shared.

Contributed Paper

9:40

1aAA5. Noise assessment for residential developments near places of entertainment in San Francisco. Jordan L. Roberts (none, Charles M. Salter Assoc., 130 Sutter St., Ste. 500, San Francisco, CA 94104, jordan.roberts@cmsalter.com)

When new multi-family residential developments are proposed in San Francisco, the City's Entertainment Commission identifies whether there are places of entertainment (POE) within 300 feet. A POE includes businesses such as music venues, clubs, and bars. The Commission's goal is to reduce complaints from new nearby residents about noise generated by existing entertainment venues. If there are too many complaints, a POE might be forced to close its business, which would have a negative impact on the City's various entertainment districts. This assessment addresses noise that is distinct from the noise associated with a typical environmental noise study (which addresses noise from street traffic, rail activity, etc.). Rather than put the onus on existing businesses, the housing developer is asked to consider bass-heavy nighttime noise sources that future residents could perceive. Thus, some facades of proposed buildings might need to provide greater noise reduction than typical construction, often times through increased window STC rating recommendations. The measurements, calculations, and mitigations of these assessments are to be presented through select data from multiple case studies.

Invited Papers

9:55

1aAA6. Typical outdoor audio system designs. Andrew N. Miller (BAi, LLC, 4006 Speedway, Austin, TX 78758, amiller@baiaustin.com)

Trends in audio system designs for outdoor entertainment venues impact the sound in surrounding communities. The author explores typical audio system designs for outdoor amplified music venues and high school football stadiums in and around Austin, Texas. The consequences of particular system designs will be discussed as they relate to sound in surrounding communities. Given special consideration are characteristics of sound at the property line of venues employing loudspeaker array and point-source systems.

10:15–10:30 Break

10:30

1aAA7. Impact and control practices of bar and pub sound in densely populated cities. Andy Chung (Smart City Maker, Smart City Maker, Copenhagen, Denmark, ac@smartcitymaker.com), W. M. To (Macao Polytechnic Inst., Macao, Macao), K. K. Iu (Supreme Acoust. Res. Ltd., Hong Kong, China), and Maurice Yeung (Macau Instituto de Acústica, Macau, China)

While patrons are enjoying the happy moments in bars and pubs, the sound generated from these premises may not be wanted by the neighboring community, in particular, during the sensitive hours. The situation becomes more challenging in a densely populated city, where various constraints exist in attempting to alleviate the noise impacts from propagation paths and the receiving ends. Controlling at source is considered to be the most effective approach. This paper presents the prevailing practices in Hong Kong, in China's Bay Area, to control the noise impacts associated with the bars and pubs, and illustrates with examples.

10:50

1aAA8. Benefits of managing source sound levels in amplified indoor entertainment venues. David S. Woolworth (Roland, Woolworth & Assoc., 356 CR 102, Oxford, MS 38655, dave@oxfordacoustics.com)

This paper investigates sound levels on stage and in the audience from field measurements and the literature for small and midsize amplified venues for various entertainment types. The effect of sound levels are also briefly examined in regard to hearing and comfort of patrons and employees. Furthermore, reduction in sound level internally is examined as a community noise control measure relative to soundproofing.

11:10

1aAA9. Entertainment sounds, the soundscape, and resources. Brigitte Schulte-Fortkamp (Inst. of Fluid Mech. and Eng. Acoust., TU Berlin, Einsteinufer 25, Berlin 101789, Germany, b.schulte-fortkamp@tu-berlin.de)

Assessing the impact of entertainment sounds in communities will need a multifaceted analysis. An evaluation process has to be considered that will cover the different facets with regard to people's understanding about an acoustic environment where the impact of entertainment sound will occur. The "meeting of music and environment" discussed by R. M. Schafer opens the mind in many directions to learn about the different aspects and related impacts in such a soundscape that is relying on given activities in an area that will lead to a harmonized or an acoustically unbalanced situation. Understanding human auditory scene analysis and the important role of auditory attention forces to outline soundscape assessment methods and to come to enhanced methodologies for inventing action parameters with regard to a sounding community. This paper will introduce regulations to harmonize the different interests regarding interventions through entertainment sounds in communities but also introduce the soundscape approach to offer new features with regard to a balanced and harmonic soundscape addressing entertainment sounds as a resource.

11:30

1aAA10. Case study of an outdoor music festival in a rural community. Robert M. Lilkendey (RML Acoust., LLC, 14688 NW 150th Ln., Alachua, FL 32615, rob@rmlacoustics.com)

A case study will be presented of the author's involvement in assisting Okeechobee County, Florida, with the development of acoustical criteria and monitoring of live music events associated with a four-day outdoor music festival that takes place annually in a rural part of the county. The presentation will include the results of these efforts, as well as a discussion of the collaborative process that continues to take place between the promoter, promoter's acoustical consultant, the county planning and zoning department, residents, elected county officials, law enforcement, and the author to work toward a solution for future festivals that balances the monetary benefits for the entire county with the needs of the relatively small percentage of county residents adversely affected by the noise.

MONDAY MORNING, 4 DECEMBER 2017 SALON F/G/H, 8:00 A.M. TO 10:05 A.M.

Session 1aAO

Acoustical Oceanography and Animal Bioacoustics: Oceanographic Contributions to the Characteristics and Variability of the Underwater Soundscape

David R. Barclay, Cochair
Department of Oceanography, Dalhousie University, PO Box 15000, Halifax, NS B3H 4R2, Canada

Bruce Martin, Cochair
JASCO Applied Sciences, 32 Troop Avenue, Suite 202, Dartmouth, NS B3B 1Z1, Canada

Chair's Introduction—8:00

Invited Paper

8:05

1aAO1. Tidal influence on underwater soundscape characteristics in a shallow-water environment off Taiwan. Shane Guan (Office of Protected Resources, National Marine Fisheries Service, 1315 East-West Hwy., SSMC-3, Ste. 13700, Silver Spring, MD 20902, shane.guan@noaa.gov), Chi-Fang Chen, Chih-Hao Wu, Dai-Hua Liu (Dept. of Eng. Sci. and Ocean Eng., National Taiwan Univ., Taipei, Taiwan), and Joseph F. Vignola (Dept. of Mech. Eng., The Catholic Univ. of America, Washington, DC)

Passive acoustic monitoring has been conducted in shallow-water environment off the coast of west Taiwan using bottom-mounted hydrophones to obtain baseline soundscape information prior to large scale wind farm constructions. Analyses of long-term acoustic datasets were conducted in three distinctive one-octave bands: 150-300 Hz, 1,200-2,400 Hz, and 3,000-6,000 Hz. Initial study was able to link the 150-300 Hz band to passages of cargo vessels. However, further research using time-frequency analysis showed an approximate 6-hour cycle of strong occurrence of this noise band, which corresponds particularly well with the local semidiurnal tidal cycle. The results suggest that at least part of the high noise levels between the high- and low-water periods in the 150-300 Hz octave band was due to flow noise from tidal movements. The results underlines the importance to consider oceanographic processes such as tidal and wave movements when studying the contribution of specific acoustic sources to the underwater soundscape.

Contributed Papers

8:20

1aAO2. Long-term measurements of the ice-free underwater noise field directionality in the shallow Beaufort Sea. Aaron Thode (SIO, UCSD, 9500 Gilman Dr., MC 0238, La Jolla, CA 92093-0238, athode@ucsd.edu), Susanna B. Blackwell (Greeneridge Sci., Inc., Aptos, CA), Alexander Conrad, and Katherine H. Kim (Greeneridge Sci., Inc., Santa Barbara, CA)

Between 2007 and 2014, over 35 Directional Autonomous Seafloor Acoustic Recorders (DASARs) were deployed over a 280 km swath of the Beaufort Sea continental shelf (20-55 m depth) during the open-water season, in order to monitor the westward bowhead whale migration. DASARs have one omnidirectional pressure sensor and two orthogonal particle velocity sensors that permit instantaneous measurement of the azimuths of both transient signals and continuous noise between 20 and 500 Hz, including diffuse wind-driven ocean noise. The lack of significant shipping or industrial noise in this region provides a rare opportunity to directly measure the properties of wind-driven noise in expanding ice-free regions. Here, we map the azimuthal directionality of the diffuse Beaufort ambient noise field as a function of frequency and location across all seven seasons. The dominant directionality of the diffuse ambient noise field varies strongly with frequency and is highly correlated with the received power spectral density. Certain directional features of the ambient noise field remained stable over seven deployment seasons, suggesting that judicious processing of the ambient noise soundscape could provide underwater navigational information in arctic waters. [Work sponsored by ONR.]

8:35

1aAO3. An analysis of the soundscape of a tidewater glacial fjord environment, LeConte Glacier, Alaska. Matthew C. Zeh (Dept. Mech. Eng. and Appl. Res. Labs., Univ. of Texas at Austin, 1300 West 24th St., Apt. 212, Austin, TX 78705, mzeh@utexas.edu), Erin C. Pettit (Dept. of GeoSci., Univ. of Alaska Fairbanks, Fairbanks, AK), and Preston S. Wilson (Dept. Mech. Eng. and Appl. Res. Labs., Univ. of Texas at Austin, Austin, TX)

Glacierized fjords present a unique set of acoustic environments that are significantly louder than other ice-covered environments, with average sound pressure levels of 120 dB (re 1 μPa) with a broad peak between 1 and 3 kHz [*Geophys. Res. Lett.* **42**, 205 (2015)]. The intensity within this peak is due to the release of bubbles from compressed air-filled pores within melting glacier ice. The glacier-ocean boundary is dynamic, sensitive to fresh- and sea-water balances governed by submarine glacier melt from heat transfer between the glacier face and ocean [*J. Phys. Oceanogr.* **46** (2016)]. This heat transfer leads to regular calving events (< 200 Hz) contributing significantly to the overall sound level of the environment. Further, as icebergs melt and break apart they may remain in the fjord forming an ice mélange top boundary layer. This talk presents acoustic measurements from and analysis of the soundscape of LeConte Bay, a glacierized fjord in southeastern Alaska, studied from October 2015 to April 2016 using a 6-element vertical hydrophone array. The hydrophones recorded simultaneously at 48 kHz for 2 minutes every hour at depths ranging from 100 to 200 m approximately 1 km from LeConte Glacier. [Work supported by NDSEG Fellowship and ONR.]

8:50

1aAO4. Experimental studies of ice cracking tests in an anechoic chamber. Matthew V. Ahlrichs (Civil Eng., Univ. of Alaska, 3211 Providence Dr., Anchorage, AK 99508, matthew.ahlrichs@gmail.com), Chenhui Zhao, Marehalli Prasad (Mech. Eng., Steven's Inst. of Technol., Hoboken, NJ), James Matthews (Civil Eng., Univ. of Alaska, Anchorage, AK), Steven Opet (Elec. Eng., Stevens Inst. of Technol., Hoboken, Alaska), James Lyon (Chemical Eng., Stevens Inst. of Technol., Hoboken, NJ), Trevor Hinds (Mech. Eng., Steven's Inst. of Technol., Hoboken, NJ), and Khiana Rogers (Civil Eng., Northeastern, Boston, MA)

As the Northwest passage becomes more frequently traveled by commercial and business interests, it is becoming important to increase existing methods for safety at sea in harsh Arctic conditions. In order to identify ice-floe movements in the Arctic, acoustical signatures for ice behaviors need to be identified and analyzed. Being able to identify the acoustic signature of ice cracking in real-time will help vessels navigate the arctic by identifying moving ice-floes and monitoring present ice conditions. This study identifies a novel method for detecting thermal cracking in ice, ice fracturing, and ice shearing events in an anechoic chamber. Comparing the power spectra and identified peak frequencies against empirically collected data from previous studies, this methodology aims to recreate the arctic environment in a lab without travel to the region for baseline data collection. Furthermore, based on a literature review, little power spectra data exist. The data collected from this study will add to the creation of a baseline identification for future ice fracture studies. Observations are made on the collected data which can be used to improve navigational methods for safety at sea.

9:05

1aAO5. The sounds of submarine volcanoes. Gabrielle Tepp, Matthew Haney, John Lyons (USGS Alaska Volcano Observatory, 4230 University Dr. Ste. 100, USGS Alaska Volcano Observatory, Anchorage, AK 99508, gtepp@usgs.gov), Robert Dziak (NOAA/PMEL, Newport, OR), Joe Haxel (OSU/CIMRS, Newport, OR), Del Bohnenstiehl (Dept. of Marine, Earth & Atmospheric Sci., North Carolina State Univ., Raleigh, NC), and William Chadwick (OSU/CIMRS, Newport, OR)

When submarine volcanoes erupt, several processes can create sounds in the ocean, mostly at low frequencies <100 Hz. Explosions may occur directly in the water column, while earthquakes and other seismicity may produce seismic waves that convert into hydroacoustic waves or boundary (Scholte) waves. Volcanic sounds can propagate large distances through the SOFAR channel. During its 2014 eruption, Ahyi seamount, Northern Mariana Islands produced repetitive signals for approximately 2 weeks at a high rate. These likely explosions were widely recorded on seismometers throughout the region and on hydrophone arrays as far as Chile, ~12,000 km distant. Bogoslof volcano, a shallow submarine volcano in the Aleutian Islands, Alaska, began erupting in December 2016. Many of the detected earthquakes associated with this eruption have large amplitude hydroacoustic phases, likely Scholte waves. A few earthquake swarms were recorded as converted hydroacoustic waves by seismometers on Tanaga volcano, ~700 km away. A few eruption sequences were also detected on the Tanaga stations, one of which included a monochromatic glide, suggesting efficient transmission of energy into the water column. A single hydrophone deployed near Bogoslof several months after the eruption began may add evidence for how eruptions of submarine volcanoes contribute to the underwater soundscape.

9:20

1aAO6. Partitioning wind and ship generated sound using vertical noise coherence. Najeem Shajahan and David R. Barclay (Oceanogr., Dalhousie Univ., 1355 Oxford St., PO Box 15000, PO Box 15000, Halifax, NS B3H 4R2, Canada, nj210471@dal.ca)

Continuous ambient noise data were recorded during April and May 2016, using a four-element vertical array deployed near the continental shelf break south of Martha's Vineyard. A technique for classifying and partitioning ambient noise using the vertical noise coherence function is proposed. Time series analysis of the noise power spectrum reveals the presence wind, distant shipping, and near-field individual ship noise in the region. The noise coherence (directionality) due to wind, distant shipping, and individual ships is analytically modeled using environmental inputs such as the time varying sound speed profile and sediment properties from the measurement site, and compared with the observation. The impact of noise due to ship traffic in the region is estimated by subtracting the best-fit theoretical coherence for wind-generated noise from the measurement. Since the wind generated vertical coherence is stable and independent of source spectrum level, it can be used to quantify the relative contributions of distant shipping and wind noise to the marine environment. Additionally, the time varying vertical coherence from near-field individual ships can be used for estimating their range and speed. [Research supported by ONR.]

9:35

1aAO7. Progress toward acoustic volume attenuation tomography. Christopher M. Verlinden, Jeffery D. Tippmann, William A. Kuperman, and William S. Hodgkiss (Marine Physical Lab., Scripps Inst. of Oceanogr., Univ. of California, San Diego, 9500 Gilman Dr., La Jolla, CA 92093-0701, cmverlin@ucsd.edu)

Experiments were carried out in the eastern Pacific and north Atlantic Oceans with the purpose of estimating volume attenuation coefficients in seawater by measuring acoustic amplitude along eigenray paths in the mid-frequency regime (3-9 kHz). A quantitative comparison of the measured attenuation coefficients in three experiment sites is presented here along with a comprehensive examination of historical measurements in these areas. The ability to make precise measurements of acoustic volume attenuation coefficients has applications in detection theory, acoustic communications, and ocean sensing. A method of inverting for properties such as temperature, salinity, and pH, which influence the attenuation of sound in seawater, is discussed.

9:50

1aAO8. Three-dimensional noise modeling. David R. Barclay (Dept. of Oceanogr., Dalhousie Univ., PO Box 15000, Halifax, NS B3H 4R2, Canada, dbarclay@dal.ca) and Ying-Tsong Lin (AOPE, Woods Hole Oceanographic Inst., Woods Hole, MA)

The ambient sound field due to wind generated surface noise and distant ship generated noise may be highly axially non-symmetric in regions with rapidly changing bathymetry. Some of this asymmetry is due to bathymetric shadowing, while in other cases horizontal refraction contributes significantly. These propagation effects can alter the omnidirectional power spectral levels, as well as the horizontal and vertical noise coherence (directionality). The special case of an idealized Gaussian submarine canyon can be described using the method of normal mode decomposition applied to a three-dimensional longitudinally invariant wave-guide. The modal decomposition is carried out in the vertical and across-canyon horizontal directions and gives a semi-analytical solution describing the three-dimensional bathymetric effects on the noise field. Reciprocal three-dimensional cylindrical co-ordinates parabolic equation (PE) and Nx2D PE sound propagation models can be used to compute the noise field in arbitrary domains. Inter-comparison of these models highlights the effect of the three-dimensional topography on the vertical coherence and mean-noise level as a function of arrival direction relative to the canyon's axis. These effects include the focusing of noise along the canyon axis and the frequency perturbation of vertical coherence minima. [Research supported by ONR.]

MONDAY MORNING, 4 DECEMBER 2017 BALCONY N, 9:00 A.M. TO 10:00 A.M.

Session 1aEAa

Engineering Acoustics: General Topics in Engineering Acoustics I

Kenneth M. Walsh, Chair
K&M Engineering Ltd., 51 Bayberry Lane, Middletown, RI 02842

Contributed Papers

9:00

1aEAa1. Legendre polynomial method for linear array beam pattern synthesis. Dehua Huang (NAVSEANPT, Howell St., Newport, RI 02841, DHHuang@cox.net)

The detailed design tools for acoustic array sidelobe suppression and beamwidth control by Legendre polynomials technique for equally spaced point source linear acoustic array synthesis are studied and introduced. The method is compared with results obtained by conventional classical Dolph-Chebyshev synthesis technique. Numerical examples were demonstrated by a group of Legendre polynomials of different degrees of N with z_0 controlling parameters. Fourier series cosine expansion is applied to calculate element weight coefficient for great advantage when the number of sensor count in an array goes large. Naturally taped sidelobe levels within a beam pattern, and better beam efficiency, a measure of the energy percentage to the mainlobe, were both obtained by Legendre polynomial array synthesis method.

9:15

1aEAa2. Acoustic beam pattern for a fringe free laser interferometric sound sensor. Michael S. McBeth (SSC Atlantic, SSC Atlantic/NASA Langley Res. Ctr., 11 West Taylor St., M.S. 207, Hampton, VA 23681, m.s.mcbeth@ieee.org)

A fringe free laser interferometric sound sensor uses parallel, non-intersecting, modulated laser beams in a sound-filled fluid. Therefore, no interference fringes are produced or required in the fluid. Light from each laser beam is collected and combined in an envelope detector where the sound pressure signal appears as angle modulated sidebands. Since the laser beams do not intersect in the sound-filled fluid, we can use a beam separation distance that maximizes the optical path difference between the laser beams to increase sound detection sensitivity. We derive an equation that relates the optical path difference between a pair of laser beams and the peak change in the index of refraction of the fluid due a sound wave. We derive a three dimensional beam pattern and function for this sensor and compare it to that of a conventional line array.

9:30

1aEAa3. Acoustic field approximation with sparse arrays through optimization. Lane P. Miller, Stephen C. Thompson (Acoust., Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, lpm17@psu.edu), and Andrew Dittberner (GN Hearing, Glenview, IL)

In a previous work, the objective of approximating the acoustic field from an arbitrary continuous aperture with constrained sparse transducer arrays was established. Results from this showed limited accuracy due to the constraints and called for further consideration. The aim of this research is to extend the analysis and increase the accuracy of the field approximation through multi-variable optimization. System parameters such as transducer excitation level and location within the aperture space are varied. The acoustic coupling between transducers is also taken into account. Results from the optimizations will provide the system parameters that minimize error between compared radiation patterns.

9:45

1aEAa4. Guidelines and principles for noise and vibration performance in an autonomous vehicle. Arun Mahapatra (Vehicle Eng., Ford Motor Co., 20800 Oakwood Blvd., Dearborn, MI 48124, amahapa1@ford.com)

The field of autonomous vehicles (AVs) has grown substantially in recent years, both in public interest and engineering efforts. This presents a unique challenge in the world of automotive engineering—many processes and principles are based on the presence of an engaged driver or operator, while a passenger in an AV is assumed to have a much lower level of engagement. Additionally, AVs are an entirely new, quickly evolving market that has no historical data to use as a basis for engineering decisions, including those related to the noise, vibration, and harshness (NVH) experienced by a passenger. This paper reviews previous studies focusing on the NVH experiences of automobiles, other forms of transportation (trains, planes), and recent psychoacoustic findings to generate a set of principles and guidelines for NVH development of an AV. While there is minimal real-world information available on a consumer's experience with an AV, a remarkable amount of information from other industries and environments can be applied to ensure an optimal NVH environment for a passenger of a self-driving vehicle.

MONDAY MORNING, 4 DECEMBER 2017 BALCONY N, 10:30 A.M. TO 11:15 A.M.

Session 1aEAb

Engineering Acoustics: General Topics in Engineering Acoustics II

Kenneth M. Walsh, Chair
K&M Engineering Ltd., 51 Bayberry Lane, Middletown, RI 02842

Contributed Papers

10:30

1aEAb1. Full-field laser vibrometer study of dissipation in a beam with a coupled oscillator array. Jenna Gietl, John A. Sterling (The Catholic Univ. of America, 620 Michigan Ave., Washington, DC 20064, jenna.gietl@gmail.com), Teresa J. Ryan (East Carolina Univ., Greenville, NC), Joseph F. Vignola, and Diego Turo (The Catholic Univ. of America, Washington, DC)

Coupled oscillator systems can be designed to manage the distribution of mechanical energy in vibration or acoustic systems. A specific implementation of an array composed of small damped mass-spring attachments on a primary is here referred to as a subordinate oscillator array. SOAs and their ability to absorb energy from a primary mass are of continued importance in analysis for both structural and acoustic systems. The ability to dissipate energy in a specific frequency band has many potential applications. This work will use laser Doppler vibrometry (LDV) to evaluate a beam mounted on an electromechanical shaker with and without an attached planar SOA composed of much smaller beams. This study is motivated by earlier work that indicated a high sensitivity to disorder in the form of fabrication error. The apparatus allows for the controlled introduction of various levels of disorder to quantify its effect and compare to a numerical model. We will show full-field forced response from the LDV with and without the SOA and the resulting apparent damping within the band of the SOA for various levels of disorder.

10:45

1aEAb2. Toward acoustic particle velocity sensors in air using entrained balloons: Measurements and modeling. Randall Williams, Donghwan Kim (The Univ. of Texas at Austin, 10100 Burnet Rd., Bldg 160, Rm. 1.108, Austin, TX 78758, r.williams.06@gmail.com), Kristofer L. Gleason (Silicon Audio, Inc., Austin, TX), and Neal A. Hall (The Univ. of Texas at Austin, Austin, TX)

Acoustic particle velocity sensors for underwater applications typically work by measuring the motion of a small, neutrally buoyant body entrained in the acoustic field. Motivated by this, we have investigated the use of balloons for acoustic particle velocity measurements in air by exploring the behavior of an elastic balloon vibrating in response to an incident acoustic wave. In this presentation, we will present the results of experiments performed in an anechoic chamber, in which a pair of laser Doppler vibrometers simultaneously captured the velocities of the front and back surfaces of a Mylar balloon in an acoustic field. From phase measurements, the motion is described in terms of contributions from odd-order vibration modes (including bulk translation) and even-order vibration modes. The measured entrainment factors for the balloon at low frequencies are seen to be in good agreement with a physical model based on the scattering from an entrained, rigid sphere. This demonstrates the feasibility of using balloons for direct measurement of acoustic particle velocity in air.

11:00

1aEAb3. An experimental investigation of the nonlinear acoustic response of acoustic sense-ports. Thomas W. Teasley and David Scarborough (Aerosp. Eng., Auburn Univ., 211 Eng. Dr., Auburn, AL 36849, twt0007@auburn.edu)

Combustion instabilities continue to hinder the development of rocket engines and high-efficiency, low NOx combustion technology used in gas turbine engines. Experimental pressure measurements remain the best method to assess combustion instabilities. However, the harsh, high-temperature environment requires remotely mounting pressure sensors using sense-ports, which cause large discrepancies in measured thrust chamber acoustic pressure amplitudes. For this study, a multi-microphone impedance tube was used to investigate the nonlinear response of an acoustic sense-port. Measurements were performed for frequencies and driving amplitudes ranging from 100 Hz to 1500 Hz and 120 dB to 175 dB, respectively. Measurements were made using four different sense-port area-contraction ratios and for different extension tube lengths. The sense-port and extension tube acoustic responses were measured separately to enable the determination of the abrupt area contraction acoustic response. Measured sense-port area contraction length corrections were found to be in close agreement with the literature. The rigidly terminated sense-port extension tube exhibited linear acoustic damping. Measurements of the abrupt area contraction acoustic response revealed highly nonlinear damping even at low acoustic pressure amplitudes due to flow separation at the abrupt area contraction caused by the local acoustic velocity.

MONDAY MORNING, 4 DECEMBER 2017 STUDIO 2, 7:50 A.M. TO 12:00 NOON

Session 1aNS

Noise and Physical Acoustics: Supersonic Jet and Rocket Noise I

Caroline P. Lubert, Cochair
Mathematics & Statistics, James Madison University, 301 Dixie Ave., Harrisonburg, VA 22801

Kent L. Gee, Cochair
Brigham Young University, N243 ESC, Provo, UT 84602

Chair's Introduction—7:50

Invited Papers

7:55

1aNS1. Sixty years of launch vehicle acoustics. Caroline P. Lubert (Mathematics & Statistics, James Madison Univ., 301 Dixie Ave., Harrisonburg, VA 22801, lubertcp@jmu.edu)

On 4 October 1957 at 7.28 pm, the first artificial low Earth orbit satellite, Sputnik, was launched by the Soviet Union. Its launch ushered in a host of new scientific and technological developments, and public reaction in the United States led to the so-called "Sputnik Crisis," and the subsequent creation of NASA. A race ensued between the United States and the Soviet Union to launch satellites using carrier rockets. At this time, very little was known about the acoustics of rocket launches, and even less about acoustic suppression. Thus, in the vicinity of the rocket, acoustic levels could reach up to 200 dB during lift-off. Such extremely high fluctuating acoustic loads were a principal source of structural vibration, and this vibro-acoustic interaction critically affected correct operation of the rocket launch vehicle and its environs, including the vehicle components and supporting structures. It soon became clear that substantial savings in unexpected repairs, operating costs, and system failures could be realized by even relatively small reductions in the rocket launch noise level, and a new discipline was born. This paper presents a review of the first 60 years of launch vehicle acoustics.

8:15

1aNS2. On the acoustic near field of a solid propellant rocket. Christopher Tam (Mathematics, Florida State Univ., 1017 Academic Way, Tallahassee, FL 323064510, tam@math.fsu.edu)

Recently, Horne *et al.* presented NASA Ames measurements of the acoustic near field of a solid propellant rocket. The experiment consists of two phases: the high-burn and the low-burn phase. The main objective of this investigation is to use this set of data for the determination of the dominant components of near field rocket noise. The data consist of spectral measurements of an array of 14 near field microphones and a single far field microphone. By itself, the data are insufficient to accomplish the stated objective. We supplement the data with information provided by the two-noise source model of hot supersonic laboratory jets. The two-noise source model is supported by the existence of two similarity spectra. By applying the similarity spectra to the data of Horne *et al.* at low-burn, we are able to show that the dominant components of near field solid propellant rocket noise are the same as those found in the far field of supersonic jets. Further, the data allow the development of a model for the spatial distribution of the dominant noise components. On applying the model to the high-burn phase of the experiment, excellent agreements are found.

8:35

1aNS3. Investigation of single and twin jet interactions with plates with cut-outs. Karthikeyan Natarajan (Experimental AeroDynam. Div., CSIR-National Aerosp. Labs., EAD, PB 1779, Old Airport Rd., Bangalore 560017, India, nkarthikeyan@nal.res.in) and Lakshmi Venkatakrishnan (Experimental AeroDynam. Div., CSIR-National Aerosp. Labs., Bangalore, Karnataka, India)

The design and testing of the launch vehicle structure and its subsystems to withstand the lift-off acoustic loads is quite a daunting task in itself. Any effort to minimize these loads can prove to be highly beneficial as it directly influences the design, weight, and qualification of the launch vehicle components and hence the overall vehicle operating cost and time. The components of launch pad such as the launch platform and jet blast deflector are known to be the principal noise sources of the intense acoustic loads generated during lift-off. They contribute to the overall noise levels experienced by the launch vehicle by either reflecting the noise generated by the jet exhaust or by creating additional sources of noise. Earlier studies showed that the presence of cut-outs in the launch platform significantly affects the overall acoustic loads experienced by the launch vehicle. The present paper attempts to characterize the influence of cut-outs in the launch platform on the noise levels experienced by the launch vehicle, by investigating single and twin jets impinging on flat plates with and without cut-outs at varying lift-off distances. The results from acoustic measurements carried out in the near and far-field of scaled down single and twin jet launch vehicle models are discussed. The paper also attempts to relate changes in acoustic field, brought about by the different platform configurations, to the changes in flow field through flow visualization using the schlieren technique.

8:55

1aNS4. Results of subscale model acoustic tests for H3 launch vehicle. Wataru Sarae, Keita Terashima (JAXA, 2-1-1 Sengen, Ibaraki, Tsukuba 305-8505, Japan, sarae.wataru@jaxa.jp), Seiji Tsutsumi (JAXA, Sagamihara, Kanagawa, Japan), Tetsuo Hiraiwa (JAXA, Kakuda, Japan), and Hiroaki Kobayashi (JAXA, Kanagawa, Japan)

A subscale acoustic test, the H3-scaled Acoustic Reduction Experiments (HARE), was conducted to predict liftoff acoustic environments of the H3 launch vehicle currently being developed in Japan. The HARE is based on 2.5% scale H3 vehicle models, which is composed with a GOX/GH2 engine and solid rocket motors, Movable Launcher (ML) models with upper deck water injection system and Launch Pad (LP) models with deflector and lower deck water injection systems. Approximately 20 instruments measured far/near field acoustic and pressure data. Preliminary results are presented in this presentation.

9:15

1aNS5. Experimental study of the aeroacoustic interaction between two supersonic hot jets. Hadrien Lambaré (CNES, CNES Direction des lanceurs 52 rue Jacques Hillairet, Paris 75612, France, hadrien.lambare@cnes.fr)

The first stage of space launchers often use multiple engines. The supersonic jet noise at lift-off is a major source of vibrations for the launcher's equipments and payloads. In parallel with the development of the Ariane 6 launcher and its launch pad ELA4, the CNES MARTEL test bench have been improved in order to study experimentally the aeroacoustic interaction between two hot supersonic jets (mach 3, 2000K). In collaboration with the PPRIME laboratory of the University of Poitiers, acoustic and PIV measurements have been made in the free jets configurations. Further test campaigns will study the interaction of jets inside the flame duct, and their impingement on the launch table.

9:35

1aNS6. Experimental study of plate-angle effects on acoustic phenomena from a supersonic jet impinging on an inclined flat plate. Masahito Akamine, Koji Okamoto (Dept. of Adv. Energy, Graduate School of Frontier Sci., Univ. of Tokyo, 5-1-5, Kashiwanoha, Kashiwa, Chiba 277-8561, Japan, akamine@thermo.t.u-tokyo.ac.jp), Susumu Teramoto (Dept. of Aeronautics and Astronautics, Graduate School of Eng., Univ. of Tokyo, Bunkyo, Tokyo, Japan), and Seiji Tsutsumi (Aerosp. Res. and Development Directorate, Res. Unit III, Japan Aerosp. Exploration Agency, Sagamihara, Kanagawa, Japan)

Acoustic waves from a rocket exhaust jet cause the intense acoustic loading. Because the exhaust jet impinges on a flame deflector at liftoff of a launch vehicle, an adequate understanding of the acoustic phenomena from a supersonic impinging jet is required for prediction and reduction of the level of the acoustic loading. The previous numerical studies on a supersonic jet impinging on an inclined flat plate [e.g., Tsutsumi *et al.* AIAA 2014-0882; Nonomura *et al. AIAA J.* **54**(3), 816-827 (2016)] suggested that the plate angle has a large impact on the characteristics of the acoustic waves from the impingement region. In the present study, experiments were carried out to discuss the effect of the plate angle on this acoustic phenomenon. By applying the acoustic-triggered conditional sampling to the schlieren visualization movies, which was proposed by the authors [AIAA 2016-2930], the phenomena around the impingement region were observed in detail. The results revealed that the plate-angle variation leads to the change in the characteristics of the acoustic waves from the impingement region, such as the source locations.

9:55

1aNS7. Modeling community noise impacts from launch vehicle propulsion noise. Michael M. James and Alexandria R. Salton (Blue Ridge Res. and Consulting, 29 N Market St., Ste. 700, Asheville, NC 28801, michael.james@blueridgeresearch.com)

Commercial space is an emerging and evolving market as evidenced by the vast array of launch vehicles under development as well as the growing number of active and proposed launch sites. Federal Aviation Administration regulations require all new spaceports and launch vehicles to acquire a license. Part of the application process requires an environmental review to address the potential noise impacts to the environment and local communities. Accurate predictions of noise exposure from launch vehicles require models that have been validated over a range of vehicle types, operations, and atmospheric conditions. A high-fidelity launch vehicle simulation model, RUMBLE, has been developed to predict community noise exposure from spaceport launch, reentry, and static rocket operations. RUMBLE implements industry standard modeling practices to efficiently compute sound pressure level time histories, maximum levels, and sound exposure levels using the vehicle's engine parameters and geo-referenced source/receiver definitions. An overview of RUMBLE's underlying physics and the results of initial validation efforts (using multiple full-scale launch measurements) will be presented.

10:15–10:30 Break

10:30

1aNS8. Large eddy simulations of launcher lift-off noise and comparisons to experiments on model flame trenches. Julien Troyes (ONERA, Châtillon Cedex, France), Francois Vuillot, Adrien Langenais (ONERA, 29 Ave. de la Div. Leclerc, CHATILLON F92322, France, francois.vuillot@onera.fr), Hadrien Lambaré, and Pascal Noir (CNES, PARIS, France)

During the lift-off phase of a space launcher, rocket motors generate harsh acoustic environment that is a concern for the payload and surrounding structures. Hot supersonic jets contribute to the emitted noise from both their own noise production mechanisms and their interactions with launch pad components, such as the launch table and flame trenches. The present work describes the results of computations performed by ONERA to predict the lift-off noise from reduced scale models of a flame trench. The results include both unsteady flow solution inside the flame trench and the computed noise on near and far field microphone arrays. Numerical computations involve two in-house codes: the flow solver CEDRE, used in LES mode to accurately predict the noise sources, and the acoustic code KIM to reconstruct the far field noise, thanks to an integral Ffowcs Williams and Hawkings porous surface approach. The computational model exactly reproduces the flame trench configurations used in a test campaign carried out by CNES at the MARTEL facility. Results are discussed and compared with experimental acoustic measurements on 48 microphones. Overall, the numerical results reproduce the acoustic measurements within 3 dB. To further improve these results, work is ongoing on acoustic nonlinear effects.

10:50

1aNS9. Refinements in RANS-based noise prediction methodology for complex high-speed jets. Dimitri Papamoschou and Andres Adam (Mech. and Aerosp. Eng., Univ. of California, Irvine, 4200 Eng. Gateway, Irvine, CA 92697-3975, dpapamos@uci.edu)

Recent efforts on RANS-based modeling of the noise reduction from multi-stream, high-speed jets have underscored the importance of flow properties on the outer surface of peak Reynolds stress (OSPS). In a time-averaged sense, the OSPS is expected to represent the locus of the most energetic eddies in contact with the ambient fluid. The acoustic Mach number on the OSPS is a proxy for the convective Mach number of those eddies, thus has strong impact on the modeling of the noise source and its suppression when the jet plume is distorted into an asymmetric shape. Therefore, accurate detection of the OSPS is a critical ingredient in the modeling. This is complicated by the fact that, in asymmetric jets, the Reynolds stress distribution can be highly irregular and detection of its maximum along a radial line can become problematic. The talk will present advanced algorithms for the detection of the OSPS and resulting improvements in the prediction of noise reduction.

11:10

1aNS10. Validation of aero-vibro acoustic simulation technique using experimental data of simplified fairing model. Seiji Tsutsumi, Shinichi Maruyama (JAXA, 3-1-1 Yoshinodai, Chuuou, Sagamihara, Kanagawa 252-5210, Japan, tsutsumi.seiji@jaxa.jp), Wataru Sarae, Keita Terashima (JAXA, Ibaraki, Tsukuba, Japan), Tetsuo Hiraiwa (JAXA, Kakuda, Japan), and Tatsuya Ishii (JAXA, Tokyo, Japan)

To predict harmful acoustic loading observed at lift-off of the launch vehicle, aero-vibro acoustic simulation technique is developed. High-fidelity large-eddy simulation with computational aeroacoustics based on the full Euler equations in time domain are employed to predict generation of the acoustic waves and their propagation to the fairing. Coupled vibro-acoustic analysis based on finite element method are applied to compute the transmitted acoustic wave into the fairing both in the time and frequency domains. Acoustic measurement of a simplified fairing model with a subscale liquid rocket engine is conducted, and validation study of the present technique is performed. Reasonable agreement is obtained for peaks of the acoustic spectrum taken inside the fairing model. Such peaks are found to be related to the internal acoustic modes and ring modes of the faring structure. However, further study is required to obtain quantitative agreement.

Contributed Papers

11:30

1aNS11. Jet noise prediction via coupling large eddy simulation and stochastic modeling. Joshua D. Blake (MS State Univ., MS State University HPC Bldg., Office 365, MS State, MS 39762, jdb621@msstate.edu), VASILEIOS SASSANIS (MS State Univ., Starkville, MS), David Thompson (MS State Univ., MS State, MS), Adrian Sescu (MS State Univ., Starkville, MS), and Yuji Hattori (Tohoku Univ., Sendai, Japan)

A novel method for efficient and accurate jet noise prediction is developed and tested in the framework of coupled LES and stochastic noise modeling. In the proposed method, the low frequency range of the acoustic spectrum, which corresponds to large turbulent structures, is resolved with an implicit LES model by employing higher-order spatial and temporal discretizations. The high frequency range, which corresponds to the fine-scale turbulence, is modeled via a stochastic broadband noise generation model. The inputs to the stochastic model are represented by statistics from an axisymmetric RANS jet simulation. The smaller synthetic scales are convected by larger scales, accounting for the effects of sweeping from the larger LES-resolved turbulent scales. The farfield acoustic data is obtained using either the Linearized Euler Equations, or an acoustic analogy based on the Ffowcs-Williams Hawkings method. The method is evaluated on cold and heated jets at different Reynolds numbers.

11:45

1aNS12. Validation of a hybrid nonlinear approach for jet noise prediction and characterization. Vasileios Sassanis, Joshua Blake, Adrian Sescu (Dept. of Aerosp. Eng., MS State Univ., 2041 Blackjack RD, 47 Scenic Pass, Starkville, MS 39759, vs501@msstate.edu), Eric M. Collins (Ctr. for Adv. Vehicular Systems, Starkville, MS), Robert E. Harris (CFDRC, Huntsville, AL), and Edward A. Luke (Comput. Sci. and Eng., MS State Univ., Starkville, MS)

In this study, a hybrid approach for non-linear jet noise predictions in complex environments is presented and validated. The method differs from traditional approaches in that interactions of the jet with the surrounding structures as well as non-linear disturbances propagating over large

distances are taken into consideration in order to quantify their effects on sound generation and propagation. The noise sources founded in the jet plume and the near-field are first computed using the full Navier-Stokes equations. The variables of interest are then interpolated into a second domain. After penalized, they are used as source terms in the non-linear Euler equations to calculate the sound propagation. The interpolation and penalization steps are performed using a buffer region designed by the principles of sponge layers. The effective one-way communication between the two domains and the capabilities of the buffer region to transfer the data from the NS to the Euler domain without any loss of detail is demonstrated. Results from two- and three-dimensional jets both in free space and interacting with solid obstacles are presented. Comparisons with DNC and experimental data in terms of sound pressure level spectra are in good agreement with the ones calculated by our method.

MONDAY MORNING, 4 DECEMBER 2017 BALCONY L, 8:55 A.M. TO 12:00 NOON

Session 1aSA

Structural Acoustics and Vibration and ASA Committee on Standards: Standards in Structural Acoustics and Vibration

Benjamin Shafer, Chair
Technical Services, PABCO Gypsum, 3905 N 10th St., Tacoma, WA 98406

Chair's Introduction—8:55

Invited Papers

9:00

1aSA1. A review of ASTM International standards relating to impact sound transmission in buildings. Jerry G. Lilly (JGL Acoust., Inc., 5266 NW Village Park Dr., Issaquah, WA 98027, jerry@jglacoustics.com)

Impact sound transmission in buildings is an important issue, especially in today's multifamily marketplace where the trend to replace carpet with hard surface flooring is becoming more popular every year. This presentation will discuss the many changes that have been made over the years to the laboratory and field test methods for evaluating impact noise in buildings, including new standards under development.

9:20

1aSA2. Acoustical Society of America Standards in Mechanical Shock and Vibration. Charles F. Gaumond (Standards, Acoust. Society of America, 14809 Reserve Rd., Accokeek, MD 20607, charles.f.gaumond.acoustics@gmail.com) and Neil B. Stremmel (Standards, Acoust. Society of America, Melville, NY)

The Acoustical Society of America maintains standards in several areas. We present the areas covered by ANSI/ASA S2: mechanical shock and vibration and ISO TC 108: mechanical shock, vibration, and condition monitoring. A brief description of each standard is given, with the overlap between ANSI and ISO standards. The work in each area is handled by committees and working groups that represent a diversity of opinions, which is crucial to creating a standard based on consensus of all interested parties. We also present the requirements for official participation in standards, that includes the rights to propose new standards, review and modify old standards, and vote. We present the reasonable fee structures for various types of organizations. Contact information for each committee and working group as well as the location of the information on the ASA Standards website are presented.

9:40

1aSA3. Development of a rating of the improvement of high-frequency impact noise. John LoVerde and David W. Dong (Veneklasen Assoc., 1711 16th St., Santa Monica, CA 90404, jloverde@veneklasen.com)

The authors have introduced a family of new impact noise isolation ratings for evaluating high-frequency impact sound, which encompasses the frequency range of 400-3150 Hz. The advantages of this rating include better correlation with subjective reaction and improved rank-ordering of finish flooring and sound mats. A natural extension of this metric is to define a rating of the improvement in impact isolation analogous to ΔIIC per ASTM E2179. The resulting metric ΔHIIC evaluates the improvement in high-frequency impact isolation due to floor coverings as measured in the laboratory. The research indicates that ΔHIIC predicts the result due to floor coverings considerably better than ΔIIC. Potential applications of the rating for the design and evaluation of floor-ceiling assemblies is presented.

10:00

1aSA4. Considerations for the evaluation of impact sound from heavy impact sources in adjacent spaces. William C. Eaton and Noral D. Stewart (Stewart Acoust. Consultants, 7330 Chapel Hill Rd., Ste. 201, Raleigh, NC 27607, chris_e1@sacnc.com)

The ability to limit a sound heard by a neighboring tenant in a mixed-use facility can be achieved by understanding how to control the source of the sound or affect its path. It is also important to evaluate or measure the sound in a way that considers what a tenant might hear. These aspects will be considered when studying impact sound resulting from the drop of heavy impact sources in adjacent spaces.

10:20–10:40 Break

10:40

1aSA5. Method for evaluating and predicting noise from hard, heavy impact sources. John LoVerde, David W. Dong, Richard Silva, and Antonella Bevilacqua (Veneklasen Assoc., 1711 16th St., Santa Monica, CA 90404, wdong@veneklasen.com)

Noise and vibration from activity in fitness facilities, in particular, drops of hard heavy weights, is a common source of disturbance and complaint in residential, commercial, and mixed-use building types. Mitigating systems exist, including specially designed rubber flooring tiles and products for architectural vibration isolation systems, but in the United States, there is no standardized method for evaluating the reduction in noise or vibration provided by these products and systems. Quantitative and comparable data of the effectiveness of these products are therefore lacking. The authors previously reported (Internoise 2015, Noise-Con 2016, ASA Boston 2017) a preliminary test method to evaluate athletic tile flooring with heavy weight drops, based on the reduction in floor vibration (delta-L_V) achieved due to the insertion of the products. Preliminary results indicated that the delta-L_V measurement adequately described the impact reduction due to athletic tile, over a certain frequency range, reasonably independent of structure. This paper continues the research and further refines the measurement method, verifies its use for designing fitness facilities, and examines additional parameters that may affect measuring and evaluating hard, heavy weight impacts.

11:00

1aSA6. Non-standard uses of the mechanical tapping machine in field measurements. Jerry G. Lilly (JGL Acoust., Inc., 5266 NW Village Park Dr., Issaquah, WA 98027, jerry@jglacoustics.com)

The mechanical tapping machine creates a continuous series of floor impacts which generate a steady-state sound field when measured indoors. Because each individual hammer has the same weight and free-falls from the same distance, the machine creates a calibrated and steady impulsive force into the floor structure. This presentation will illustrate potential future test methods for assessing sound transmission in buildings with examples to evaluate structure-borne sound transmission into structurally isolated anechoic and reverberation rooms.

11:20

1aSA7. Methods for measuring the vibratory response of the ground. James E. Phillips (Wilson Ihrig, 6001 Shellmound St., Ste. 400, Emeryville, CA 94608, jphillips@wiai.com)

A draft American National Standards Institute (ANSI) standard on "Methods for Measuring the Vibratory Response of the Ground" is nearing completion for review. This standard is being developed in parallel with a separate draft standard on "Methods for the Prediction of Ground Vibration from Rail Transportation Systems." The intent of both documents is to standardize methods that were initially developed more than 30 years ago and adopted by the Federal Transit Administration. This paper will outline the topics in the draft standard, the measurement, and analysis techniques typically associated with the methods described, and how the draft standard relates to existing standards pertaining to measuring transfer mobility.

11:40

1aSA8. Methods for calculating flanking noise transmission. David W. Dong and John LoVerde (Veneklasen Assoc., 1711 16th St., Santa Monica, CA 90404, wdong@veneklasen.com)

Airborne sound isolation between spaces is determined not only by the direct sound transmission through the separating assembly, but also flanking transmission, which is the structureborne transmission of vibration due to acoustical excitation in the source room. In many cases, the flanking paths dominate, such as where the separating assembly has a high level of sound transmission loss, and the ability to calculate these paths is therefore crucial if the isolation is to be accurately predicted. Methods for calculating impact noise have been developed in Europe [standard EN ISO 12354] and are currently in use in Europe and Canada; however, the methods are not commonly known or used in the United States. The authors report on use of these methods in calculating flanking noise transmission in recent projects.

Session 1aSP

Signal Processing in Acoustics, Underwater Acoustics, and Engineering Acoustics: Source Tracking with Microphone/Hydrophone Arrays I

Kainam Thomas Wong, Cochair
Dept. of Electronic & Information Engineering, Hong Kong Polytechnic University, DE 605, Hung Hom KLN, Hong Kong

Siu Kit Lau, Cochair
Dept. of Architecture, National Univ. of Singapore, Block SDE3, #01-06, 4 Architecture Drive, Singapore 117566, Singapore

Invited Papers

9:00

1aSP1. Weighted coherent processing on sparse volumetric vector sensor arrays. Brendan Nichols, James S. Martin (Mech. Eng., Georgia Inst. of Technol., 801 Ferst Dr. NW, Atlanta, GA 30309, bnichols8@gatech.edu), Christopher M. Verlinden (Phys., U.S. Coast Guard Acad., La Jolla, CA), and Karim G. Sabra (Mech. Eng., Georgia Inst. of Technol., Atlanta, GA)

A network of drifting sensors, such as vector sensors mounted to freely drifting buoys, can be used as an array for locating acoustic sources underwater. Localizing a source using traditional coherent processing methods has been improved through arbitrary selection of element-wise weights of the covariance matrix [Nichols and Sabra, JASA 2015, Vol. 138]. However, the selection of weights offers an opportunity to optimize localization performance measures such as accuracy or precision. Here, the performance of source localization is compared between optimal weightings and traditional weightings for both simulated and at-sea data collected from a freely-drifting vector sensor array deployed in the Long Island Sound.

9:20

1aSP2. Tracking aircraft flybys using a microphone array. R. Lee Culver (ARL, Penn State Univ., PO Box 30, State College, PA 16804, rlc5@psu.edu), Stephen M. Tenney, and John Noble (Acoust., US Army Res. Lab, Adelphi, MD)

Ferguson and Quinn (JASA 96(2), 1994) develop equations to describe the time-frequency characteristics of an acoustic signal radiated by an aircraft and received by a stationary microphone. These equations enable construction of curves in time-frequency space whose shape depends upon several parameters, namely, aircraft speed and height above the microphone. We have utilized Ferguson and Quinn's equations to track an aircraft fly by under very low signal-to-noise conditions using six microphones which are spaced too far apart to form beams. We make use of signal coherence across the array, and we develop a time-frequency "matched filter" with replicas that are based upon aircraft speed, height above ground, and closest point of approach distance.

9:40

1aSP3. Source tracking with linear and circular compact vector sensor arrays. Berke M. Gur (Mechatronics Eng., Bahcesehir Univ., Istanbul, Turkey), Tuncay Akal, Abdurrahman Uslu, and Melih Yildirim (SUASIS, Karanfil Sok No:1 Masukiye, Kartepe, Kocaeli 41295, Turkey, tuakal@suasis.com)

Vector sensors are directional acoustic sensors that can make collocated measurements of both the acoustic pressure and the particle motion (in general velocity or acceleration). Combining the particle acceleration or velocity measurements with pressure, it is possible to estimate the intensity of the acoustic field, which in turn is related to the direction of the net acoustic energy propagation. Recently, several novel beamforming and array processing methods have been proposed that enable the development of compact linear and circular vector sensor arrays with inter-sensor spacing much less than the traditional spacing of one-half the design wavelength. These methods, albeit differing in implementation, both rely on the extraction and processing of the so-called "acoustic modes" of the sound field and have shown to be successful in estimating source direction relative to the array in a 2-D setting. The work described here, builds on previous results and extends the direction-of-arrival estimation methods to source tracking. Several algorithms developed for this purpose and implemented on both array types are introduced. The proposed approaches are experimentally validated using air-borne and underwater sources for compact pressure and 1-D vector sensor arrays.

10:00

1aSP4. Tracking time varying multipath phase at very low signal to noise ratios. Paul J. Gendron, Hanna Desiltes (ECE Dept., Univ. of Massachusetts Dartmouth, 285 Old Westport Rd., North Dartmouth, MA 02747, pgendron@umassd.edu), and Jacob L. Silva (ECE Dept., Univ. of Massachusetts Dartmouth, Darmouth, MA)

Diverse acoustic environments support multipath propagation. Estimating the arrival gains allows receiver structures to coherently combine acoustic energy for improved reception. In the case of mobile acoustic sources and receivers or dynamic boundaries, the amplitudes and phases of those arrivals are time varying such that estimation is made more difficult and typically entails some kind of phase tracking loop. This can be challenging as trackers are often formed as recursions in time employing previous observations to predict the phase at present, updating each phase estimate with the most recent observation. At extremely low SNR, recursive tracking can suffer loss of lock. Since a great deal of information about the instantaneous phase is contained in observations before and after it is judicious to seek estimators that exploit all of the observations in the signalling interval to infer the time varying response phase. Presented here is a scheme for simultaneous time varying phase estimation from the reception of a long duration, high time-bandwidth product transmission. In this approach, the phase process is derived from a minimum mean square error estimate of the sparse acoustic time varying response under an assumed sparse mixture model prior over Doppler and frequency. Demonstrations in shallow water at less than -12 dB received signal to noise ratio are presented.

10:20–10:35 Break

10:35

1aSP5. Detection, localization, and classification of multiple ocean vehicles over continental-shelf regions with passive ocean acoustic waveguide remote sensing. Chenyang Zhu (Northeastern Univ., 360 Huntington Ave., Boston, MA 02115, zhu.che@husky.neu.edu), Nada Saghir (Lawrence Technolog. Univ., Southfield, MI), Haoqing Li (Northeastern Univ., Boston, MA), WEI HUANG (Northeastern Univ., Malden, MA), Olav Rune Godø (Inst. of Marine Res., Bergen, Norway), and Purnima R. Makris (Northeastern Univ., Boston, MA)

Multiple ocean vehicles, including both surface ships and unknown submerged vehicles, can be simultaneously monitored over instantaneous continental-shelf scale regions via passive ocean acoustic waveguide remote sensing (POAWRS) by employing a large-aperture densely sampled coherent hydrophone array system. Here, the approach is demonstrated for multiple merchant ships present in the Norwegian Sea in 2014. The sounds radiated underwater by ocean vehicles, dominated by narrowband tonals and cyclostationary signals, are detected in the beamformed spectrograms. Coherent beamforming of the receiver array data significantly enhances the signal-to-noise ratio of the ship-radiated signals enabling detection of ocean vehicles roughly two orders of magnitude more distant in range than a single hydrophone. The estimated bearing-time trajectory of a sequence of detections are employed to determine the horizontal location of each vehicle using the moving array triangulation technique. The estimated locations are verified by comparison with historical database of merchant ships present in the region. Several time-frequency characteristics are extracted from the detected signals and used to classify and track the ships.

10:55

1aSP6. Echolocation and flight strategies of aerial-feeding bats during natural foraging. Emyo Fujioka (Organization for Res. Initiatives and Development, Doshisha Univ., 1-3 Tatara-miyakodani, Kyotanabe, Kyoto 610-0321, Japan, emyo.fujioka@gmail.com), Fumiya Hamai, Miwa Sumiya, Kazuya Motoi (Faculty of Life and Medical Sci., Doshisha Univ., Kyoto, Japan), Dai Fukui (Graduate School of Agricultural and Life Sci., The Univ. of Tokyo, Hokkaido, Japan), Kohta I. Kobayasi, and Shizuko Hiryu (Faculty of Life and Medical Sci., Doshisha Univ., Kyoto, Japan)

Aerial-feeding bats actively emit sonar sounds and capture large amounts of airborne insects a night. Microphone-array system allows us to know not only the positions where the bat emits sonar sounds (i.e., 3-D flight path) but how the bats dynamically control the acoustical field of view during searching and approaching target-prey. Here, we show echolocation strategy of bats during natural foraging revealed by the large-scale microphone-array system which covered the horizontal area of approximately 20 m $\times$ 20 m. *Pipistrellus abramus* was found to expand the width of their sonar beams in both horizontal and vertical planes just before the prey-capture. Since the bats emit echolocation pulses at a high rate (i.e., feeding buzz) just before capturing, the capture positions can additionally be measured. Recently, we have investigated the relationship between flight patterns, capture positions, and foraging efficiency of *Myotis macrodactylus* during natural foraging above the pond. Further investigation from the viewpoint of the optimal foraging would reveal echolocation and flight strategies of the bats to efficiently search and approach prey items. [This research was supported by a Grant-in-Aid for Young Scientists (B) and Scientific Research on Innovative Areas of JSPS, and the JST PRESTO program.]

11:15

1aSP7. Adaptive cubature Kalman filtering for distant speech tracking using a circular microphone array. Xiang Pan, Yue Bao, Yiting Zhu, Zefeng Cheng, and T. C. Yang (School of Information and Electron. Eng., Zhejiang Univ., Zhe Rd. 38, Hangzhou, Zhejiang 310027, China, panxiang@zju.edu.cn)

A joint processing framework for distant speech tracking is proposed based on combination of an adaptive cubature Kalman Filter with a 80-element circular microphone array. The deconvolved convention beamforming (DCB) [T. C. Yang, 10.1109/JOE.2017.2680818] is carried out over the distant speech data for speech enhancement and bearing estimate. DCB provides superdirective beams and offers the same robustness as conventional beamforming. With the estimated speech bearing, a "current" statistical (CS) motion model is combined with a cubature Kalman filter to track the speech of interest. The CS model assumes a modified Rayleigh distribution for the acceleration probability density (of the moving source) whose mean value is the current acceleration. Based on this model, we utilize a zoom factor achieved by computing the norm of the tracking innovation to adjust the dynamic state model adaptively. Both numerical simulation and the outdoor experimental results show that the proposed framework in this paper can effectively track a maneuvering speech source.

Session 1aUW

Underwater Acoustics: Underwater Acoustic Scattering and Reverberation

Brian T. Hefner, Cochair
Applied Physics Laboratory, University of Washington, 1013 NE 40th Street, Seattle, WA 98105

Edward Richards, Cochair
Scripps Oceanography, University of California, San Diego, 9500 Gilman Drive, La Jolla, CA 92093

Contributed Papers

8:15

1aUW1. Coherence in shallow water reverberation. Dajun Tang (Appl. Phys. Lab, Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, djtang@apl.washington.edu)

Coherent structure of reverberation intensity versus time and depth is observed during TREX13, and the structure resembles that of sound propagation versus range and depth. The experiment was conducted in shallow water approximately 20 m depth with both source and vertical receiving arrays moored, so the ping-to-ping incoherence in reverberation is believed to be caused by time-dependence of water column properties, especially rough sea surfaces. To understand the observed coherence, simulation of reverberation field is performed, where reverberation pressure, rather intensity, is calculated so phase coherence is included. Possibility of inferring sound propagation information from measured reverberation is discussed. [Work supported by the US Office of Naval Research.]

8:30

1aUW2. Modelling broadband scatter from periodic sea surfaces. Edward Richards, William S. Hodgkiss, and Hee-Chun Song (Scripps Oceanogr., Univ. of California, San Diego, 9500 Gilman Dr., La Jolla, CA 92093, edwardlrichards@ucsd.edu)

Surface waves are the primary source of short time-scale variation of the channel impulse response in underwater transmission scenarios when source and receiver positions are fixed. This study proposes an exact method for modeling the single surface bounce path of the channel impulse response, which contains most of short time-scale effect. This method is designed for one dimensional, periodic wave profiles, and solves the scattering problem as a series of motionless surfaces. To create a time series for each surface, the proposed method solves the Helmholtz equation at many frequencies, performing three steps at each frequency. First, the point source is decomposed into a series of plane waves, each of which scatter from the surface at a finite number of Bragg angles. Second, the amplitudes of these discrete scattered plane waves are found with an exact method. Finally, the surface response for a three-dimensional point source is recovered with a plane wave synthesis of these scattered plane waves. The results of this study are intended as a reference to inform the interpretation of surface scattering results from more common underwater acoustic models.

8:45

1aUW3. Delay-Doppler characteristics of surface interacting underwater sound. Sean Walstead (Naval Undersea Warfare Ctr. (NUWC), 1176 Howell St., Newport, RI 02841, sean.walstead@navy.mil)

The interaction of underwater sound with the ocean surface is investigated. Laboratory measurements of the delay-Doppler structure of the surface multipath at a transmission frequency of 300 kHz are compared to physics based analytic predictions. Channel impulse responses and channel scattering functions are measured when the surface is characterized by wind generated roughness, swell-like roughness, and a mix of wind and swell. The distribution of surface scattered energy across delay time and Doppler shift is presented as a function of wind speed ranging from 0.5 m/s to 11.0 m/s. The effect of wave shape and nominal out-of-plane grazing angle on the delay time spread and Doppler spread of the surface multipath are also considered. Proper characterization of surface interacting acoustic energy has direct application to the improved performance of shallow water and near surface underwater acoustic communications systems.

9:00

1aUW4. Machine learning the acoustic impulse response of a rough surface. Richard S. Keiffer and Steven Dennis (Acoust. Div., Naval Res. Lab., Bldg. 1005, Rm. C-2, Stennis Space Ctr., MS 39529, richard.keiffer@nrlssc.navy.mil)

Due to time constraints imposed by operational considerations, reverberation models used by the Navy split the computational work by modeling the propagation and the scattering separately. The scattering module is typically a "look-up" table linking the incident sound field at any point on the boundary to a pre-computed statistic of the scattering. There are at least two significant drawbacks to this standard modeling approach. First, each acoustic interaction with the boundary generates a unique realization of a stochastic process, and consequently, no single statistic of the scattering always can be justified and the correct modeling of the statistics of reverberation time series must acknowledge this event-to-event variation. Second, pre-computing the scattering based on an assumed statistical description of the unresolved boundary irregularities without consideration of the resolved boundary irregularities prohibits the coherent inclusion in that scattering calculation of the effects that the resolved boundary irregularities have on the scattering. Machine-learned models for the boundary scattering hold the promise of addressing both of these issues. The purpose of this talk is to explore these issues and the prospect of developing a machine-learned model for the acoustic impulse response of a multi-scale rough ocean boundary. [This work was supported by the U.S. Office of Naval Research.]

9:15

1aUW5. Modal analysis of split-step Fourier parabolic equation solutions in the presence of rough surface scattering. Mustafa Aslan (Turkish Naval Acad., Istanbul, Turkey) and Kevin B. Smith (Naval Postgrad. School, 833 Dyer Rd., Bldg. 232, Rm. 114, Monterey, CA 93943, kbsmith@nps.edu)

Determining accurate solutions of acoustic propagation in the presence of rough surface scattering is of significant interest to the research community. As such, there are various approaches to model the effects of rough surface scattering. Typically, the models used assume an idealized pressure release condition at the sea surface boundary. This boundary condition can easily be accommodated through a variety of modeling techniques for flat

surfaces, but it is necessary to introduce additional complex methods in numerical models for rough surfaces. Parabolic equation (PE) models utilizing split-step Fourier (SSF) algorithms have been employed to treat the rough surface displacements. These include methods such as the field transformational technique (FFT), and direct modeling of the physical water/air interface discontinuity. Previous work highlighted phase errors in SSF-based models with large density discontinuities at the interfaces, which were minimized by employing a hybrid split-step Fourier/finite-difference approach. However, such phase errors were largely absent in the presence of rough surface scattering. In this work, the PE solutions are decomposed into normal modes in order to determine which modes dominate the phase error in the presence of flat surfaces, and to confirm that these modes are highly scattered in the presence of rough surfaces.

9:30

1aUW6. An autonomous, bottom-mounted sonar for measurement of mid-frequency reverberation. Brian T. Hefner, Dajun Tang, and Taebo Shim (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, hefner@apl.washington.edu)

While bottom reverberation is typically dominant over surface reverberation at mid-frequencies in shallow water, sea surface roughness has a significant impact on transmission loss and hence on reverberation. Measurements of this effect can be difficult with a system deployed from a moving ship due to both the system motion and the constraints of operating the ship safely in the rough seas of interest. To overcome these difficulties, the Autonomous Reverberation Measurement System (ARMS) has been developed under an ONR-sponsored DURIP. This system is a benthic lander with a directional source and receive array mounted on a rotation stage. Powered by batteries and programmed prior to deployment, the system can measure reverberation from 2-6 kHz as a function of direction autonomously for up to 3 months. This system was deployed in the spring of 2017 as part of an experiment conducted off Geoje Island, Republic of Korea. During this experiment, the ARMS was deployed in 28 m of water and measured reverberation in a muddy sand environment with a range-dependent water depth that was punctuated with rock outcroppings on the seafloor. [Work supported by the U.S. Office of Naval Research.]

9:45

1aUW7. Experiments on the sound transmission at the water-air interface for different source-interface distances. Daniel Wehner and Martin Landrø (Dept. of GeoSci. and Petroleum, Norwegian Univ. of Sci. and Technol., S.P. Andersens Veg 15a, Trondheim, Sør-Trøndelag 7031, Norway, daniel.wehner@ntnu.no)

The water-air interface is a nearly perfect reflecting boundary for acoustic waves due to the high impedance contrast between the two media. In many marine applications, ranging from geophysical measurements to bioacoustics, the sea surface has an important impact. A rough sea surface topography leads to complex scattering and interference for acoustic frequencies with similar wavelength or higher than the wavelength of the interface. In geophysical applications and infrasound the frequencies of interest often have larger wavelengths. At the same time, the source within these applications is mostly located close to the interface with respect to its wavelength. In this case, an increased transmission could be expected as experiments with acoustic transducers demonstrate [D. C. Calvo *et al.*, *J. Acoust. Soc. Am.* **134**, 3403-3408 (2013)]. We design two experiments with sources close to the interface while changing the distance between the source and the water-air surface. For the first experiment, a water gun, which creates large cavities, is placed in water. Second, a signal gun, as used in athletics, is positioned in air. The acoustic transmission from both sides is measured and investigated. We find an increasing transmission coefficient for lower frequencies and decreasing distance between source and interface.

10:00–10:15 Break

10:15

1aUW8. Progress on applying the small-slope approximation to layered seafloors. Darrell Jackson (Appl. Phys., Univ. of Washington, 1013 NE 40th St, Seattle, WA 98105, drj@apl.washington.edu)

The small-slope approximation (SSA) for scattering by rough interfaces is an attractive alternative to the classic small-perturbation and Kirchhoff approximations, as it combines the best aspects of both and may be more accurate over all angular ranges. While SSA has most frequently been applied to impenetrable interfaces and interfaces bounding homogeneous half spaces, it has also been developed for layered media. There is no unique extension of SSA to layered media, and three different methods have appeared in the acoustics and electromagnetics literature. These methods will be reviewed and a new, possibly superior, method will be introduced. This method arises naturally when SSA is expressed in coordinate space rather than k-space. The various methods will be compared using numerical examples. [Work supported by ONR.]

10:30

1aUW9. Data analysis and modeling of broadband acoustic propagation perpendicular to internal wave fronts and sand dune crests in the South China Sea. D. Benjamin Reeder (Oceanogr., Naval Postgrad. School, 833 Dyer Rd., Bldg. 232, SP-311B, Monterey, CA 93943, dbreeder@nps.edu), A. Y. Chang (National Sun Yat-Sen Univ., Kaohsiung, Taiwan), Chi-Fang Chen (National Taiwan Univ., Taipei, Taiwan), Ching-Sang Chiu (Oceanogr., Naval Postgrad. School, Monterey, CA), Linus Chiu (National Sun Yat-Sen Univ., Kaohsiung, Taiwan), Chis Miller (Oceanogr., Naval Postgrad. School, Monterey, CA), Steven R. Ramp (Soliton Ocean Services, Carmel Valley, California), Ruey-Chang Wei (National Sun Yat-Sen Univ., Kaohsiung City, Taiwan), and Y. J. Yang (National Taiwan Univ., Taipei, Taiwan)

Very large subaqueous sand dunes were discovered on the upper continental slope of the northeastern South China Sea (SCS) in the spring of 2007 during an ONR 322OA-funded field experiment which was designed to study the large transbasin internal solitary waves (ISW) that are generated by tidal forcing in the Luzon Strait. These internal waves and sand dunes are important acoustical features, as it is expected that they will cause significant anomalies in the acoustical field. In the spring of 2014, a broadband source was deployed to transmit 850-1150 Hz LFM signals to receivers on a mooring in the center of the sand dune field. The acoustic transect was oriented perpendicular to the dune crests, ISW fronts and isobaths. Data analysis and extended modeling are presented to quantify the degree to which these features impact the propagation of broadband signals in the 100-2000 Hz band as a function of source depth and frequency.

10:45

1aUW10. Measurement of mid-frequency acoustic backscattering from the sandy bottom at 6–24 kHz in the Yellow Sea. Guangming Kan (Marine Geology and Geophys., First Inst. of Oceanogr., Rm. 109, Main Bldg., 6th Xianxialing Rd., Laoshan District, Qinfdao, Shandong 266061, China, kgming135@fio.org.cn), Baohua Liu, Zhiguo Yang, Shengqi Yu, Kaiben Yu (National Deep Sea Ctr., Qingdao, Shandong, China), and Yanliang Pei (Marine Geology and Geophys., First Inst. of Oceanogr., Qingdao, Shandong, China)

In a typical sandy bottom area of the South Yellow Sea, measurement of acoustic bottom backscattering strength within a frequency range of 6–24 kHz was conducted using omnidirectional sources and omnidirectional receiving hydrophones. In this experiment, with interference from scattering off the sea surface being avoided and the far-field condition being satisfied, we obtained acoustic bottom backscattering strength values ranging from -31 to -17 dB within a grazing angle range of $18°$–$80°$. In the effective grazing angle range, the acoustic scattering strength generally increases with the increase in the grazing angle, but the variation trends were different in different frequency bands, which reflects different scattering mechanisms. The frequency dependence of the acoustic backscattering strength is characterized by a segmented correlation. In the frequency band of 6–11 kHz, the scattering strength is positively correlated with the frequency, and the average slope of the linear correlation is about 0.83 dB/octave; in the frequency band of 12–24 kHz, the scattering strength generally exhibits a negative correlation with the frequency, and the slope of the linear correlation is about -0.42 dB/octave.

11:00

1aUW11. Time-domain acoustic scattering by elastic objects using finite element analysis. Blake Simon (Appl. Res. Labs., Univ. of Texas, 1400 Briarcliff Blvd., Austin, TX 78723, blakesimon8@gmail.com), Aaron M. Gunderson (Appl. Res. Labs., Univ. of Texas, Pullman, Washington), and Marcia J. Isakson (Appl. Res. Labs., Univ. of Texas, Austin, TX)

Finite element analysis provides an accurate way of calculating acoustic scattering from underwater objects. The method provides an exact solution to the Helmholtz equation to the order of the discretization. Although commercial finite element software is capable of solving fully 3D scattering problems, it has been previously shown by Zampolli *et al*. [*J. Acoust. Soc. Am.* **122**, 1472–1485 (2007)] that 3D axisymmetric targets can be solved more efficiently using 2D geometry. This method uses an axial wavenumber decomposition technique, which simulates an incident plane wave from an off-axis direction. In this study, results from this analysis are converted from the frequency domain to the time domain using Fourier synthesis. Elastic spheres and cylinders are considered because the analytical solutions for scattering by these targets are known and can be used to verify the finite element results. The success of this model to simulate time domain scattering will inform the applicability of finite element analysis for more complex targets. [Work supported by ONR, Ocean Acoustics.]

11:15

1aUW12. Late backscattering enhancement from a rubber spherical shell associated with waveguide coupling and propagation. Aaron M. Gunderson (Appl. Res. Labs., Univ. of Texas at Austin, Appl. Res. Labs., 10000 Burnet Rd., Austin, TX 78758, aaron.gunderson01@gmail.com), Timothy Daniel, Philip L. Marston (Phys. and Astronomy Dept., Washington State Univ., Pullman, WA), and Marcia J. Isakson (Appl. Res. Labs., Univ. of Texas at Austin, Austin, TX)

Acoustic scattering from rubber targets has received modest attention in the past due in part to rubber's subsonic sound speeds. In particular, slow shear waves in rubber get rapidly attenuated, giving rubber the fluid-like property of negligible shear coupling. This makes rubber a material of interest for coating or cloaking underwater objects and vehicles. In this study, backscatter from a rubber spherical shell in water is considered experimentally and through such models as partial wave series, finite elements, and waveguide normal mode analysis. The target is a commercially available American handball of unspecified rubber composition. Experimental and modeled results exhibit the importance of a strong, slightly subsonic late echo, which is demonstrated through path timing models and frequency analysis to be due to waveguide propagation through the shell wall. The various models use wave speeds for the rubber determined experimentally, allowing for determination of the phase and group velocities of the lowest order waveguide mode within the shell. These are found through normal mode analysis for both flat and curved waveguides, and again through Sommerfeld-Watson theory. Frequency domain results show how the waveguide path interferes with reflective paths from the shell in a frequency-dependent manner. [Work supported by ONR.]

11:30

1aUW13. On target excitation by modulated radiation pressure. Ahmad T. Abawi (HLS Res., 3366 North Torrey Pines Court, Ste. 310, La Jolla, CA 92037, abawi@hlsresearch.com), Ivars P. Kirsteins (NUIC, Newport, RI), Philip L. Marston, and Timothy Daniel (Phys., WSU, Pullman, WA)

Modulated radiation pressure (MRP), which can be produced by modulating an ultrasound carrier beam with a low-frequency signal provides the capability to interrogate elastic objects in a novel manner: The high-frequency carrier beam enables surgical investigation of target of interest, while the low-frequency modulation signal "shakes" the target to extract useful physical information. By sweeping the modulation frequency, target resonances can be identified, and by physically scanning the target at a modulation frequency that is commensurate with one of its resonant frequencies, the corresponding mode shape can be extracted. Up until now, the application of this technique has been limited to medical ultrasound, where it has been used to look for tumors and kidney stones among other things. The other applications of this technique have been particle trapping and non-contact manipulations. In this presentation, we will show results from experiments conducted at Washington State University involving scaled targets, complemented by finite element modeling results. We will discuss coupling between two targets near each other, while one of them is excited by MRP. We will also discuss how the radiated field from a target excited by MRP scales with its size. [Work supported by ONR.]

Session 1pAA

Architectural Acoustics and Psychological and Physiological Acoustics: Perceived Diffuseness I

Jin Yong Jeon, Cochair
Department of Architectural Engineering, Hanyang University, Seoul 133-791, South Korea

Ning Xiang, Cochair
School of Architecture, Rensselaer Polytechnic Institute, Greene Building, 110 8th Street, Troy, NY 12180

Chair's Introduction—1:00

Invited Papers

1:05

1pAA1. Review of the perception of diffusive surfaces in architectural spaces. Peter D'Antonio, Jeffrey Madison (RPG Acoust. Systems LLC, 99 South St., Passaic, NJ 07055, pdantonio@rpgacoustic.com), and Trevor J. Cox (Acoust. Eng., Univ. of Salford, Salford, United Kingdom)

The use of sound diffusors started in antiquity in the form of statuary, balustrades, coffered ceilings, and surface ornamentation. While these surfaces added both beauty and useful scattering, their bandwidth was limited. It was not until the invention of the reflection phase grating by Manfred Schroeder in 1973 that acousticians were able to design number theoretic surfaces with a specified broad bandwidth. Later Cox and D'Antonio expanded diffusive design options beyond number theoretic surfaces to include fractal surfaces, binary amplitude diffusors and beautiful curvilinear architectural shapes acceptable to the architectural community, using shape optimization software. Many peer review publications and books by the authors, ushered in the widespread use of sound diffusion to complement sound absorption. Significant research ensued to study various shapes, bandwidth, location, coverage, and their perceptual aspects. This presentation will review how sound diffusion modified the sound fields originally in recording control rooms, followed by home theaters, rehearsal spaces, performance stages and auditoria, worship spaces, and sound reproduction spaces. Many applications will be presented, as well as subjective questionnaires and objective measures.

1:25

1pAA2. Improving scattering surface design with rapid feedback by integrating parametric models and acoustic simulation. Louena Shtrepi, Jessica Menichelli (DENERG, Politecnico di Torino, C.so DC Degli Abruzzi 24, Torino 10129, Italy, louena.shtrepi@polito.it), Arianna Astolfi (DENERG, Politecnico di Torino, Turin, Italy), Tomas Mendez Echenagucia (Inst. of Technol. in Architecture Block Res. Group, ETH Zurich, Zurich, Switzerland), and Marco C. Masoero (DENERG, Politecnico di Torino, Torino, Italy)

Acoustic panels need to be optimal for both acoustic performance and aesthetic values, in order to respond to the requirements of both architects and acousticians. However, current practice shows that there is a gap in interaction between these professionals. Hence, new strategies in enhancing the efficiency of the design should be investigated. This study proposes a new design process for diffusive surfaces by integrating parametric models and acoustic simulation, aiming to provide architects with rapid visual and acoustic feedback. The new process consists of three parts: (1) basic geometric guidelines for diffusive surface design, which give the necessary information to architects with no acoustic background; (2) programming a parametric model of a diffusive surface sample in *Rhinoceros,* which allows for very quick variations in the design and suits the flexibility required by architects' design practice; (3) connecting *Rhinoceros* and *a simple ray tracing*, which adds acoustic simulation functionality and visual output to facilitate architects' utilization and feedback comprehension. Finally, a characterization of the scattering coefficient is performed in a scale model based on ISO 17497-1 in order to verify the effectiveness of the new design process.

1:45

1pAA3. Exploration of perceptual differences of virtually enhanced sound fields using timbre toolbox. Song Hui Chon and Sungyoung Kim (Elec., Comput., and Telecommunications Eng. Technol., Rochester Inst. of Technol., ECT Eng. Technol., ENT Bldg. 82-2155, 78 Lomb Memorial Dr., Rochester, NY 14623, songhui.chon@rit.edu)

In this presentation, we approach the analysis of perceived spatial differences from the perspective of timbre using the Timbre Toolbox [Peeters *et al*., 2011, *J. Acoust. Soc. Am.*, 130, 2902–2916]. The Timbre Toolbox, using MATLAB, calculates 88 descriptors that have been reported in various timbre perception literature. These include temporal (such as RMS energy), spectral (both in magnitude and power spectra), and harmonic descriptors, as well as those based on the Equivalent Rectangular Bandwidth (ERB) model, which approximates the auditory processing. Analyses were performed on 27 stimuli of nine string quartet groups performing one excerpt in three room conditions, as well as the data from perceptual experiments reported in an earlier study [Chon *et al*., 2015, Proc. Audio Eng.

Soc.]. The three room conditions have different spatial profiles, which consist of one natural room and two virtually enhanced rooms, where many physical parameters were closely controlled (e.g., ST1 and ST2). The experimental data include both the musicians' ratings of spatial quality for performance (e.g., tonal balance and envelopment) as well as recording engineers' ratings of recorded spaces (e.g., clarity and room size). The best timbre descriptor for each question will be reported, and an overall merit of this approach discussed.

2:05

1pAA4. Distinguishing between varying amounts of diffusion subjectively and objectively. Jay Bliefnick and Lily M. Wang (Durham School of Architectural Eng. & Construction, Univ. of Nebraska-Lincoln, 1110 S 67th St., Omaha, NE 68182-0816, jbliefnick@huskers.unl.edu)

This paper summarizes findings from a study of both subjective perception and objective quantification of varying amounts of diffusion. Numerous impulse response measurements were collected from a physical acoustics testing facility that featured reversible absorptive/diffusive/reflective wall panels. The collected room impulse response measurements were utilized in subjective perception trials as well as an objective metric analysis. Subjective testing used auralizations from the measured impulse responses in audio comparison trials to determine how well diffusive room conditions could be discerned. It was found that more than 50% coverage of diffusive surface area was required for the average subject to discriminate between diffusive and absorptive wall conditions. Subjects were even less capable of discerning between diffusive and reflective wall conditions. The objective metric analysis identified the Number of Peaks methodology as the most effective metric for assessing diffusive room conditions from amongst those tested.

2:25

1pAA5. Assessing perceived diffuseness in music venues. Jin Yong Jeon and Kee Hyun Kwak (Dept. of Architectural Eng., Hanyang Univ., Seoul 133-791, South Korea, jyjeon@hanyang.ac.kr)

The effect of diffusing elements on the variance of indoor acoustical quality is defined as "diffuseness," and the perception of diffuse sounds is mainly affected by the absorption and diffusion coefficients of surface materials in performance spaces. To characterize sound-field diffuseness, the acoustic parameters of a room, along with impulse-based diffusivity indices are calculated from the binaural room impulse responses. Then, auditory experiments are carried out for using convolved sounds, by investigating the acoustical attributes defined for perceived diffuseness: density of reflection, smoothness of decay, smoothness of reflection, and isotropic directivity. Diffuseness normally reduces the bass ratio and increases the brilliance, thereby increasing the clarity but decreasing the sound strength. In addition, the installed diffusers objectively increase envelopment, intimacy, and diffusion perception. It is seen that the direct sound level supported by the amount of diffuse reflections are related to the perception of diffuseness. With regard to the diffuseness for the acoustical preference in a space, the parameter Np, the number of peaks computed for the measured impulse responses, is effective for evaluating the diffuse sound fields influenced by early scattering reflections.

2:45–3:00 Break

3:00

1pAA6. Comparing the binaural recordings of 22.2- and 2-channel music reproduced in three listening rooms. Sungyoung Kim (RIT, ENT-2151, 78 Lomb Memorial Dr., Rochester, NY 14623, sungyoungk@gmail.com), Madhu Ashok (Univ. of Rochester, Rochester, NY), Richard L. King (McGill Univ., Montreal, QC, Canada), and Toru Kamekawa (Tokyo Univ. of the Arts, Tokyo, Japan)

A room interacts with sound sources. It alters both timbral and spatial impression of produced and reproduced sound field(s), and affects the overall sonic experiences. In this study, we approached the room-induced effect in the context of an immersive audio rendering, and investigated a relationship between room acoustics and audio reproduction formats. First, three classical music pieces were recorded and mixed optimally for the 22.2- and 2-channel reproduction formats. Subsequently, we generated a set of binaural responses to each of 22.2- and 2-channel reproduced music at three distinct listening rooms (varying dimensions and reflecting surfaces). Eleven listeners participated in a listening experiment; they compared two randomly selected binaural stimuli and rated perceived dissimilarity. The collected ratings were analyzed through the individual differential scaling (INDSCAL) to determine a perceptual space of the stimuli. The results show that (1) the listeners perceived the stimuli differences through two factors-the reproduction format (dimension 1) and the listening room (dimension 2); and (2) the room-induced perceptual difference was less dominant for the listeners compared to the format-induced one. The importance of each dimension was determined via the goodness of fit of the INDSCAL results as follows: Dimension 1-0.7138 and Dimension 2-0.1649.

3:20

1pAA7. On the effect of acoustic diffusers in comparison to absorbers on the subjectively perceived quality of speech in ordinary rooms. Ali Sanavi, Beat Schäffer, Kurt Heutschi, and Kurt Eggenschwiler (Lab. for Acoust. and Noise Control, Empa - Swiss Federal Labs. for Mater. Sci. and Technol., Überlandstrasse 129, Dübendorf 8600, Switzerland, a.sanavi@gmail.com)

Acoustic diffusers are used to control unwanted reflections responsible for degrading sound quality, or to increase sound diffuseness in spaces such as auditoria. To control unwanted reflections, diffusers are not the only solution. An alternative approach is the use of acoustic absorbers. Whether diffusers or absorbers are chosen as treatment depends on whether the energy conserved by the application of diffusers improves or detracts other aspects of room acoustics including the subjectively perceived qualities. However, not much is known to date about subjectively perceived qualities, e.g., of speech, in ordinary rooms. The aim of this study was twofold. The first aim was to investigate the effect of acoustic diffusers on the subjectively perceived quality of speech in ordinary rooms. The second aim was to determine if and to what extent there are perceptual differences if the diffuser are replaced by acoustic absorbers. Two separate listening tests were performed with stimuli obtained from the convolution of measured binaural impulse responses of a meeting room with excellent speech intelligibility and speech samples recorded in an anechoic chamber. The results of the listening tests confirm that despite the already excellent speech intelligibility and low values of early decay time, speech quality can be further improved by introducing diffusers or acoustic absorbers, with absorbers improving the subjectively perceived speech quality slightly more than the diffusers.

3:40

1pAA8. Theoretical model of reverberation decay in a rectangular room—Potential for predicting flutter echo. Toshiki Hanyu (Junior College, Dept. of Architecture and Living Design, Nihon Univ., 7-24-1, Narashinodai, Funabashi, Chiba 274-8501, Japan, hanyu.toshiki@nihon-u.ac.jp)

Reverberation is the most important factor governing the acoustic design of a room. It is known that non-exponential reverberation decay may occur in a room with unevenly distributed sound absorption, namely, with non-diffuse sound field. We propose the theoretical model of reverberation decay in non-diffuse sound fields. The reverberation decay is characterized by absorption and scattering properties of the surfaces of a room. As a first step for developing reverberation theory of non-diffuse sound field, the proposed model deals with rectangular rooms, because rectangular rooms are very common in any kind of buildings and architecture. Spatial absorption coefficients for three orthogonal directions are defined as the parameters which characterize the energy decay curve in the rectangular room. Within this model, the reverberation in such rectangular room becomes a combination of exponential and power law decay. The power law dependence as a function of time has a singularity. This issue is resolved within the model. The model was verified by comparison with the results of computer simulation using sound ray tracing method. Theoretical solutions of the proposed model agree well with the results of the computer simulation. Potential for predicting flutter echo by this model is also discussed.

4:00

1pAA9. Acoustic design of teachers' cafeteria in Tsinghua University. Xiang Yan (School of architecture, Tsinghua Univ., Rm. 104, Main Academic Bldg., Haidian District, Beijing 100084, China, yx@abcd.edu.cn) and Hui Li (Deshang Acoust., Beijing, China)

By the end of 2015, Tsinghua University has built a new cafeteria for 600 people. Dining period in university campus is not just eating, but an important conversation opportunity. The university president insists the noisy environment in past campus cafeteria must not happen again. So a comprehensive architectural acoustic design was taken. In order to meet the architect's aesthetic requirements, the restaurant uses a large area of seamless porous sandstone sound-absorbing material. By both ceilings/walls absorption treatments and dining tables layout, three acoustic design goals were reached: (1) mid-frequency reverberation time = 0.7s; (2) at the same table, the speaker's sound level attenuation of not more than 6dB, and passed to the adjacent table sound level attenuation of more than 9dB, this ensures the hearing's signal to noise ratio; (3) in busy meal peak, the average background noise does not exceed 65dB (A). Teachers are satisfied with the sound environment of the cafeteria.

MONDAY AFTERNOON, 4 DECEMBER 2017 SALON F/G/H, 1:00 P.M. TO 4:20 P.M.

Session 1pAB

Animal Bioacoustics and Acoustical Oceanography: Bioacoustic Contributions to the Characteristics and Variability of Soundscapes, Underwater, or Terrestrial

Bruce Martin, Cochair
JASCO Applied Sciences, 32 Troop Avenue, Suite 202, Dartmouth, NS B3B 1Z1, Canada

David R. Barclay, Cochair
Department of Oceanography, Dalhousie University, PO Box 15000, Halifax, NS B3H 4R2, Canada

Chair's Introduction—1:00

Contributed Papers

1:05

1pAB1. Investigating the performance of soundscape metrics using known data sources and numerical simulations. Bruce Martin (Oceanogr., Dalhousie Univ., 32 Troop Ave., Ste. 202', Dartmouth, NS B3B 1Z1, Canada, bruce.martin@jasco.com), David R. Barclay (Oceanogr., Dalhousie Univ., Halifax, NS, Canada), and Aidan Cole (JASCO Appl. Sci., Halifax, NS, Canada)

Metrics such as the acoustic complexity index, acoustic diversity index, entropy, evenness, and roughness have been correlated with anthropogenic and biologic sound sources in terrestrial soundscapes. The metrics offer the possibility of characterizing the presence of sound sources and the biodiversity of an environment in large datasets without having to perform detailed automated or manual analysis. However, the metrics are less successful at separating soundscape components and measuring biodiversity when applied to marine soundscapes. The reasons provided for the are the spectral overlap between anthropogenic and biologic sources, the wide range of source levels, long propagation ranges, and the difficulty comparing acoustics to observations, especially for species that are soniferous at night and hidden by day. In this analysis, we examine several popular soundscape metrics for marine data containing typical ambient noise, anthropogenic sources, or a single species. We then examine how the metrics change as different magnitudes and repetition rates of simulated marine life vocalizations are added to the data files.

1:20

1pAB2. Improving the evaluation of soundscape variability via blind source separation. Tzu-Hao Lin (Res. Ctr. for Information Technol. Innovation, Academia Sinica, 128 Academia Rd., Section 2, Nankang, Taipei 115, Taiwan, schonkopf@gmail.com), Tomonari Akamatsu (National Res. Inst. of Fisheries Sci., Japan Fisheries Res. and Education Agency, Ibaraki, Japan), Mao-Ning Tuanmu, Chun-Chia Huang (Biodiversity Res. Ctr., Academia Sinica, Taipei, Taiwan), Chiou-Ju Yao (National Museum of Natural Sci., Taichung, Taiwan), Shih-Hua Fang (Dept. of Elec. Eng., Yuan Ze Univ., Chung-Li, Taiwan), and Yu Tsao (Res. Ctr. for Information Technol. Innovation, Academia Sinica, Taipei, Taiwan)

Evaluation of soundscape variability is essential for acoustic-based biodiversity monitoring. To study biodiversity change, many researchers tried to quantify the complexity of biological sound. However, the analysis of biological sound remains difficult because the soundscape is made up of multiple sound sources. To facilitate the acoustic analysis, we have applied non-negative matrix factorization (NMF) to separate different sound sources in an unsupervised manner. NMF is a self-learning algorithm which factorizes a non-negative matrix as a basis matrix and an encoding matrix. Based on the periodicity information learned from the encoding matrix, biological chorus and the other noise sources can be efficiently separated. Besides, vocalizations of different species can also be separated by using the encoding information learned from multiple layers of NMF and convolutive NMF. In this presentation, we will demonstrate the application of NMF-based blind source separation in the analysis of long-duration field recordings. Our preliminary results suggest that NMF-based blind source separation can effectively recognize biological and non-biological sounds without any learning database. It can also accurately separate different vocalizing animals and improve acoustic-based biodiversity monitoring in a noisy environment.

1:35

1pAB3. Soundscape components in a shallow-water zone off the coast of Goa. Shyam Kumar Madhusudhana and Bishwajit Chakraborty (Geological Oceanogr. Dept., National Inst. of Oceanogr., CSIR - National Inst. of Oceanogr., Dona Paula, Goa 403004, India, bishwajit@nio.org)

Underwater acoustic data were collected around Grande Island off the coast of Goa, India using moored autonomous recording equipment during March 2016 and March 2017 for four and ten days, respectively. The study site, a shallow-water area in the vicinity of the Zuari river estuary and in proximity of a major shipping port, is presumed to lie along cetacean migration routes. Two different locations were chosen for data collection during each year—a reef site and an off-reef site. All four datasets were dominated by circadian choruses of fish and snapping shrimp. Fish species that were found to contribute significantly to the soundscape includes terapontidae, toadfish, and sciaenidae. Other biological contributions included vocalizations from cetaceans. For example, beaked whale clicks were observed in the 2016 recordings and humpback whale vocalizations were observed for six days in the 2017 recordings. Anthropogenic influence in the soundscape remained low during both years and the only contributing sources were sporadic vessels passing in the vicinity. Given the low measured levels of anthropogenic contribution, the findings of this study could be used as a baseline for performing impact assessments during any future offshore operations in the region.

1:50

1pAB4. Modeling sound fields generated by multi-nodal communications networks found in underwater industrial settings. Michael A. Stocker (Ocean Conservation Res., P.O. Box 559, Lagunitas, CA 94938, mstocker@OCR.org)

Increasingly as the sea becomes host to marine industrial operations, mid-to-high frequency multi-nodal communication systems are being employed to monitor equipment states and to control remotely controlled and autonomous underwater vessels used to service and tend underwater equipment. From a regulatory standpoint most of this equipment is modeled as single point noise sources and propagation concerns are ameliorated by high-frequency attenuation due to sound absorption through frequency-dependent chemical "relaxation" or "elasticity" characteristics of boric acid (BH_3O_3) and magnesium sulphate ($MgSO_4$) components in seawater, and assumptions of the directionality of high-frequency signals. But due to the multi-nodal aspect of their operations, these networks actually set up a broad sound field that can be operationally detected at distances in excess of 10km. In this paper we approximate the characteristics of these sound fields by propagation modeling under realistic, but hypothetical scenarios that would be found in offshore and subsea fossil fuel extraction and renewable energy harvesting operations.

2:05

1pAB5. Characterization of diverse coral types using passive acoustic recorded data from the lagoon area of the Kavaratti—An atoll of the Lakshadweep archipelago. Bishwajit Chakraborty (Geological Oceanogr. Dept., National Inst. of Oceanogr., CSIR - National Inst. of Oceanogr., Dona Paula, Goa 403004, India, bishwajit@nio.org), Kranthi Kumar Chanda, and Shyam Kumar Madhusudhana (Geological Oceanogr. Dept., National Inst. of Oceanogr., Goa, India)

The presence of sound in marine habitats can be an indicator of biological processes. Therefore, sound monitoring of the reef system is important to develop an in-depth understanding of the biodiversity, vulnerabilities, and potential changes of ecosystems. In order to understand the diversity related to the coral types, we have computed Acoustic Complexity Index (ACI) (temporal and spectral scales) using passive acoustic recorded data from the lagoon area off the Kavaratti—an atoll of the Lakshadweep archipelago. ACI is found to be the highest in location "A" where the coral type of *Acropora* sp. and ample fish diversity is observed. In location "C" where massive *Porites* sp. and *Pocillopora* sp. with significant acoustic assemblages are observed, ACI was found to be comparatively lower than the location "A." Whereas for "B," "D," and "E" locations, relatively low level of ACI index is obtained. The ACI is the lowest in location "B" where macroalgae with no fish species are seen. Among these three locations, "E" shows higher ACI due to the shrimp sound dominance in the rock substrate having *Porite* sp. coral type.

2:20

1pAB6. Variation in the soundscapes of Pacific coral reefs over multiple spectral, temporal, and spatial scales. Marc Lammers (Hawaii Inst. of Marine Biology, 46-007 Lilipuna Rd., Kaneohe, HI 96744, lammers@hawaii.edu), Eden Zang (Oceanwide Sci. Inst., Makawao, HI), Maxwell B. Kaplan, T Aran Mooney (Woods Hole Oceanographic Inst., Woods Hole, MA), Pollyanna I. Fisher-Pool (National Oceanic and Atmospheric Administration, Haleiwa, HI), and Russell Brainard (National Oceanic and Atmospheric Administration, Honolulu, HI)

Biological sounds occurring on coral reefs are increasingly recognized as important factors influencing reef dynamics and ecological processes. Soundscapes of coral reefs can be broadly divided into a low-frequency band (<1 kHz), dominated by sounds produced by acoustically active fish, and a high-frequency band (2–20 kHz) dominated by snapping shrimp and other invertebrates. Because acoustic activities in both bands are influenced by a variety of ecological (biotic) and environmental (abiotic) factors, coral reef soundscapes are characterized by considerable spatial and temporal variability. The drivers of this variability are not yet well understood, but likely provide important insights into ecosystem processes and condition. We report on an effort to quantify the acoustic activity in both the fish and snapping shrimp frequency bands across twelve coral reef sites in the Pacific Ocean separated by distances ranging from hundreds of meters to thousands of kilometers, including reefs across the Hawaiian Archipelago, the Northern Mariana Islands, and American Samoa. We use data obtained from long-term, bottom-moored acoustic recorders to document the variability observed on multiple temporal scales and examine environmental drivers correlated with this variability at each location and differences among locations.

2:35–2:50 Break

2:50

1pAB7. Contemporary distribution of the eastern North Pacific right whale in the Bering Sea. Dana Wright (Joint Inst. for the Study of the Atmosphere and Ocean, Univ. of Washington, 3737 Brooklyn Ave., Seattle, WA 98105, dana.wright@noaa.gov), Jessica Crance (Marine Mammal Lab, NOAA Alaska Fisheries Sci. Ctr., Seattle, WA), Daniel Woodrich, Arial Brewer (Joint Inst. for the Study of the Atmosphere and Ocean, Univ. of Washington, Seattle, WA), and Catherine L. Berchok (Marine Mammal Lab, NOAA Alaska Fisheries Sci. Ctr., Seattle, WA)

The eastern population of the critically endangered North Pacific right whale (NPRW; *Eubalaena japonica*) historically ranged in the eastern Bering Sea from the Aleutian Islands to St. Matthew Island (60.4°N), with limited ($n < 20$) detections further north (which some consider bowhead whale; *Balaena mysticetus*). Since the 1980s, most NPRW sightings have been isolated to the southeastern Bering Sea. In order to describe the current spatio-temporal distribution of NPRW, long-term passive acoustic recorders throughout the Bering Sea (2012–2016) were analyzed manually (10,204.2 days; 27–29% duty cycle) for the presence of NPRW, which were identified using the "up" and "gunshot" calls. NPRW were consistently detected during ice-free months in the southeastern Bering Sea, and intermittently during the same months northward to 59°N. NPRW were also detected at low calling activity within two eastern Aleutian Passes. Notably, NPRW were detected north of St. Matthew (61.6°N) in summer 2016 (July–Aug.). Up and gunshot calls north of 62°N in ice-free months and north of 58°N in ice-associated months could not be distinguished from bowhead whale. Together, these results indicate that NPRW currently range with certainty from the eastern Aleutian Passes to 61.6°N, but may range as far north as the Bering Strait.

3:05

1pAB8. Soundscape fishing: Spatial variability in a low-frequency fish chorus in the southern California kelp forest. Camille M. Pagniello, Jack Butler, Gerald L. D'Spain, Jules Jaffe, Ed Parnell, and Ana Širović (Scripps Inst. of Oceanogr., UC San Diego, 9500 Gilman Dr. #0205, La Jolla, CA 92093-0205, cpagniello@ucsd.edu)

The kelp forests off the coast of southern California support a diverse assemblage of fishes, many of which are known to produce sound. Here, the spatial variability of a low-frequency (325–545 Hz) fish chorus recorded at three sites near the kelp forests off La Jolla, California, is described. This chorus dominated the dusk soundscape at all sites in May/June 2015, 2016, and 2017. During these times, spectral levels around 400 Hz increased by approximately 30 dB over a period of 3 h between 19:00 and 22:00 local time. The location of the fish chorus was estimated during each year using beamforming and time difference of arrival (TDOA) techniques on signals recorded by either a two-element 30-m aperture linear seafloor array or an array with four-elements, 20-m aperture in a tetrahedral-shaped configuration. This location was relatively constant during the chorusing each night. Environmental factors such as temperature, macroalgae assemblage and bottom cover, and geological features were investigated as possible drivers of the spatial distribution of the chorus. [Research supported by California Sea Grant (R/HCME-28) and a Natural Sciences and Engineering Research Council of Canada (NSERC) Postgraduate Scholarship-Doctoral (PGS D-3).]

3:20

1pAB9. Soundscape stability in king penguin colonies. Daniel P. Zitterbart (Appl. Ocean Phys. & Eng., Woods Hole Oceanographic Inst., 213 Bigelow Lab., MS#11, Woods Hole, MA 02543, dpz@whoi.edu), Camille Toscani (CEFE, CNRS, Montpellier, France), Anna Nesterova, Celine Le Bohec (IPHC, DEPE, CNRS, Strasbourg, France), and Francesco Bonadonna (CEFE, CNRS, Montpellier, France)

King penguin colonies present an acoustically rich environment due to the highly vocal nature of these birds and the large abundance of birds within a colony (100.000+). Individual identification in adults and chicks is based on vocal signals. Penguins can recognize the call of their partner or chick even among thousands of calling individuals. However, the range of such identification is limited to a few meters, 8.8 m on average in the colony. In spite of the fact that king penguins are famous for their highly developed acoustic abilities, little is known regarding the overall acoustic colony structure. Using a distributed synchronized microphone array, we collected soundscape information over several months at several locations. We find location specific features in the soundscape that are stable over several weeks, regardless of environmental conditions and only require short integration time. We explore if current acoustic indices are able to capture these features, if soundscape stability information might be useful for navigation and if this information can be associated with current colony structure and spatial use.

3:35

1pAB10. An exploration of rhythm perception in African penguins (*Spheniscus demersus*). Irene A. Fobe (Psych., Rochester Inst. of Technol., 1319 Hornblend St., Apt. 8, San Diego, CA 92109, iaf8034@rit.edu), Caroline M. DeLong (Psych., Rochester Inst. of Technol., Rochester, NY), and K. T. Wilcox (Statistics, Rochester Inst. of Technol., Rochester, NY)

Rhythmic properties in penguin vocalizations may be unique to individuals. Rhythm perception is a cognitive ability previously thought to be exclusive to vocal-learning species who have the neurological complexities required to mimic conspecific and heterospecific vocalizations. Discovering rhythm perception in penguins would provide insight on penguins' ability to recognize kin using auditory cues, and discount theories constraining rhythm perception to vocal-learning animals. The goal of this study was to learn if African penguins (*Spheniscus demersus*) could perceive changes in rhythm using a habituation-dishabituation paradigm. Subjects were 32–38 African penguins housed at the Seneca Park Zoo in Rochester, NY. Penguins were played four rhythms at 4 kHz and head turns per bird were counted in 24 sessions. Each session was composed of ten familiarization trials followed by six test trials that alternated between the familiar and novel rhythm. The number of head turns per bird did not significantly increase from the last three familiarization trials to the first novel test trial. Results were inconclusive in showing evidence for auditory rhythm perception in penguins. This may be because subjects met the habituation criterion in only 9 out of 24 sessions. More research on auditory rhythm perception in penguins is needed.

3:50

1pAB11. Aerodynamic Sound Production from Flying Beetles. Rintaro Hayashi, John S. Allen (Mech. Eng., Univ. of Hawaii -Manoa, 2540 Dole St., Holmes Hall 302, Honolulu, HI 96822, alleniii@hawaii.edu), and Daniel Jenkins (Molecular BioEng., Univ. of Hawaii Manoa, Honolulu, HI)

The flapping wings of insects generate sounds during their flight. Recent interdisciplinary studies have extensively investigated the lift and thrust mechanisms produced from flapping flight; however, the associated sounds produced are less understood. Most studies have examined the amplitude and frequency with respect to wing beat reporting on the associated harmonic structure. The directionality and phase relationships have been examined in more detail in a few recent studies for flies and mosquitoes. In this study, we examine the sound generation mechanism from two invasive beetle species to Hawaii which are the Oriental Flower Beetle and the Coconut Rhinoceros Beetle. The wing beat, amplitude, and phase relationships are measured and determined with an eight element spherical microphone array. The mechanism of second harmonic generation for the different species is investigated with adaptive signal processing (Empirical Mode Decomposition) and complementary high speed optical video. The rotational motion of wing has a significant role in the second harmonic for the Coconut Rhinoceros Beetle. The different location and phase oscillation of the elytron for the two species results in different vortex generation during the down stroke.

4:05

1pAB12. The soundscape of bat swarms. Laura Kloepper, Yanqing Fu, Morgan Kinniry (Biology, Saint Mary's College, 262 Sci. Hall, Saint Mary's College, Notre Dame, IN 46556, lkloepper@saintmarys.edu), Robert L. Stevenson (Univ. of Notre Dame, Notre Dame, IN), Caroline Brighton, Christian Harding (Oxford Univ., Wytham, United Kingdom), Paul Domski (New Mexico Falconry Assoc., Albuquerque, NM), and Graham Taylor (Oxford Univ., Wytham, United Kingdom)

Brazilian free-tailed bats form large maternal colonies numbering in the millions across the American Southwest. Each night, the bats emerge from their roost to travel to foraging locations. During this emergence, individuals fly together in a dense, linear stream and exhibit collective group behavior. Upon morning return to the roost, the flight behavior of bats change and individuals fly in unpredictable paths, with little to no apparent collective group behavior, creating a swarm. To understand the different sensory challenges each of these flight scenarios pose to bats, we recorded the soundscapes of bats in streams and swarms using four different recording platforms: (1) stationary, ground-based directional and omnidirectional microphones, (2) a zip-line microphone that maneuvered through the bat stream and swarm and was monitored by video, (3) a microphone and thermal camera on a quadcopter that recorded bat flight behavior and signals at high altitudes during swarm re-entry, and (4) a trained hawk that flew through the bat stream while carrying a microphone unit and monitored with unique video. We report on the characteristics and variability of soundscapes for bat swarms, including different noise profiles the bats experience during streaming and swarming and the adaptive time-frequency signatures and flight behavior of individuals during group flight.

MONDAY AFTERNOON, 4 DECEMBER 2017 — BALCONY M, 1:00 P.M. TO 4:10 P.M.

Session 1pAO

Acoustical Oceanography and Underwater Acoustics: Acoustic Scattering from Hydrocarbons and Hydrothermal Vents

Daniela Di Iorio, Cochair
Department of Marine Sciences, University of Georgia, 250 Marine Sciences Building, Athens, GA 30602

Alexandra M. Padilla, Cochair
School of Marine Science and Ocean Engineering, University of New Hampshire, Forest Park APT 281, Durham, NH 03824

Christopher Bassett, Cochair
Resource Assessment and Conservation Engineering, National Marine Fisheries Service, Alaska Fisheries Science Center, 7600 Sand Point Way NE, Seattle, WA 98115

Chair's Introduction—1:00

Invited Papers

1:05

1pAO1. Long-term, quantitative observations of seafloor hydrothermal venting using an imaging sonar. Guangyu Xu (Geology and Geophys., Woods Hole Oceanographic Inst., 266 Woods Hole Rd., Woods Hole, MA 02543, gxu@whoi.edu), Darrell Jackson (Appl. Phys. Lab. Univ. of Washington, Seattle, WA), Karen G. Bemis (Dept. of Marine and Coastal Sci., Rutgers Univ., NB, NJ), and Anatoliy Ivakin (Appl. Phys. Lab. Univ. of Washington, Seattle, WA)

While connected to the NEPTUNE observatory operated by Ocean Networks Canada (ONC) at the Endeavour Segment of the Juan de Fuca Ridge (http://www.oceannetworks.ca/observatories/pacific), the Cabled Observatory Vent Imaging Sonar (COVIS) recorded an unprecedented long-term (>4 years) acoustic dataset capturing local hydrothermal venting. Processing of the acoustic backscatter data yields three-dimensional (3-D) images of plumes rising tens of meters from black smoker vents on a sulfide structure named Grotto. More importantly, analysis of the Doppler frequency shift in acoustic backscatter yields estimates of the flow rates of those plumes and their volume fluxes. Subsequent calculations based on the vertical variation in volume flux and a theoretical heat-to-volume-flux relationship for buoyancy-driven plumes give estimates of plume heat flux, which are essential for studying the temporal evolution of a hydrothermal system and its coupling with geological, oceanic, and biological processes. In addition to black-smoker plume observations, the ping-to-ping decorrelation of seafloor backscatter recorded by COVIS provides an acoustic indicator of diffuse-flow (i.e., low-temperature, clear hydrothermal discharge) distribution over Grotto and its surrounding areas. Furthermore, acoustic techniques for quantifying the temperature fluctuations, and ultimately, heat flux of diffuse-flow venting are currently under development. [Work supported by NSF.]

1:25

1pAO2. Observing the evolution and fate of free methane in the ocean. Thomas C. Weber, Elizabeth F. Weidner, Alexandra M. Padilla, Kevin M. Rychert, and Scott Loranger (Univ. of New Hampshire, 24 Colovos Rd., Durham, NH 03824, tom.weber@unh.edu)

Free methane gas is increasingly observed escaping the seabed in the world's oceans from sources that are either biogenic, typically in shallow sediments, or from deeper geologic reservoirs. Methane gas bubbles undergo a complicated journey as they rise toward the sea surface. Some or all of the methane may pass through the gas-liquid boundary, into aqueous solution, where it is eventually oxidized and can impact ocean chemistry. Some of the methane, particularly in shallow environments, may reach the atmosphere where it acts as a strong greenhouse gas. One of the key questions regarding the transport of methane upward from the seabed is how much goes where, and this question is being increasingly addressed using acoustic remote sensing techniques. Answering this question begins with seep detection and localization, now routinely performed on data collected with split-beam and multibeam echo sounders. Once located, observations of the bubble-plume backscattering cross section can be used to address questions of flux and vertical gas transport. Both narrow- and broad-band techniques and some of the associated challenges, including wobbly bubbles and multiple scattering in dense plumes, will be discussed.

Contributed Papers

1:45

1pAO3. Investigating bubble transport and fate in the watercolumn with calibrated broadband split-beam echosounder data. Elizabeth F. Weidner (Earth Sci., Univ. of New Hampshire, 24 Colovos Rd., Durham, NH 03824, eweidner@ccom.unh.edu), Thomas C. Weber (Mech. Eng., Univ. of New Hampshire, Durham, NH), and Larry Mayer (Earth Sci., Univ. of New Hampshire, Durham, NH)

The transport and eventual fate of gas bubbles in the marine environment is a topic of interest to researchers in numerous fields. Acoustic systems are commonly used to study bubble ebullition sites as they provide synoptic measurements of the watercolumn. However, the visualization of individual bubbles has traditionally required the use of point-source equipment, such as vehicle-mounted cameras. Here, we present an acoustic methodology for studying individual bubbles using a calibrated broadband split-beam echosounder. The extended bandwidth (14–24 kHz) provides a vertical resolution on the order of 10 cm, which allows for the discrimination of individual bubbles in the echogram. Split-beam phase differentiation provides phase-angle data which can be used to compensate for beam-pattern effects and precisely locate bubbles within the watercolumn. Bubble target strength is measured and compared to analytical models to estimate bubble radius, and bubbles are tracked through the watercolumn to estimate rise velocity. The resulting range of bubble radii (1–6 mm in radius) is similar to those found in other investigations, and the rise velocities are consistent with published models. Together, the observations of bubble radius and rise velocity offer a measure of gas flux.

2:00

1pAO4. Evidence of low-frequency multiple scattering of methane gas bubbles at Coal Oil Point, Santa Barbara, California. Alexandra M. Padilla, Scott Loranger, and Thomas C. Weber (Ctr. for Coastal and Ocean Mapping, Univ. of New Hampshire, Forest Park APT 281, Durham, NH 03824, apadilla@ccom.unh.edu)

Coal Oil Point, located in the Santa Barbara Channel, CA, is known for its prolific natural hydrocarbon seepage. These hydrocarbons are released at depths between 30 and 80 m, shallow enough that the hydrocarbons reach the surface. Transport via bubbles and droplets through the water column enables the exchange of hydrocarbons between the ocean and the atmosphere. Hydroacoustic observations from previous surveys have been used to map hydrocarbon seep distributions and estimate the flux of hydrocarbons from the seafloor in Coal Oil Point. On September 14, 2016, this area was revisited and surveyed using low-frequency sub-bottom profilers operating from 1 to 10 kHz. The goal is to estimate the gas flux of hydrocarbons from the hydroacoustic measurements and compare them to historic estimates for this area. Visual observations of dense bubble cloud structures of these seeps were seen and evidence of multiple scattering effects were observed in the hydroacoustic measurements. We examine how acoustic scattering of natural hydrocarbon gas bubbles is effected by multiple scattering and how this process affects the quantification of gas flux.

2:15–2:30 Break

2:30

1pAO5. Enhanced subsea leakage detection with broadband active acoustic sensor system. Geir Pedersen (Christian Michelsen Res. AS, P.O. Box 6031, Bergen 5892, Norway, geir.pedersen@cmr.no)

For field re-activation, improved/enhanced oil recovery through injection of gas in reservoirs, and geological storage of CO_2 (carbon dioxide), robust barriers must be provided and confirmed to avoid leakage. Natural seeps from the sea floor represents an additional challenge, as many detection systems will misinterpret them as an infrastructure leakage. Active acoustics (backscattering) has been successfully used for detection of even moderate underwater hydrocarbon gas releases at long ranges, using, e.g., scientific echosounders. However, CO_2 leakage behavior is particularly complex and operations in cold and deep waters make such leakages difficult to detect with active acoustics, due to its behavior and low acoustic impedance in liquid phase. Methodology for detecting CO_2 even in liquid phase is a prerequisite for many deep water operations, and at present no adequate technology exists for this purpose. In order to design an active acoustic system that can detect subsea leakage of hydrocarbons and CO_2 in liquid phase, the behavior and backscattering by droplets as a function of environmental parameters and acoustic frequency is simulated. Theoretical backscattering is further compared with *in situ* acoustic and optical measurements of CO_2, CH_4 (methane), and air, released and measured at depths from 1300m to surface, using a custom built gas release and measurement frame.

2:45

1pAO6. Measurements of methane bubble size distribution from a remotely operated platform in the presence of oil. Theodore F. Argo (Appl. Res. Assoc., Inc., 7921 Shaffer Parkway, Littleton, CO 80127, targo@ara.com) and Paul Panetta (Appl. Res. Assoc., Inc., Gloucester Point, VA)

Subsurface releases of crude oil and methane gas may occur naturally or due to manmade, sometimes catastrophic, events. To determine the quantity of oil and gas being released, measurements can be performed on both the crude oil droplets and methane bubbles to determine their sizes and concentration in the water column. An acoustic backscatter measurement, in the 20–110 kHz range, was developed using transducers mounted on a commercial ROV insonifying an intentional subsurface oil and gas release in a large wave tank. Bubble size distributions for methane mixed in a plume of crude oil were obtained by performing an inversion on acoustic backscattering measurements after accounting for transducer properties and sound transmission through the water column. Due to the rapid nature of the measurements, monitoring of the evolution of the plume in time was also possible.

Distributions were consistent with independent measurements using laser scattering techniques.

3:00

1pAO7. Acoustic investigations of natural seeps at GC600. Mahdi Razaz, Daniela Di Iorio, and James B. Kelly (Marine Sci., Univ. of Georgia, 247, Marine Sci. Bldg., 325 Sanford Dr., Athens, GA 30602, mrazaz@uga.edu)

We examine the acoustic signature of natural seeps in the Green Canyon block 600 of the Gulf of Mexico. The survey site was 2200'1600 m^2 at a depth of 1200 m. Near-bottom multibeam backscatter, side-scan sonar mosaic, and chirp sub-bottom profiles were collected using an AUV cruising at 40 m above the seafloor. The water-column profiles collected by the Kongsberg EM2040 at 200 kHz central frequency were utilized to detect and localize seeps. Geological information from the chirp sonar and side scan images will also augment these data to further our understanding of the geologic structure of natural seeps. Identified seeps from the backscatter intensity were visited with a Comanche ROV for a visual inspection. An oily plume was selected for further investigations and vertical upwelling will be monitored with an Acoustic Scintillation Flow Meter (ASFM), for the first time. Concurrent measurements include nearby Acoustic Doppler profilers, a vertical array of conductivity/temperature instruments and two video cameras (VTLC) for identifying bubble/droplet sizes and distribution and will provide invaluable reference data on temporal variability.

Invited Paper

3:15

1pAO8. Detection and characterization of hydrocarbon droplets using broadband echosounders. Scott Loranger and Thomas C. Weber (Earth Sci., Univ. of New Hampshire, 24 Colovos Rd., Durham, NH 03824, sloranger@ccom.unh.edu)

Investigation of the fate and transport of liquid hydrocarbons is limited by the small field of view of current instrumentation. Mass spectrometers, fluorometers, and megahertz sonars—the typical instrumentation for detection and classification of liquid hydrocarbons in the marine environment are limited to detections are ranges of less than a few tens of meters. Lower frequency (80–500 kHz) broadband acoustic backscattering from weakly scattering liquid hydrocarbon targets has been investigated using a novel droplet making device. Results show that such instrumentation should be capable of detections at significantly greater ranges than current instrumentation. The results are compared to a variety of models of acoustic scattering from spherical targets to determine the most accurate model for predicting the frequency response of weakly scattering spheres. The frequency response can be used to characterize the liquid hydrocarbon droplets, as long as the acoustic impedance of the hydrocarbon is well known for the range of temperatures and pressures affecting the droplet.

Contributed Paper

3:35

1pAO9. The acoustic properties of three crude oils at oceanographically relevant temperatures and pressures. Scott Loranger (Earth Sci., Univ. of New Hampshire, 24 Colovos Rd., Durham, NH 03824, sloranger@ccom.unh.edu), Justin P. Cole (Chemistry, Univ. of New Hampshire, Durham, NH), Christopher Bassett (Alaska Fisheries Sci. Ctr., National Marine Fisheries Service, Seattle, WA), and Thomas C. Weber (Mech. Eng., Univ. of New Hampshire, Durham, NH)

The detection and quantification of crude oil in ocean environments is dependent on adequately constrained acoustic properties (e.g., density and sound speed). However, there is a paucity of published acoustic property measurements of crude oil at oceanographically relevant temperatures and pressures. Three medium crude oil samples (Alaska North Slope, Angola Bavuca, and Angola Xikomba) were tested to better constrain these properties for oceanographic applications. A temperature (-10 to $30\,^{\circ}$C) and pressure (0.1 to 19.3 MPa) controlled sound speed chamber was developed for highly accurate differential time of flight measurements. Density and viscosity were also measured over the same temperature range. Finally, differential scanning calorimetry measurements (-40 to $50\,^{\circ}$C) were conducted to identify phase changes in crude oil constituents that may contribute to nonlinearities in acoustic properties as a function of temperature at a constant pressure. Results are compared to previously available models for sound speed, such as the PC-Shaft model, and density as a function of temperature and pressure. The results can also be used to fully constrain models of the shape of oil droplets in the marine environment as a function of size, an important input for models of acoustic scattering.

3:50–4:10 Panel Discussion

Session 1pEAa

Engineering Acoustics: General Topics in Engineering Acoustics III

Kenneth M. Walsh, Chair
K&M Engineering Ltd, 51 Bayberry Lane, Middletown, RI 02842

Contributed Papers

1:00

1pEAa1. A preprocessing method with Hankel matrix-based SVD in strong interference environment. Ge Yu, Shenchun Piao, Xiao Han, and Kashif Iqbal (Acoust. Sci. and Technol. Lab., Harbin Eng. Univ., Harbin, Harbin 150001, China, yuge221@hrbeu.edu.cn)

In sonar system, reverberation is an ignored interference as its characteristic is as same as target echo. It is hard to distinguish target in sever environment. Therefore, suppressing reverberation to increase signal-to-noise ratio is very important in underwater signal processing. In this paper, we proposed a pre-processing method which is based on singular value decomposition and Hankel matrix. This method utilized the correlation between target and reverberation is bad to detect target. The proposed method can suppress both the white noise and reverberation as is evident by the simulation results. The proposed method is also verified by processing the experimental data.

1:15

1pEAa2. Rotor noise control using 3D-printed porous materials. Con Doolan, Chaoyang Jiang (School of Mech. and Manufacturing Eng., Univ. of New South Wales, UNSW Sydney, Sydney, NSW 2052, Australia, c.doolan@unsw.edu.au), Danielle Moreau (Mech. and Manufacturing Eng., Univ. of New South Wales, Adelaide, South Australia, Australia), and Yendrew Yauwenas (Mech. and Manufacturing Eng., Univ. of New South Wales, Sydney, NSW, Australia)

The self-noise created by rotor blades is an important component of the noise generated by aeroengine fans, wind turbines, and propellers. It is created by the passage of boundary layer turbulence over the trailing edge of the blade, where a portion of the unsteady pressure near-field is converted to acoustic waves that subsequently propagate to an observer in the far-field. This paper explores the use of 3D-printed porous materials for controlling rotor self-noise. It is reasoned that the flexibility offered by 3D-printing makes it possible to produce blade trailing edges with tailored, spatially varying porosity that can maximise noise reduction, either by modifying the near-field turbulence or by affecting the production of waves at the trailing edge itself. A number of 3D-printed porous material samples were acoustically characterised using an impedance tube. Using this information as a guide, special rotor blade tips were 3D-printed with varying levels of porosity and evaluated using a rotor-rig and microphone array. Simultaneously, computational modelling of the rotor-rig was performed to understand the flow field and turbulence levels over the blade tip region.

1:30

1pEAa3. On the character and mechanics of flow-induced wall-mounted finite airfoil noise production. Danielle Moreau and Con Doolan (School of Mech. and Manufacturing Eng., UNSW, UNSW, Sydney, NSW 2052, Australia, d.moreau@unsw.edu.au)

A major source of noise involves flow interaction with wall-mounted attachments such as fins, guide vanes, aircraft airframe components, and wind turbine blades. In a simplified sense, these objects can be viewed as wall-mounted airfoils of finite length. A wall-mounted finite airfoil has one end immersed in the free stream and the other end fixed to a wall that is subject to a developing boundary layer. The free-end and wall junction introduce three-dimensional flow structures into the wake that affect noise generation in ways that are not yet fully understood. The primary goal of this research is to determine how airfoil end-effects and three-dimensional aerodynamics influence airfoil noise production. To accomplish this goal, anechoic wind tunnel studies have been conducted to determine the characteristics and mechanisms of flow-induced noise production for various wall-mounted finite airfoil configurations. The test models include combinations of airfoil profile shape, aspect ratio, angle of attack and free stream velocity. The motivation for this work is to gain a better understanding of sound generation from wall-mounted finite airfoils to aid future innovation in quiet aerodynamic design and optimisation.

1:45

1pEAa4. An experimental study on the role and function of the diaphragm in modern acoustic stethoscopes. Lukasz Nowak (Inst. of Fundamental Technolog. Res., Polish Acad. of Sci., ul. Pawinskiego 5B, Warszawa 02-106, Poland, lnowak@ippt.pan.pl)

Vibrations of a diaphragm of an acoustic stethoscope in contact with the body of an auscultated patient are the source of the sound transmitted to the ears of a physician performing examination. Mechanical properties of a diaphragm are supposed to significantly affect the parameters of the transmitted bioacoustic signals. However, the exact relation remains mostly unclear, as the underlying phenomena involve complex effects of acoustic coupling between the diaphragm and the body of a patient. The present study introduces a detailed methodology for determining vibroacoustic behavior of a diaphragm of a stethoscope during an auscultation examination. A laser Doppler vibrometer is used to measure the velocity of various points on the surface of a diaphragm during heart auscultation. Synchronized recordings of electrocardiography signals are used for segmentation. Representative data sets are selected and analyzed for various kinds of diaphragms. The results show significant differences in vibration velocity levels and their distribution across the surfaces of the considered structures. In this regard, it is also shown that the currently available solutions can be significantly improved by using structures better matched to the acoustic impedance of a body.

2:00

1pEAa5. A study on a friendly automobile klaxon production with rhythm. SangHwi Jee, Myungsook Kim, and Myungjin Bae (Sori Sound Enginnering Lab, Soongsil University, Seoul 06978, South Korea, slayernights@ssu.ac.kr)

We are always exposed to traffic noise in our lives. Especially in unproven situations, exposure to a sudden sound may result in loss of auditory cells and exposure to retaliatory driving. However, if you do not use horn sounds, you will not be free from traffic accidents when driving. The Clackson is used as a car horn and has a simple mechanical structure, which is easy to use. However, it has a disadvantage that it can not redesign the sound pressure of fluid sound. In this study, the power supply time width of Clackson was adjusted to five (0.01 s, 0.02 s, 0.03s, 0.06 s, and 0.13 s). The

result was a sound level of five (80 dB, 85 dB, 90 dB, 100 dB, and 110 dB) sound levels of the Klaxon. The experimental result shows that the maximum sound pressure (pmax = 110 dB) after operating the clackson is tmax. Ps (dB) = 110 (dB)—{10log (ton / (ton + toff)) + 20log (ton / tmax) The preference was evaluated for five types of Clackson sounds of 5 seconds. The MOS evaluation was performed on 100 participants after three times of five kinds of Klaxon sounds. The evaluation items were MOS measurement for risk, sound size, unpleasantness, and stress. As a result of evaluation, when designing various forms of clackson sound, it was quickly felt that it is danger to give a sound rhythm to a horn sound rather than a conventional sound. In particular, if you change the rhythm of the sound of Klaxon, the risk perception becomes faster even though the average sound level is lowered by −20 dB.

MONDAY AFTERNOON, 4 DECEMBER 2017 BALCONY N, 2:45 P.M. TO 4:15 P.M.

Session 1pEAb

Engineering Acoustics: General Topics Engineering Acoustics IV

Kenneth M. Walsh, Chair
K&M Engineering Ltd., 51 Bayberry Lane, Middletown, RI 02842

Contributed Papers

2:45

1pEAb1. Comparison and analysis of the methods defined by ASTM standard E2235-04, ISO 3382-2-2008, and EASE acoustical modeling software to determine reverberation time RT60 in ordinary rooms. Juan C. Montoya (none, 41 Churchill St., Springfield, MA 01108, jmontoya@csacoustics.com)

Determining the reverberation time RT60 is a well known acoustical parameter used by acousticians to characterize acoustically a room. This paper discusses the differences in the results obtained from measuring RT60 using common standards methods defined by ASTM and ISO. It also compares and contrasts those results with acoustical modeling software (EASE) for a small office room, a gymnasium, and a small church sanctuary. A discussion between the procedures utilized, which includes type and location of sound source, position and requirements of the measurement microphones, repeatability of the measurement sample, signal used for the measurement, and frequency range, are attempted in order to evaluate common ground and a relative best practice.

3:00

1pEAb2. Measurement of speed of sound profile as a function of altitude. Zhuang Li, Brett Schaefer, Brian Schaefer, William Dever, Tyler Morgan, and Matthew Foltz (Chemical, Civil, and Mech. Eng., McNeese State Univ., Box 91735, Lake Charles, LA 70609, zli@mcneese.edu)

The goal of this undergraduate research project is to design a balloon payload of less than 500 g to measure speed of sound and atmospheric data including temperature, pressure, humidity from 0 to 100,000 feet altitude. The payload is made of polystyrene foam due to its lightweight and good thermal isolation property. An ultrasonic sensor with a reflection mirror are installed outside payload box to measure speed of sound. As the ultrasonic sensor's working temperature is above −40 °C, a heater is attached to the sensor but isolated from the environment. Temperature, pressure, and humidity sensors were calibrated. Power consumption was also calculated and power is supplied by 2CR5 batteries. Software for the project was developed in Arduino DUE. An SD card shield is connected to the Arduino board for data storage and RTC clock. Tests were conducted in vacuum chamber and freezer prior to flight. Balloon launch was successfully conducted in the NASA CSBF on May 16 2017. Speed of sound was also calculated using measured environmental data. Measured and calculated speed of sound profiles were compared and reasonably agree with each other. An improved payload will launch in Carbondal, IL, on August 21, 2017, during the solar eclipse.

3:15

1pEAb3. Experimental study of nonlinear acoustic effects of orifices in duct systems. Samuel T. Kawell, Thomas Teasley, and David Scarborough (Aerosp. Eng., Auburn Univ., 211 Eng. Dr., Auburn, AL 36849, stk0011@auburn.edu)

Researchers focused on reduced emissions hydrocarbon fuel combustion systems have developed new high-efficiency furnace and gas turbine engine technologies that, unfortunately, regularly suffer from detrimental combustion instabilities. Engineering design tools developed to predict these instabilities in combustion systems frequently neglect nonlinear acoustic effects. However, boundaries and duct junctions, e.g., area changes, valves, and orifices, often exhibit nonlinear effects even at low acoustic pressure amplitudes. Experimental data of these nonlinear effects are required to accurately model engine and duct acoustics and to predict combustion instabilities. This experimental investigation focuses on understanding the nonlinear acoustic response of an orifice with and without steady flow. Various orifices were mounted in a multiple-microphone impedance tube and the acoustic impedance of the combination was used to measure the nonlinear acoustic response and impedance of the orifices at amplitudes from 114 to 190 dB and over frequencies from 100 to 1500 Hz. The preliminary results of the experimental study suggest that the impedance is dependent on the acoustic velocity amplitude. This indicates significant nonlinear effects. These experimental data will assist the development of nonlinear acoustic models of orifices in combustion systems.

3:30

1pEAb4. Extension of the phase and amplitude gradient estimator method for acoustic intensity to multiple tones. Kelli Succo, Scott D. Sommerfeldt, Kent L. Gee, and Tracianne B. Neilsen (Phys., Brigham Young Univ., N203 ESC, BYU, Provo, UT 84602, kelli.fredrickson7@gmail.com)

The phase and amplitude gradient estimator (PAGE) method [Thomas *et al.*, *J. Acoust. Soc. Am.* **137**, 3366–3376 (2015)] has proven successful in improving the accuracy of measured energy quantities over the traditional p-p method in several applications. One advantage of the PAGE method is the use of phase unwrapping, which allows for increased measurement bandwidth. However, phase unwrapping works best for broadband sources and fields with high coherence. Narrowband sources often do not have coherent phase information over a sufficient bandwidth for a phase unwrapping algorithm to unwrap properly. Even for narrowband signals, the PAGE

method has been shown to provide correct intensity measurements for frequencies up to the spatial Nyquist frequency. This is improved bandwidth over the p-p method. Previous work with sawtooth waves in a plane wave tube shows that in cases with multiple tones, the PAGE method with a few additional steps of processing accurately calculates intensity above the spatial Nyquist frequency provided one tone is below it. A variety of further experiments with multiple tones are explored to determine if any extra steps in processing or ingenuity in data acquisition can reasonably be used to achieve comparable results in a free-field environment. [Work supported by NSF.]

3:45

1pEAb5. A new lumped parameter model for the design of the free-flooded ring transducer. Kyounghun Been, Seungwon Nam (Dept. of Mech. Eng., Pohang Univ. of Sci. and Technology (POSTECH), PIRO 416, Hyoja-dong, Nam-gu, Pohang-si, Gyeongbuk 790-784, South Korea, khbeen@postech.ac.kr), Haksue Lee, Hee-seon Seo (Agency for Defense Development, Changwon, South Korea), and Wonkyu Moon (Dept. of Mech. Eng., Pohang Univ. of Sci. and Technology (POSTECH), Pohang, Kyungbuk, South Korea)

The free-flooded ring (FFR) transducer is the well-known low-frequency sound sources in underwater because its operating frequency bandwidth is broad and relatively small size. Previous researches were preformed to predict the characteristics of FFR transducers using ECM which is a type of LPM because ECM is widely used to understand its characteristics in transducer design process. However, it is hardly to predict the characteristics of an FFR transducer because the acoustic field is generated from its top and bottom openings, connected by the inner fluid, as well as the cylindrical ring surface. Here, the authors investigated an ECM of an FFR transducer consisting of three parts: the piezoelectric ring, the cylindrical cavity, and the radiation load. In addition, an LPM which can consider mutual radiation loads was proposed to improve the accuracy of the model. The proposed models were verified using commercial finite element method (COMSOL Multiphysics). It was confirmed that LPM could predict characteristics of FFR transducer more accurately than ECM. [Work supported by NRF 2016R1E1A2A02945515).]

4:00

1pEAb6. Fluid instabilities in turbocharger compression systems. Rick Dehner, Ahmet Selamet, and Emel Selamet (Mech. and Aerosp. Eng., The Ohio State Univ., 930 Kinnear Rd., Columbus, OH 43212, dehner.10@osu.edu)

Turbochargers increase the power density of internal combustion engines, which allows the displaced volume to be reduced and the fuel efficiency to be improved for the same level of performance. However, the demand for turbocharger centrifugal compressors to deliver increasingly elevated boost pressures, over a wide air flow range, results in unfavorable flow fields that generate narrow and broadband noises within the engine air induction system. The present study is a combined computational and experimental effort, focusing on identifying unsteady fluid-flow instabilities in a turbocharger compression system installed on a bench top stand. This work concentrates primarily on prediction of mild and deep surge instabilities, since they limit the low-flow operating range. Surge occurs near the Helmholtz resonance frequency of the compression system (including a compressor inlet duct, centrifugal compressor, compressor outlet duct, plenum, plenum outlet duct, and a valve) and results in low frequency pressure fluctuations throughout the compression system. Additionally, the compressor flow field adapts to operation at reduced flow rates by forming rotating instabilities in the impeller inducer and vaneless diffuser, which contribute to the occurrence of surge and produce noise as they interact with the rotating impeller and stationary volute.

MONDAY AFTERNOON, 4 DECEMBER 2017 STUDIO 4, 1:00 P.M. TO 4:00 P.M.

Session 1pMU

Musical Acoustics: Marching Band Instruments

D. Murray Campbell, Cochair
School of Physics and Astronomy, University of Edinburgh, James Clerk Maxwell Building, Mayfield Road, Edinburgh EH9 3JZ, United Kingdom

Thomas Moore, Cochair
Department of Physics, Rollins College, 1000 Holt Ave., Winter Park, FL 32789

Chair's Introduction—1:00

Invited Papers

1:05

1pMU1. Saxophone acoustics and marching bands. Vincent Gibiat (ICA, Toulouse Univ., 118 Rte. de Narbonne, Toulouse 31400, France, vincent.gibiat@univ-tlse3.fr), Jerome Selmer, and Jonathan Cottier (Henri Selmer Paris, Paris, France)

Historically saxophones appear in bands far after the beginning of Jazz history. Marching bands were mainly built around brass instruments and the saxophone appears alone in these bands. The main question that arises is "how this brass-woodwind succeeded to mix in such brass bands? The power of brass instruments is known to cover most woodwinds but the saxophone survives when flutes, with the exception of piccolo, disappeared. The answer could be in the design of saxophone and mainly in the design of tenor saxophone.

The saxophone has been built and designed by A. Sax to be a powerful instrument able to cover or to compete with the sound of brass instruments, and he has been successful. We will show that in addition with the native sound power of its side, the saxophone exhibits a formant in the most sensitive part of our hearing system that gives it such "voicy" sound. Finally, we will show that saxophones are as wild instruments as the other members of marching bands with theses two characteristics power and voice.

1:25

1pMU2. Single-reed woodwinds: From physical modeling to sound radiation. Vasileios Chatziioannou and Alex Hofmann (Dept. of Music Acoust., Univ. of Music and performing Arts Vienna, Anton-von-Webern-Platz 1, Bldg. M, Vienna 1030, Austria, chatziioannou@mdw.ac.at)

Single-reed woodwind instruments have been traditionally used in marching bands, along with brass and percussion instruments. However their sound is often covered by louder instruments, making them less audible to the audience and the members of the band. Indeed, the saxophone has been designed in order to replace the clarinet in military bands, making the woodwind section of the band more prominent. This study discusses the sound generation mechanism of single-reed woodwind instruments in an attempt to investigate which reed and mouthpiece properties significantly affect the amplitude of the radiated sound. To this end, transfer function measurements and numerical investigations using physical modeling are employed. The former may indicate how the use of different mouthpiece geometries and various types of reeds may have an effect on the sound magnitude. The latter can be used to systematically vary reed and embouchure related parameters while observing how changes in these parameters affect the radiated sound.

1:45

1pMU3. Traditional lip-blown aerophones in China. Stewart Carter (Music, Wake Forest Univ., 1833 Faculty Dr., Winston-Salem, NC 27106, carter@wfu.edu)

Bands and orchestras in present-day China universally employ Western-style valved trumpets and horns, but "natural" lip-blown instruments were known in China as early as the Han Dynasty, when *guchui* (wind and drum) ensembles played an important role in court rituals and military processions. Drawing on early artworks, treatises, and modern ethnographic studies, my paper demonstrates the enduring use of traditional lip-blown instruments in China, from early imperial times to the present day. The earliest trumpets in China probably were made from animal horns. In the Mogao Grottoes near Dunhuang, on the ancient Silk Road, a mural shows musicians playing animal horns associated with drums, celebrating the defeat of the Tibetan army by Chinese forces in 848, but the trumpets in two brick reliefs dating from approximately three centuries earlier appear to have been made of metal, probably bronze. In China, today natural lip-blown instruments endure in Buddhist ritual music, primarily in Tibet and Mongolia, as well as in certain minority cultures, where they are employed primarily in processions accompanying weddings and funerals.

2:05

1pMU4. Serpents on parade. D. Murray Campbell and Arnold Myers (Univ. of Edinburgh, James Clerk Maxwell Bldg., Mayfield Rd., Edinburgh EH9 3JZ, United Kingdom, d.m.campbell@ed.ac.uk)

The serpent which was used to accompany plainchant in French churches from the beginning of the seventeenth century was a wooden lip-excited wind instrument with a wide conical bore and six fingerholes. Toward the end of the eighteenth century, the serpent found another role as a bass instrument in civilian and military bands. The traditional form of the church serpent was modified in various different ways to make it more convenient for playing on the march. This paper compares the acoustical and musical properties of the church serpent, the horizontal English military serpent, and several types of upright serpent designed for band use.

2:25–2:45 Break

2:45

1pMU5. Timbre of marching-band brass instruments. Robert W. Pyle (P & S Horns, 11 Holworthy Pl., Cambridge, MA 02138, rpyle@icloud.com), Sabine Klaus (National Music Museum, Vermillion, SD), and Arnold Myers (Univ. of Edinburgh, Edinburgh, United Kingdom)

The tone quality of brass instruments for outdoor use has varied from time to time and from place to place. Instruments currently in use can be recorded and analyzed. However, many older instruments, typically those in museum collections, are not allowed to be played. Hence, it is necessary to rely on measurements of physical dimensions and acoustic input impedance to form comparisons. In this paper, such measurements are used to estimate two important timbral parameters: the spectral centroid at moderate playing levels and the brassiness potential. For different instruments of the same pitch, like trumpet and flugelhorn, the spectral centroid gives a measure of the brightness or darkness of the tone. The brassiness potential gives a measure of the degree to which the tone develops a "brassy" edge at high playing levels. As one might expect, the trumpet has both a higher spectral centroid and a higher brassiness potential than the flugelhorn. Results will be shown comparing instruments built in different countries and at different times.

3:05

1pMU6. Acoustics, performance, and instrument invention at Cyclophonica Bicycle Orchestra. Leonardo Fuks (Musicology and Music Education, UFRJ Rio de Janeiro Federal Univ., Rua do Passeio 98, Rio de Janeiro, Rio de Janeiro 20021-290, Brazil, fuks.leonardo@gmail.com)

Cyclophonica Bicycle Orchestra was created in 1999 in Rio de Janeiro- Brazil by the present author as a platform for experimenting new ways of music making, developing new instruments and techniques. Inspired by the traditional marching and animal-mounted bands, present all over the world, the project of Cyclophonica aimed at exploring open-air or closed ambients, involving or moving with the audience and being able to control "chamber" music performance and instruments, while ensuring safety to the players and listeners.

Several questions were still open: which were the instruments that could be played?; would musicians be able to accelerate the tempo and reduce the bicycle speed independently?; how well would the performers coordinate music without a conductor or looking to each other?; how would the listeners receive such music that would be produced and diffused in a different way? A group of six musicians was formed in the beginning. Presently, after more than eighteen years later, the Cyclophonica has ten fixed members and two additional ones, and has performed more than two hundred and fifty times, in different festivals, events, and official ceremonies, almost always being paid for, being perhaps the only professional group of this kind in the world.

3:25

1pMU7. A history of brass bugles in American drum and bugle corps. Jack Dostal (Phys. Dept., Wake Forest Univ., P.O. Box 7507, Winston-Salem, NC 27109, dostalja@wfu.edu)

Bugles have long played an important role in American bands and corps, reaching beyond their military origins. As early as the 1920s, competitive drum and bugle corps performed in national competitions sponsored by the American Legion and the Veterans of Foreign Wars. Modern drum corps continue to compete through organizations such as Drum Corps International and Drum Corps Associates. While brass bugles in these competitive drum corps began as military signaling devices, successive modifications made them capable of greater ranges of music. These traditionally valveless, key-of-G bugles evolved to include pistons and rotors, gaining notes beyond a single harmonic series. Instrument sizes were varied to permit pitch ranges from contrabass to soprano. The corresponding music in these shows thus expanded from bugle calls to classical, jazz, and popular genres. This talk will trace the evolution of bugles and their capabilities in American drum corps from their 20^{th} century history through the current day.

Contributed Paper

3:45

1pMU8. Experimental and computational investigation of the acoustics of ErgoSonic Percussion angled shell drums. Mahdi Farahikia, Ronald Miles (Mech. Eng., SUNY Binghamton, 13 Andrea Dr. Apt. A, Vestal, NY 13850, mfarahi1@binghamton.edu), and Ken Turner (ErgoSonic Percussion, Apalachin, NY)

Acoustic principles of ErgoSonic Percussion's patented angled shell drums were studied through the Finite Element Method (FEM) predictions and acoustic measurements in our anechoic chamber. Unlike existing drums, this drum design provides improved playability and the ability to easily modify the resonant decay and character of the sound. When used in marching bands, this design is found to significantly reduce strain and fatigue of the musician. Computational and experimental results are presented to examine the effects of the shape of the drum, and the size, location, and orientation of a port on the resonant head on the pitch, sound decay and overall tuning of the instruments. This angled drum design is found to provide a smaller, lighter, and more ergonomic instrument that produces sound that is essentially equivalent to that of conventional drums. It is also shown that the acoustic properties of the port can be easily adjusted during use to significantly modify the resonant character of the sound by changing the system damping. Adjustment of the decay in the sound of the drums can be easily accomplished by modifying the cover on the port using different porous materials, each of which leads to a different decay rate.

Session 1pNS

Noise and Physical Acoustics: Supersonic Jet and Rocket Noise II

Seiji Tsutsumi, Cochair
JEDI center, JAXA, 3-1-1 Yoshinodai, Chuuou, Sagamihara 252-5210, Japan

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

Chair's Introduction—1:00

Invited Papers

1:05

1pNS1. Observations regarding the noise radiated from full-scale heated, supersonic jets. Kent L. Gee (Dept. of Phys. and Astronomy, Brigham Young Univ., N243 ESC, Provo, UT 84602, kentgee@byu.edu)

In this presentation, the characteristics of supersonic jet noise are reviewed. Observations from more than a decade's worth of measurements of high-performance jet engines and large solid rocket motors are compared against each other and laboratory-scale findings. These include apparent source location and extent, directivity, spectral shape, relative importance of different noise components, and presence of nonlinear propagation effects.

1:25

1pNS2. Spatiotemporal analysis of high-performance military aircraft noise during ground run-up. S. Hales Swift, Kent L. Gee, Tracianne B. Neilsen (Phys. and Astronomy, Brigham Young Univ., N221 ESC, Provo, UT 84602, hales.swift@gmail.com), Alan T. Wall (Battlespace Acoust. Branch, Wright-Patterson Air Force Base, Air Force Res. Lab., Wright-Patterson AFB, OH), Micah Downing, and Michael M. James (Blue Ridge Res. and Consulting, LLC, Asheville, NC)

Recent measurements of high-performance military aircraft noise have revealed that full-scale jet noise has features and structures that are still only partly understood, such as the presence of multiple acoustic radiation lobes in the aft direction at certain frequencies. Spatiotemporal analyses of a ground-based microphone array measurement of the noise from a tethered F-35 at various engine conditions are used to investigate these features of the sound field. The ground array covered an angular aperture of 35–152 degrees relative to the front of the aircraft. The large angular aperture allows for a detailed investigation of the correlation and coherence at frequencies exhibiting multi-lobe behavior. This spatiotemporal analysis yields further evidence of the characteristics of multi-lobe behavior in high-performance, full-scale jet noise. [Work supported by an Office of Naval Research grant, a USAFRL SBIR, and the F-35 JPO. Distribution A: Approved for public release; distribution unlimited. Cleared 07/10/2017; JSF17-714.]

1:45

1pNS3. Noise sources in a commercial supersonic jet. Christopher J. Ruscher and Sivaram Gogineni (Spectral Enegies, LLC, 5100 Springfield St., Ste. 301, Dayton, OH 45431, cjrusche@gmail.com)

Stringent noise regulations currently limit commercial aviation. These regulations make supersonic commercial flight impractical. The development of an engine that can meet these strict rules is paramount to making supersonic commercial flight a reality. One method of noise reduction is to add additional streams to an engine. As such, the three-stream jet has potential to help reduce exhaust noise. Understanding the noise sources in the jet plume can help to design nozzles that are quieter. To accomplish this, high-fidelity, high-speed data are required. Data for an axisymmetric and offset three-stream nozzle were generated using the LES code JENRE developed by the Naval Research Laboratory. The simulation data has been shown to match well with experimental data. Advanced analyses methods that are based on Proper Orthogonal Decomposition (POD), wavelet decomposition, and Stochastic estimation have been applied to extract noise sources in the jet plume.

2:05

1pNS4. High-performance aircraft short-takeoff and vertical-landing noise measurements on an aircraft carrier. Alan T. Wall, Richard L. McKinley (Battlespace Acoust. Branch, Air Force Res. Lab., Bldg. 441, Wright-Patterson AFB, OH 45433, alantwall@gmail.com), Allan C. Aubert, Russell W. Powers, Michael J. Smith, Charles J. Stouffer (Naval Air Systems Command, Naval Air Station Patuxent River, MD), and James C. Ku (Naval Air Systems Command, Navail Air Station Patuxent River, MD)

The noise levels caused by high-performance aircraft are relatively high in the close proximity experienced by crew on board aircraft carriers, which can interfere with communications and may pose a risk for hearing loss. This paper reports on preliminary results of noise measurements of the operations of F-35B aircraft performing short-takeoff and vertical-landing (STOVL) operations on the flight deck of an LHA aircraft carrier. This noise measurement campaign was performed in late 2016, by scientists from the Air Force Research Laboratory (AFRL) in collaboration with the Naval Air Systems Command (NAVAIR) and the F-35 Integrated Task Force (ITF). The measurements were taken using hand-held noise recorder systems, and the recording engineers shadowed actual locations of crew. These data will be used to validate STOVL models of crew noise exposures on deck. [Work supported by F-35 JPO.]

2:25

1pNS5. Characterization of broadband shock-associated noise from high-performance military aircraft. Tracianne B. Neilsen, Aaron Vaughn, Kent L. Gee (Brigham Young Univ., N311 ESC, Provo, UT 84602, tbn@byu.edu), Alan T. Wall (Air Force Res. Lab., Wright-Patterson AFB, OH), Micah Downing, and Michael M. James (Blue Ridge Res. and Consulting, LLC, Asheville, NC)

For nonideally expanded jets, broadband shock-associated noise (BBSAN) is a feature in the sideline and forward directions. While BBSAN has been studied fairly extensively for laboratory-scale jets, its presence and characteristics in full-scale, tactical aircraft noise need to be evaluated. Noise measurements on a tied-down F-35 provide the opportunity to characterize full-scale BBSAN using a linear ground array that spanned a large angular aperture: 35–152 degrees relative to the front of the aircraft. The main questions are whether the full-scale BBSAN shares the same characteristics as those observed in laboratory-scale BBSAN and if current models capture the features of full-scale BBSAN. The variation in the spectral shape, peak frequency, and peak level of full-scale BBSAN across angle for different engine powers is explored and compared to prior laboratory studies. Comparisons are also made with models for BBSAN based on stochastic theory ([Tam *et al.*, *J. Sound Vib.* **140**, 55–71 (1990)]) and the simplified model used in Kuo *et al.* [AIAA Paper 2011–1032 (2011)] for lab-scale BBSAN. Frequency-dependent convective speed estimates obtained from the current BBSAN models are compared to estimates based on directivity. [Work supported by the Office of Naval Research and the F-35 JPO.]

2:45–3:00 Break

Contributed Papers

3:00

1pNS6. Spectral decomposition of turbulent mixing and broadband shock-associated noise from a high-performance military aircraft. Aaron Vaughn, Tracianne B. Neilsen, Kent L. Gee (Brigham Young Univ., C110 ESC, Provo, UT 84602, aaron.burton.vaughn@gmail.com), Alan T. Wall (Air Force Res. Lab., Wright-Patterson AFB, OH), Micah Downing, and Michael M. James (Blue Ridge Res. and Consulting, Asheville, NC)

Sound from high-performance military aircraft originates primarily from the turbulent mixing noise, but at smaller inlet angles, broadband shock-associated noise (BBSAN) is present. The similarity spectra of the two components of turbulent mixing noise developed by Tam *et al.* [AIAA Paper 96–1716 (1996)] represent noise associated with fine and large-scale turbulent structures and provide reasonable fits for ideally expanded, supersonic jet noise. For non-ideally expanded jet flow, BBSAN contributions to the spectral shape need to be included in spectral decompositions in the sideline and forward directions. A model proposed by Tam *et al.* [*J. Sound Vib.* **140**, 55–71 (1990)] and later simplified by Kuo *et al.* [AIAA Paper 2011–1032 (2011)] provides a spectral function that models the BBSAN spectral shape. The ability of the BBSAN and similarity spectra shapes to account for the measured spectra is evaluated for ground-based microphones that covered a spatial aperture from 35 to 152 degrees. Spectral decompositions at low and high engine powers are compared. Using turbulent mixing noise similarity spectra decomposition in conjunction with BBSAN empirical fits, a better equivalent source model can be developed. [Work supported by the Office of Naval Research and the F-35 JPO. Distribution A: Approved for public release; distribution unlimited.]

3:15

1pNS7. Modeling shock formation and propagation in high-performance jet aircraft noise. Brent O. Reichman, Kent L. Gee, Tracianne B. Neilsen (Brigham Young Univ., 453 E 1980 N, #B, Provo, UT 84604, brent.reichman@byu.edu), Alan T. Wall (Battlespace Acoust., Air Force Res. Lab., Wright-Patterson AFB, OH), Micah Downing, and Michael M. James (Blue Ridge Res. and Consulting, LLC, Asheville, NC)

Nonlinear propagation can play an important role in both time and frequency-domain features of far-field supersonic jet noise. Many aspects of nonlinear propagation, such as waveform steepening and greater-than-expected high-frequency spectral levels, have been previously predicted for select angles and engine conditions. This paper builds on previous successes and presents a comparison of nonlinear and linear predictions for the F-35B aircraft. Results are shown over a wide spatial and angular range and over varying engine power conditions, including showing evidence of nonlinear propagation in the forward direction at the highest engine conditions. In addition, specific features, such as individual shocks, are compared between numerically propagated and measured waveforms, highlighting the successes and deficiencies of current propagation models. Weather and multipath interference effects are also addressed and corrected using an empirical model. [Work supported by USAFRL through ORISE. Distribution A: Approved for public release; distribution unlimited; Cleared 07/10/2017; JSF17-714.]

3:30

1pNS8. Measurement methods for high-performance jet aircraft noise inside a hardened aircraft shelter. Richard L. McKinley, Alan T. Wall (Battlespace Acoust., Air Force Res. Lab., 2610 Seventh St., AFRL/711HPW/RHCB, Wright-Patterson AFB, OH 45433-7901, rich3audio@aol.com), Theo A. van Veen (NLR: Nederlands Lucht- en Ruimtevaartcentrum – Netherlands Aerosp. Ctr., The Hague, Netherlands), and Jaap van't Hof (TNO: Nederlandse organisatie voor Toegepast Natuurwetenschappelijk Onderzoek – The Netherlands Organisation for Appl. Sci. Res., The Hague, Netherlands)

High-performance military aircraft regularly operate inside hardened aircraft shelters (HAS). The F-35A aircraft must be certified as safe to operate inside a HAS before it can be deployed and used in such structures worldwide. Acoustic levels at maintainer locations allow for noise dose estimates and regulation of personnel mission support in HASs to mitigate risks of hearing damage. Acoustic levels impinging on the airframe are compared against engineering design limits in order to prevent a reduction in operational lifespan of the aircraft due to acoustic fatigue. The Air Force Research Laboratory, the Royal Netherlands Air Force, and the Dutch national laboratories NLR and TNO collaborated on a set of acoustic measurements for an F-35A operating inside a HAS at Leeuwarden airbase in the Netherlands. The methods, analysis, and qualitative findings of the acoustic measurements are presented here. [Work supported by RNLAF and by the F-35 JPO. Distribution A: Approved for public release; distribution unlimited. Cleared 01/24/2017; JSF17-035.]

3:45

1pNS9. Calculating the frequency-dependent apparent source location using peak cross-correlation between near-field and far-field microphone arrays. Jacob A. Ward, S. Hales Swift, Kent L. Gee, Tracianne B. Neilsen (Phys., Brigham Young Univ., N243 ESC, Provo, UT 84602, jacob.ward@live.com), Koji Okamoto, and Masahito Akamine (Dept. of Adv. Energy, Graduate School of Frontier Sci., The Univ. of Tokyo, Kashiwa, Chiba, Japan)

The apparent acoustic source region of jet noise varies as a function of frequency. In this study, the variation of the apparent maximum source location with frequency is considered for an ideally expanded, unheated, Mach-1.8 jet with exit diameter of 20 mm and a Reynolds number of 6.58e6. In this study, the source location is ascertained for one-third octave bands by evaluating peak cross-correlation between near-field linear microphone arrays at three sideline distances and a far-field microphone arc. The impact of the hydrodynamic field on correlation results is considered. Source locations determined by these means are compared with intensity analyses for the same jet [K. L. Gee *et al.*, AIAA Paper 2017–3519 (2017)]. Correlational methods together with filtering can provide a straightforward measure of the acoustic origin as a function of frequency and thus inform optimal microphone array layout for specific frequency regimes.

4:00

1pNS10. Numerical validation of using multisource statistically optimized near-field acoustical holography in the vicinity of a high performance military aircraft. Kevin M. Leete (Brigham Young Univ., Provo, UT 84604, kevinmatthewleete@gmail.com), Alan T. Wall (Battlespace Acoust. Branch, Air Force Res. Lab., Wright-Patterson AFB, OH), Kent L. Gee, Tracianne B. Neilsen (Brigham Young Univ., Provo, UT), Micah Downing, and Michael M. James (Blue Ridge Res. and Consulting, LLC, Asheville, NC)

Multisource statistically optimized nearfield acoustical holography (M-SONAH) is an advanced holography technique [Wall *et al.*, *J. Acoust. Soc. Am.* **137**, 963–975 (2015)] that has been used to reconstruct the acoustic field from measurements taken in the vicinity of a high-performance military aircraft [Wall *et al.* **139**, 1938 (2016)]. The implementation of M-SONAH for tactical jet noise relies on creating an equivalent wave model using two cylindrical sources, one along the jet centerline and one below the ground as an image source, to represent the field surrounding an aircraft tethered to a reflecting ground run up pad. In this study, the spatial and frequency limitations of using the M-SONAH method to describe the field of a tethered F-35 is explored by using the same measurement geometry as at a recent test, but substituting the sound field obtained from a numerical source for the measurement data. The M-SONAH reconstructions are then compared to numerical benchmarks. A spatial region and frequency bandwidth where bias errors are low are identified and provide validation for the use of this method in tactical jet noise source and field reconstructions. [Work supported by USAFRL through ORISE and the F-35 JPO. Distribution A: Approved for public release; distribution unlimited. Cleared 07/10/2017; JSF17-714.]

4:15

1pNS11. Sound quality analysis of far-field noise from a high-performance military aircraft. S. Hales Swift, Kent L. Gee, Tracianne B. Neilsen (Phys. and Astronomy, Brigham Young Univ., N221 ESC, Provo, UT 84602, hales.swift@gmail.com), Alan T. Wall (Battlespace Acoust. Branch, Wright-Patterson Air Force Base, Air Force Res. Lab., Wright-Patterson AFB, OH), Micah Downing, and Michael M. James (Blue Ridge Res. and Consulting, LLC, Asheville, NC)

Noise from high-performance military aircraft can pose challenges to community relations near airfields. Accurately predicting and quantifying community impacts is important for efforts to minimize such impacts and reduce annoyance. In this study, sound recordings measured 305 m from a tethered F-35 aircraft operating at various engine conditions are analyzed using sound quality metrics. The calculated metrics are inputs for a model of perceived annoyance used to estimate the relative contributions of loudness and other sound quality features to annoyance. These results can help inform future efforts at noise reduction by identifying potentially relevant sound quality components of the jet noise as well as helping inform discussions of noise policy on military bases. [Work supported by a USAFRL SBIR and the F-35 JPO. Distribution A: Approved for public release; distribution unlimited. Cleared 07/10/2017; JSF17-714.]

4:30

1pNS12. Spatial interpolation of noise monitor levels. Edward T. Nykaza (ERDC-CERL, 2902 Newmark Dr., Champaign, IL 61822, edward.t.nykaza@usace.army.mil)

Continuously recording noise monitoring stations provide feedback of the noise environment at monitor locations. While this feedback is useful, it only provides information at a few point locations, and in many cases it is of interest to know the noise level(s) at the locations between and beyond noise monitoring locations. In this study, we test the accuracy of several spatial interpolation models with experimental data collected during the Strategic Environmental Research and Development Program (SERDP) Community Attitudes Towards Military Blast Noise study. These datasets include 9 months of blast noise events captured at two different study locations. In both cases, a small number of monitors (e.g., 3–9) were located over a large region of interest (e.g., 1–8 km^2), thus providing realistic operational conditions. The utility of deterministic (e.g., nearest neighbor, Delaunay triangulation, thin plate splines, etc.) and stochastic (e.g., geostistical or kriging) interpolation models for estimating single-event and cumulative noise levels is examined using leave-one-out cross validation. The accuracy of each approach is assessed with the root-mean-square-error (RMSE), and we discuss the practical implications of implementing such approaches in real-time systems.

4:45

1pNS13. Acoustic excitation impact on aerodynamic drag measured in aeroacoustic liners. Christopher Jasinski (Univ. of Notre Dame, 54162 Ironwood Rd., South Bend, IN 46635, chrismjasinski@gmail.com) and Thomas Corke (Univ. of Notre Dame, Notre Dame, IN)

Research interest has steadily grown for understanding the aerodynamic drag produced by acoustic liners for commercial turbofan engines. This is driven by an aim to understand the phenomena fundamentally as well as for application in flight. Stringent government regulations on aircraft noise and next generation aircraft designs that may include liners on more surfaces are key drivers for industry involvement. While the conventional perforate-over-honeycomb liner has proven effective acoustically for decades, liner drag production has not been fully understood. When an acoustic liner sample is excited with sound pressure levels above 140dB re: 20 micropascals, a

measurable drag increase is observed at flight velocity. Recent measurements have shown that tonal noise at the same level can produce more than a 50 percent increase in drag coefficient for a liner sample at lower test speeds. By testing liner samples at low speed in the Notre Dame Hessert Laboratory, detailed hotwire probe measurements near the wall have been made and drag coefficient comparisons have been made with the use of a linear air-bearing force balance. The development of the measurement setup, the results produced, and a discussion of implications will be included in this paper.

MONDAY AFTERNOON, 4 DECEMBER 2017 BALCONY I/J/K, 1:25 P.M. TO 4:00 P.M.

Session 1pPA

Physical Acoustics, Biomedical Acoustics, and Engineering Acoustics: 30th Anniversary of the National Center for Physical Acoustics

Richard Raspet, Cochair
Univ. of Mississippi, University, MS 38677

Craig Hickey, Cochair
National Center for Physical Acoustics, University of Mississippi, 145 Hill Drive, P.O. Box 1848, University, MS 38677-1848

Josh R. Gladden, Cochair
Physics & NCPA, University of Mississippi, 108 Lewis Hall, University, MS 38677

Chair's Introduction—1:25

Invited Papers

1:30

1pPA1. Reflections on the origins of the National Center for Physical Acoustics. Lawrence A. Crum (APL, Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, lacuw@uw.edu)

The National Center for Physical Acoustics arose from a group of faculty within the Physics Department at the University of Mississippi that had a strong interest in the general area of physical acoustics. With Hank Bass's leadership, and support of the University of Mississippi's central administration, this group constantly sought to expand their interest and their expertise in this area. Financial support from ONR permitted the expansion of their research space and the establishment of the Institute for Technology Development in the State of Mississippi led to eventual support from Congress. In October of 1986, by an act of the Congress of the United States of America, the National Center for Physical Acoustics was established. This presentation will provide some of the history concerning this event.

1:50

1pPA2. Before there was a National Center for Physical Acoustics: Memories of the first employee. Kenneth E. Gilbert (National Ctr. for Physical Acoust., Univ. of MS, P.O. Box 35, 1703 Hunter Rd., Thaxton, MS 38871, kgilbert@olemiss.edu)

New laboratories, like newborn babies, usually come into the world along with a mix of excitement, joy, anxiety, and pain. Most of these feelings were present in the Ole Miss physics department when I arrived on the campus in August of 1985. Over the next few years, I watched what started as an attempt to obtain a metal building for acoustics research, evolve into what is today the National Center for Physical Acoustics (NCPA). In this presentation, I will tell some stories about the birth and early days of the NCPA as seen by a newly arrived outsider. Some of the stories are funny and others are not. I hope to convey what it was like to know something important was happening even though no one, especially me, seemed to know exactly where things were headed.

2:10

1pPA3. From underwater thunder to cavitation from pulsed ultrasound: One graduate student's perspective of the years leading up to the formation of NCPA. Anthony A. Atchley (Graduate Program in Acoust., Penn State Univ., College of Eng., 101C Hammond Bldg., University Park, PA 16802, atchley@psu.edu)

The Physical Acoustics Research Group at the University of Mississippi (PARGUM) fostered a vibrant, multidisciplinary research environment in the early 1980s that incubated the concept of the National Center for Physical Acoustics (NCPA). It also provided graduate students, new to physical acoustics, an enticing glimpse into the breadth of the field and an exposure to a range of important contemporary problems. The research programs included terrestrial lightning detection and location, the underwater sound generated by lightning striking the ocean surface, long range sound propagation, propagation through porous materials, sound amplification by selective molecular excitation, acoustic levitation of bubbles and droplets, studies of acoustic cavitation thresholds and sonoluminescence, numerical simulation of bubble dynamics, and others. Beyond the scientific activities, PARGUM introduced a number of new acousticians to the ASA, many of whom have contributed to the leadership of the Society. The purpose of this presentation is to review some of the research results produced by the members of group from the time period as well as their contributions to the ASA.

2:30

1pPA4. Postdoc at NCPA: Explosive growth and entertainment. W. P. Arnott (Dept. of Phys., Univ. of Nevada Reno, Reno, NV 89557, arnottw@unr.edu)

My postdoctorate from August 1988 to December 1991 at the University of Mississippi and NCPA favorably influenced my career. Thinking back on the cast of characters and the dramatic science being pursued there brings a smile. My main mentors were Dr. James Sabatier, Dr. Richard Raspet, and Dr. Henry Bass. I was involved with theory and experiments for acoustic to seismic coupling; land mine detection; nondestructive measurements of soil properties; and fundamental analysis of Celcor extruded ceramic catalysts as materials for thermoacoustic stacks. We often worked in real-world settings—cotton and soybean fields, and at “Audi Acres” where we used massive low frequency speaker elevated with construction scaffolding and a winch as sound sources. I worked hours with Mike the machinist constructing acoustical resonators and thermoacoustic heat engines. The vibrant social and political scene was something to behold—including early morning jogging in the Faulkner woods, regular basketball games at lunch; visiting politicians and program managers, dancing on tables at the Gin; Friday nights at Dr. Bass’ house; visitors from acoustic institutions around the world; Starnes catfish restaurant.

2:50–3:05 Break

3:05

1pPA5. National Center for Physical Acoustics Aeroacoustics Group. Lawrence Ukeiley (Mech. and Aerosp. Eng., Univ. of Florida, MAE-A Rm. 312, PO Box 116250, Gainesville, FL 32611, ukeiley@ufl.edu), Nathan E. Murray (NCPA, Univ. of MS, University, MS), and Bernard Jansen (NCPA, Univ. of MS, Oxford, MS)

The Aeroacoustics Group at the National Center for Physical Acoustics (NCPA) was formed in 1999 to open a new avenue of research at the center. Since its inception, the group has strived to study fundamental problems of flow-generated pressure fluctuations (both acoustic and hydrodynamic) and transition them to applied problems predominantly in the aerospace field. Over the years these problems have included vibrations in bodies traveling at supersonic speeds, jet noise associated with supersonic and subsonic exit conditions, and flow over open cavities to name a few. As part of these efforts, the group has added substantial unique infrastructure to the NCPA expanded the capabilities of the research done in house. The presentation will highlight key features of the problems, which have been studied by the group along with the key personnel.

3:25

1pPA6. Current research thrusts at NCPA. Josh R. Gladden (Phys. & NCPA, Univ. of MS, 108 Lewis Hall, University, MS 38677, jgladden@olemiss.edu)

Over the 30 year history of the National Center for Physical Acoustics, the technologies have advanced and research has evolved; however, the mission has remained the same: to pursue solutions to practical problems using a fundamental physics approach, to educate the next generation of physical acousticians, and to serve as a translational bridge between basic research generated by our academic communities and the private sector. In this talk, I will discuss major research thrusts and accomplishments of NCPA in the last 5–8 years. Topical areas covered will be infrasound and long range propagation, aeroacoustics, structural health monitoring, physical ultrasound, and acoustics in porous media. I’ll include some discussion of prominent scientists driving this research along with a few funny stories.

Contributed Paper

3:45

1pPA7. The importance of tuning curves and two-tone tests in nonlinear acoustic landmine detection. Miahnna Nguyen and Murray S. Korman (Phys. Dept., U.S. Naval Acad., 572 C Holloway Rd., Annapolis, MD 21402, m204578@usna.edu)

During the 1990s through ∼2010, Jim Sabatier’s group (Univ. Mississippi and NCPA) had made significant progress in understanding airborne acoustic landmine detection. In particular, using their development of scanning laser Doppler vibrometry, plastic anti-tank landmines could be detected in soil along roadbeds. Research on the nonlinear acoustic detection of buried landmines by Dimitry Donskoy (Stevens Tech) lead to lumped element models of the target which included a nonlinear spring-like mechanism. Tom Muir studied seismic excitations. Sabatier and Korman suggested that the soil nonlinearity and the soil interaction with the flexural vibration of the top-plate of the landmine or drum-like simulant were both significant factors in nonlinear landmine detection. Nonlinear tuning curves of the soil vibration response vs. frequency (directly over the target) along with rich combination frequency generation in two-tone tests suggested that the vibration was similar to Paul Johnson’s group results on mesoscopic nonlinear elasticity found in geomaterials. Recent airborne acoustic nonlinear landmine detection experiments compare drum-like simulant tuning curves to results using a soil plate oscillator apparatus (two circular flanges sandwiching and clamping a thin circular elastic plate that supports a cylindrical level column of sand above the plate) to help model flexural mesoscopic nonlinear hysteretic behavior.

Session 1pSA

Structural Acoustics and Vibration: General Topics in Structural Acoustics and Vibration

Benjamin Shafer, Chair
Technical Services, PABCO Gypsum, 3905 N 10th St., Tacoma, WA 98406

Contributed Papers

3:30

1pSA1. Generalized control metric for active structural acoustic control: A novel understanding for controlling symmetric structures. Yin Cao and Hongling Sun (Key Lab. of Noise and Vib. Res., Inst. of Acoust., Chinese Acad. of Sci., Qing Nian Hui Jia Yuan Bldg. 6, No. 411, Beijing 100022, China, caoyfive@gmail.com)

Active structural acoustic control (ASAC) is a strategy aiming at controlling the radiated sound of the structure by controlling the vibration. Generally, controlling the vibration is not the same as controlling the radiated sound. Therefore, the choice of the control metric also referred to as the cost function in ASAC is a very important research topic and has been received a lot of attentions by researchers. Previous research has identified a novel control metric named the weighted sum of spatial gradients (WSSG). An analysis has shown that WSSG is able to closely approximates the performance of using radiated sound power as the control objective by choosing the right weights. In this paper, based on previous findings, a generalized control metric of active structural acoustic control for symmetric structures is discussed and analyzed. For symmetric structures, the radiated resistance matrix is always bisymmetric. It will be shown that controlling the vibration is able to achieve the performance of directly controlling the radiated sound power provided that the secondary plant matrix is overdetermined. A uniformed control metric for active structural acoustic control of symmetric structures will be presented and analyzed.

3:45

1pSA2. Experimental study of the impact noise through concrete floor with resilient layers. Abdelouahab Bouttout (DPBE, CNERIB, National Ctr. of Studies and Integrated Res. of Bldg. Eng., CNERIB,Cité Nouvelle El-Mokrani, Souidania, Algeria, Souidania 16097, Algeria, Bouttout@gmail.com)

In this paper, we present an experimental approach to evaluate the sound transmission through floors and building structures. This method is developed in our laboratory at the National Center of Studies and Integrated Research of Building Engineering, CNERIB, Algeria. The method is based on measurement of sound due to impact vibration generated by a standard tapping machine placed on the upper surface of a small concrete floor. This floor has dimensions of 1.2(m)×0.8(m)×0.2(m) and simply supported on four elastic supports with 2 cm of rubber layer. The natural frequency of the floor is measured with an ambient vibration apparatus, and the obtained value is $f = 19$ Hz. The measurement consists of recording vibration levels in the upper and lower surfaces of the floor, in 1/3 octave frequency bands between 16 Hz and 16 KHz using piezoelectric accelerometer Brüel and Kjær 4508. The sound level meter Brüel and Kjær 2270 has been used to analyze the vibration signal for different cases. The weighted sound reduction index of the floor has 4 dB without any cover. Some samples of cover floor are tested using this procedure. The results show that the weighted sound reduction index of resilient layer can reach 18 dB, 11.6 dB and 6.6 dB for the rubber, Bitumen and PVC material, respectively. The important results obtained in this paper can be used as a platform to correct the impact sound insulation in multi-storey residential building and renovation plans using recycled resilient materials.

4:00

1pSA3. Prediction of ground-borne vibrations due to elevated railway traffic. Salih Alan and Mehmet Caliskan (Middle East Tech. Univ., Ankara 06800, Turkey, caliskan@metu.edu.tr)

This study investigates the impact of ground-borne vibrations on people in dwellings nearby elevated railway traffic. Assessments are conducted over predicted vibrations with respect to national regulations. Main excitation mechanism is taken as the dynamic loading due to rail and wheel irregularities. Frequency response functions at the rail head are obtained from harmonic analyses by finite element modeling of the elevated structure. In the prediction procedure, dynamic model of the vehicle is coupled with these frequency response functions. Dynamic loading on the railhead is determined and dynamic reaction forces on the legs of the elevated structure are calculated in frequency domain. Ground vibrations are then estimated by implementing a Fourier transform based theoretical model for the layered ground. Assessment in the frequency range from 1 Hz to 100 Hz in one-third octave bands are conducted for groundborne vibrations calculated at the foundation level of dwellings.

4:15

1pSA4. Vibrometric characterization of a TN-32 dry storage cask for spent nuclear fuel. Kevin Y. Lin (Phys. and Astronomy, Univ. of MS, 145 Hill Dr., PO BOX 1848, Oxford, MS 38677-1848, klin@go.olemiss.edu), Wayne E. Prather, Zhiqu Lu, Joel Mobley, and Josh R. Gladden (National Ctr. for Physical Acoust., Dept. of Phys. and Astronomy, Univ. of MS, University, MS)

The assessment of the internal structural integrity of dry storage casks for used high burnup nuclear fuel assemblies is of critical importance before these are transported to permanent repositories. The size of the casks (5.2 m in height and 2.4 m in diameter), structural complexity, and the inability to access the interior make this a challenging task. This project addresses these difficulties through a multi-modal approach involving nuclear, charged particle, and acoustic methods. In this talk, we report on linear and nonlinear vibrational spectra of intact TN-32 casks. These studies use both impulsive and swept continuous-wave excitations with a variety of sensor placement configurations. From the resulting spectra, resonant frequencies, quality factors, and harmonic responses of various vibrational modes were determined. A detailed finite element model of the TN-32 was constructed and the experimental results are compared to the modal structure determined numerically. [This work was supported by DOE NEUP Award: DENE0008400.]

4:30

1pSA5. The determination of mode shapes of a scaled model TN-32 spent nuclear fuel dry storage cask. Kevin Y. Lin (Phys. and Astronomy, Univ. of MS, 145 Hill Dr., PO Box 1848, Oxford, MS 38677-1848, klin@go.olemiss.edu), Wayne E. Prather, Zhiqu Lu, Joel Mobley, and Josh R. Gladden (National Ctr. for Physical Acoust., Dept. of Phys. and Astronomy, Univ. of MS, University, MS)

The assessment of the internal structural integrity of dry storage casks for used high burnup nuclear fuel assemblies is of critical importance before

these are transported to permanent repositories. The size of the casks (5.2 m in height and 2.4 m in diameter), structural complexity and the inability to access the interior make this a challenging task. This project addresses these difficulties through a multi-modal approach involving nuclear, charged particle and acoustic methods. In this work, we report on measurements of the vibrational spectra and mode shapes using a 6:1 scaled TN-32 model cask constructed in our lab. 2-D vibrometric scans were performed with a laser Doppler velocimeter to measure the vibrational mode shapes exhibited by the cask. Good agreement is observed between the experimentally measured mode shapes and those determined numerically using a detailed finite element model. Several modes are identified that will be important in assessing the internal characteristics from external measurements. [This work was supported by DOE NEUP Award: DENE0008400.]

MONDAY AFTERNOON, 4 DECEMBER 2017 ACADIA, 2:00 P.M. TO 5:00 P.M.

Session 1pSC

Speech Communication: Voice, Tone, and Intonation (Poster Session)

Cynthia P. Blanco, Chair
Department of Psychology, Northwestern University, 2029 Sheridan Road, Evanston, IL 60208

All posters will be on display from 2:00 p.m. to 5:00 p.m. To allow authors an opportunity to view other posters in their session, authors of odd-numbered papers will be at their posters from 2:00 p.m. to 3:30 p.m. and authors of even-numbered papers will be at their posters from 3:30 p.m. to 5:00 p.m.

Contributed Papers

1pSC1. Differential pitch analysis for tone and accent languages. Ettien N. Koffi (English, Saint Cloud State Univ., 720 Fourth Ave. South, Saint Cloud, MN 56301, enkoffi@stcloudstate.edu)

According to the Critical Band Theory, the auditory perception of F0 data is the same for all human beings. However, when F0 signals are transferred through the auditory cortex to specialized areas of the brain, they are perceived and processed differently depending on whether the language is tonal or accentual. In tone languages, F0 data appears to be processed in Heschl's gyrus (Schneider 2005), whereas in accent languages, it is processed via the thalamus to inferior frontal regions (Myers 2017). Furthermore, in accent languages, F0 signals are computed on a nominal scale, but in tone languages, an ordinal scale is used (Speaks 2005). These insights support the long-held linguistic view that accent and tone languages are prosodically different. Terms such as strong/weak, stressed/unstressed are used to describe pitch variations in accent languages, whereas in tone languages, the terms used are extra low, low, mid, high, and extra high. Due to these prosodic and processing differences, it is not advisable to apply the same interpretive framework in analyzing pitch variations in tone and accent languages. Examples will be provided from English and Anyi, a West African language, to underscore the pitfalls in doing so.

2:00

1pSC2. What happens when f0 movements and prosodic units were randomly aligned? Wei Lai, Nari Rhee, and Mark Liberman (Dept. of Linguist, Univ. of Pennsylvania, 3401-C Walnut St., Ste. 300, C Wing, Philadelphia, PA 19104, weilai@sas.upenn.edu)

According to intonation phonology, surface f0 is generated from superimposition of local f0 events associated with hierarchical prosodic units (e.g., Pierrehumbert, 1980). This study conducted two experiments to investigate whether or to what extent the misalignment between f0 patterns and prosodic units would result in awkward-sounding English intonation. In Experiment 1, f0 contours of utterances from Obama Weekly Address were extracted via Praat, interpolated and smoothed using a quadratic spine, and superimposed to another Obama-Weekly-Address stretch by PSOLA algorithm. The perception result, surprisingly, showed that the resynthesized sentence still sounded quite natural in intonation, regardless of the random alignment between f0 patterns and the location of stresses and phrases. In Experiment 2, the Obama-Weekly-Address stretches were superimposed with f0 contours of three different languages that differ in the size of basic prosodic domains: Mandarin (a tone language), Korean (an accent phrase language) and Ukrainian (a stress language). The perception results showed that utterances superimposed with Korean and Ukrainian f0 contours still sounded quite natural and native, while Mandarin f0 contours sounded somehow fluctuational on English segments. These results suggested that f0 movements might not be largely determined by prosody structures, but by language-independent constrains such as physiological plausibility and expediency.

1pSC3. Native perception of Cantonese tones in high-variability conditions. Yan Chen (Linguist Dept., Univ. of Arizona, TUCSON, AZ 85721, yanchen@email.arizona.edu)

This study investigates how native speakers of Cantonese perceive and produce Cantonese tone pairs T2-T5, T3-T6, T4-T5, T4-T6, and T5-T6 in high-variability conditions. Thirty five native speakers of Cantonese from Hong Kong participated in a 6AFC identification task (screening test), a mixed-talker AXB task (ISI = 1500 ms), and a mixed-talker repetition task (delay = 1500 ms). Results from the screening test showed that the participants did not merge the tones examined in this study. Results from the AXB task showed that (1) perception of tones with f0-height differences (T2-T5 and T3-T6) was more challenging than perception of tones with f0-contour differences (T4-T5, T4-T6, and T5-T6), (2) tones contrasting in direction of f0 change (T4-T5) were the easiest, and (3) perception of level tones (T3-T6) was more difficult than perception of contour tones with an f0-height difference (T2-T5). Data from the repetition task were fitted with quadratic polynomial equations ($y = a + bx + cx^2$) and the analysis revealed that the participants produced the two level tones—T3 and T6—distinctively (in terms of coefficient-a) and the two rising tones—T2 and T5—distinctively (in terms of coefficient-c). This indicates that the participants perceived different level tones and different rising tones in the repetition task (i.e., no tone merger).

1pSC4. English focus prosody processing and production by Mandarin speakers. Chikako Takahashi, Hyunah Baek, Sophia Kao, Alex. H. L. Yeung, Marie K. Huffman, Ellen Broselow (Dept. of Linguist, Stony Brook Univ., SUNY at Stony Brook, Stony Brook, NY 11794-4376, chikako.takahashi@stonybrook.edu), and Jiwon Hwang (Asian and Asian American Studies, Linguist, Stony Brook, NY)

Our study compared the processing and production of English focus prosody by native speakers of English and Mandarin. Twenty-one Mandarin speakers living in the US and 21 English speakers participated in two tasks. In the processing task, participants responded to instructions that contained natural or unnatural contrastive prosody (Click on the purple sweater; Now click on the SCARLET sweater/Now click on the PURPLE jacket.) In the production task, participants guided an experimenter to place colored objects on a white board, with some contexts designed to elicit contrastive focus (Put the yellow arrow over the ORANGE arrow/yellow DIAMOND, please). All adjectives and nouns were bisyllabic trochees. The two groups differed in their realization of focus, with English speakers tending to align the pitch peak with the stressed syllable and Mandarin speakers with the right edge of the focused word. However, comparison of reaction times for the processing task indicated that both groups responded more quickly to instructions with natural than unnatural prosody, although English speakers' response times were significantly faster in both conditions. We argue that although Mandarin speakers show Mandarin-like realization of focus in their production, they can nonetheless use the English prosodic patterns in their processing.

1pSC5. Towards a model of Tatar intonational phonology. Adam J. Royer (Linguist, UCLA, 3125 Campbell Hall, Los Angeles, CA 90095-1543, ajroyer@ucla.edu)

This study is a preliminary report of our ongoing research investigating an Autosegmental-Metrical model of intonational phonology of Kazan Tatar, a Turkic language spoken in Tatarstan, Russia. Tonal patterns of neutral focus utterances were examined by varying the length of words and phrases, the location of stresses, syntactic structures, and sentence types. Results suggest that Tatar has three prosodic units marked by intonation. They are the Accentual Phrase (AP), the Intermediate Phrase (ip), and the Intonational Phrase (IP). An AP has one or more words and has an optional initial high tone (Hi), realized at the left edge or leftmost stressed syllable of the AP, and an obligatory rising pitch accent (L + H*) aligned with the rightmost stressed syllable of the AP. Interestingly, the unstressed syllables after the pitch accented syllable show high f0 until the end of the AP, suggesting an optional AP-final H tone (Ha). An ip is marked by a phrase-final high tone (H-) realized on an ip-final syllable, which is slightly longer than AP-final syllable. Finally, an IP is marked by a phrase-final boundary tone realized on a substantially lengthened syllable. A fuller model of Tatar will be compared with the intonation models of other Turkic languages.

1pSC6. The nature of variation in the tone Sandhi patterns of Wuxi Wu. Hanbo Yan (School of Chinese Studies and Exchange, Shanghai International Studies Univ., 550 West Dalian Rd., Bldg. 2, Rm. 418, Shanghai, Shanghai 200083, China, yanhanbo@shisu.edu.cn) and Jie Zhang (Linguist, Univ. of Kansas, Lawrence, KS)

The Northern Wu Chinese dialect of Wuxi has two different tone patterns in disyllables—a pattern with tone sandhi that involves a synchronic chain-shift and a no sandhi pattern, and the two patterns apply variably. This study investigates the nature of this variation. Seventy-one native speakers participated in three rating experiments that investigated disyllables "subjective frequency, semantic transparency, and the variant forms" perceptual goodness. Results show that modifier + noun combinations prefer the sandhi form more than verb + noun combinations. Lexical frequency has a positive effect on sandhi application in both modifier + noun and verb + noun items. Semantically transparent items are less likely to undergo tone sandhi, but only for verb + noun combinations. These results are interpreted with respect to the properties of wordhood in Wuxi and Chinese dialects in general and the productivity of the chain-shift tone sandhi pattern observed in an earlier production study.

1pSC7. Eye movements and speech prosody in the processing of information structure: An exploratory study in Mandarin Chinese. Li Liu, Ying Chen, and Xueqin Zhao (English, School of Foreign Studies, Nanjing Univ. of Sci. and Technol., 200 Xiaolingwei St., Nanjing, Jiangsu 210094, China, ychen@njust.edu.cn)

An exploratory study on information processing in Mandarin discourse was carried out using the remote system of eye tracker. A map was designed to display four destination images, four written words of orientations and four values of distance from the map center to the destinations. Participants answered five pre-recorded questions while reading the map on the computer screen in three trials. The experimenter pretended not to have recorded Trial 1 and requested the participants to do the task again in Trial 2. The questions in Trial 3 were recorded with a different gender to indicate a new inquirer. The preliminary results show some major tendencies: (1) gaze points and saccade routes in Trials 2 and 3 were fewer and less complicated compared to Trial 1; (2) fixation duration on the target words was shorter when the information was old than when it was new; (3) duration, F0 and intensity of the target words were reduced in Trials 2 and 3 compared to Trial 1; (4) the duration tended to be longer, the F0 higher but the intensity weaker in Trial 3 than in Trial 2, suggesting a gender difference in the speech, which was also reflected in the eye-movement data.

1pSC8. Prosodic asymmetry in phonetic reorganization of Seoul Korean 3-way voiceless stop contrast. Yoonjeong Lee and Louis Goldstein (Linguist, Univ. of Southern California, 3601 Watt Way, Grace Ford Salvatori 301, Los Angeles, CA 90089, yoonjeol@usc.edu)

In younger generation Seoul Korean speakers, a phonetic reorganization of VOT and f0 in the phrase-initial stop contrast (i.e. aspirated, lenis, fortis) has been well documented. The current study elucidates how this local consonant effect on f0 further interacts with the global tonal patterns of the accentual phrase (AP). We test how the initial 3-way stop contrast is phonetically realized in AP-initial and AP-internal prosodic positions for six younger generation speakers (born 1980-1990). Our results show that the consonant effect on f0 is *categorical* in AP-initial position, compared to exhibiting a *gradient* effect in AP-internal position. We confirm an AP-initial VOT merger between aspirated and lenis stops, accompanied by an increased between-consonant f0 difference. In AP-internal position, along with a small but significant f0 difference, we found a near-merger of VOT between lenis and fortis stops, arising from the substantially reduced occurrence of intervocalic lenis voicing. Taken together, our findings provide novel evidence of an intricate interaction between phrase-level prosody and the local phonetic reorganization in the newly emerged phonetic system of Seoul Korean stops. [Work supported by NIH.]

1pSC9. Acoustic similarities among voices. Part 2: Male speakers. Jody E. Kreiman, Patricia Keating, and Neda Vesselinova (UCLA, 31-24 Rehab Ctr., 1000 Veteran Ave., Los Angeles, CA 90095-1794, jkreiman@ucla.edu)

Little is known about how to characterize normal variability in voice quality within and across utterances from normal speakers. Our previous study of female voices suggested that only a few acoustic parameters consistently distinguish speakers, with most of the work being done by idiosyncratic subsets of parameters. The present study extends this research to samples of 50 men's voices. The men read 5 sentences twice on 3 days—30 sentences per speaker. The VoiceSauce analysis program estimated means and standard deviations for many acoustic parameters (including F0, harmonic amplitude differences, harmonic-to-noise ratios, and formant frequencies) for the vowels and approximant consonants in each sentence. Multidimensional scaling and linear discriminant analysis were used to examine the acoustic characteristics of the overall voice space, and to measure how well each speaker's set of 30 sentences could be acoustically distinguished from all other speakers' sentences. Additional analyses of small subsets of voices compared the importance of global versus local details in discriminating voices acoustically. Results will be compared to those for female speakers, and implications for recognition by listening will be discussed. [Work supported by NSF and NIH.]

1pSC10. Voice quality variation over the course of the English utterance. Elizabeth M. Bird (BioEng., UCSD, 9500 Gilman Dr, 9252 Regents Rd., E, La Jolla, CA 92093, embird@ucsd.edu) and Marc Garellek (Linguist, UCSD, La Jolla, CA)

The presence of atypical voice quality (e.g. breathy and creaky voice) can be used to diagnose voice disorders, but it is also clear that healthy English speakers' voices vary as a function of phrasing. For example, ends of declarative utterances tend to be produced with creakier voice quality. Yet for speakers who have no diagnosed voice disorder, it is still unclear what factors affect voice variation over an utterance, and how consistent these variations are across speakers. This in turn limits diagnostic utility of perceived non-modal voice. Therefore, the goal of this project is to determine how the voice varies over the course of English utterances. We recorded electroglottographic (EGG) and audio waveforms of 20 male and female speakers of Californian English reading sentences of various lengths that were designed to avoid non-modal voice associated with certain segments. The EGG waveforms were analyzed for contact quotient and f0 at the beginning, middle, and end of the utterance. Discussion will focus on how voice quality differs by position in utterance, and the extent to which it is predictable from f0. We also discuss whether voice quality varies as a function of gender, utterance length, and speakers' average voice quality.

1pSC11. Live versus art in voice change: The case of Maria Callas. Nina Eidsheim and Jody E. Kreiman (UCLA, 31-24 Rehab Ctr., 1000 Veteran Ave., Los Angeles, CA 90403, jkreiman@ucla.edu)

Many commentaries on the voice of singer Maria Callas note that her voice changed markedly over the course of her career, with changes often attributed to "ferocious dieting." Such claims are particularly troubling in the absence of evidence that weight loss affects voice acoustics, and in the relative absence of acoustic data testing specific hypotheses regarding expected changes in voice with dieting. This paper examines recordings from early and late in Callas's career, and attempts to determine whether observed changes are more consistent with the acoustic effects of physiological changes associated with extreme and rapid weight loss (changes in hormone levels, respiratory changes, differences in tongue size/vocal tract dimensions, reflux, etc.), with aging (e.g., increasing vocal instability, changes in resonance frequencies, changes in F0), or with artistic choices.

1pSC12. Phonatory characteristics of *ex-vivo* aged sheep models. Michael Döllinger (Div. for Phoniatrics and Pediatric Audiol. at the ENT Dept., Univ. Hospital Erlangen, Bohlenplatz, 21, Erlangen, Bavaraia 91054, Germany, michael.doellinger@uk-erlangen.de), Markus Gugatschka (ENT University Hospital, Dept. of Phoniatrics, Medical Univ. of Graz, Graz, Austria), Olaf Wendler (Dept. of Otorhinolaryngology, Head and Neck Surgery, Experimental ENT Res. Lab. I, Univ. Hospital Erlangen, Erlangen, Bavaria, Germany), Claus Gerstenberger (ENT University Hospital, Dept. of Phoniatrics, Medical Univ. of Graz, Graz, Austria), Hossein Sadeghi, and Stefan Kniesburges (Div. for Phoniatrics and Pediatric Audiol. at the ENT Dept., Univ. Hospital Erlangen, Erlangen, Bavaria, Germany)

For humans, voice quality reduces with age. Due to the increasing life span and the increase of the older population, it is necessary to develop new strategies to treat people concerned. Since potential new treatment technologies as functional electrical stimulation have to be first established in animal models, we investigated phonatory characteristics in aged sheep larynges. Ex-vivo dynamic experiments were performed on nine aged sheep larynges providing normative phonatory data. The larynges were analyzed at sustained phonation for varying subglottal pressure levels. Additionally, three different weights were successively attached to the thyroid cartilage to induce prestress forces towards the thyroarytenoid muscle and to therefore simulate longitudinal tension of the vocal folds. Laryngeal vibrations, airflow and acoustics were recorded. Afterwards, larynges were analyzed for tissue damage using histological standard methods. Overall, the larynges showed rather asymmetric vibrations and exhibited very soft and pliable vocal fold tissue characteristics. The histological analysis showed (in two cases) minimal damage in form of slight epithelial layer detachments and smaller cracks in the lamina propria. Further and detailed results will be presented. Next, the gathered data and characteristics will be compared to young sheep as well as to aged sheep being treated with functional electrical stimulation.

1pSC13. Comparing roughness in sustained phonations and connected speech using a matching task. Supraja Anand, David A. Eddins (Commun. Sci. and Disord., Univ. of South Florida, 4202 E. Fowler Ave., PCD 1017, Tampa, FL 33620, suprajaanand@usf.edu), and Rahul Shrivastav (Office of the Vice President for Instruction, Univ. of Georgia, Athens, GA)

Experiments on perception of dysphonic voice quality have typically relied on sustained vowel phonations. The strong preference for vowels could be explained by their ease of production, their time invariance, and by the absence of confounding articulatory (e.g., consonant, dialect) and prosodic (e.g., stress, rate) changes. However, the magnitude of roughness perceived from sustained vowels may not reflect the roughness in connected speech produced by the same speaker. We examined how stimulus type impacts the perception of roughness. Ten naïve listeners judged roughness for both vowel /a/ and sentences selected from ten dysphonic speakers using a single-variable matching task. Stimuli were selected to ensure a continuum of vocal roughness for vowels. The intra- and inter-listener reliability estimated using intra-class correlation (ICC) as well as matching values and variability will be compared across each stimulus type. This work is essential for extending models of dysphonic voice quality perception from vowels to connected speech, which are likely to correspond more closely with perceived handicap and better represent relevant treatment targets for patients with dysphonia.

1pSC14. Measurement scales, psychophysical models, and reliability statistics in voice quality perception. Noah H. Silbert (Commun. Sci. and Disord., Univ. of Cincinnati, French East 344, 3202 Eden Ave., Cincinnati, OH 45267, silbernh@ucmail.uc.edu)

Perceptual judgments of voice quality can be given on discrete or continuous scales. Discrete scales are typically described as consisting of "equal appearing intervals," and continuous judgments may be given on visual analog scales (VAS) or via direct magnitude estimation. Interrater agreement on discrete scales is highly variable and often poor, though there is evidence that application of psychophysical principles (i.e., perceptual variability, response bias) can increase reliability. There is some evidence that interrater reliability may be higher with continuous scales, though the influences of psychophysical assumptions and measurement properties on the reliability of continuous scales has not been analyzed particularly thoroughly. The present work considers the effects of different assumptions about underlying perceptual and decisional processes on the measurement properties and reliability of perceptual voice quality scales. An adaptation of Stevens' power law combined with different decision rules provide a model of the mapping between the acoustic properties known to influence voice quality (e.g., H1-H2, HNR) and discrete or continuous clinical judgment scales (e.g., "breathiness," "roughness"). The effects of variation in model parameters on reliability and agreement statistics will be analyzed, and novel agreement statistics based on the properties of ratio-scale judgments will be explored.

1pSC15. On the restoration of Shackleton's voice. SangHwi Jee and Myungjin Bae (Commun. Eng., Soongsil Univ., Seoul, Seoul 06978, South Korea, slayernights@ssu.ac.kr)

Shackleton's voice, which attempted to explore Antarctica 100 years ago, was vividly restored with voice-processing technology. The restored voice was used as a restoration criterion after restoring the sound quality to a sound quality by applying noise processing technology to the voice of Shackleton who finished the exploration and gave a speech at a university. The basic voices of the voices are similar to those of family members and relatives because of the similarity of the gene structure and the body structure. Using Shackleton's great-grandson voices, he restored Shackleton's voice by adjusting the rate of speech, intensity, intonation, duration, tone changes, and tone averages as analyzed in Shackleton's voice. The subjective evaluation of the restored voice showed a similarity of more than 90% in the spectral SNR compared to the original voice, and when MOS evaluation was performed by the linguistic expert, 4.5 / 5 was obtained and the restoration of the voice of Shackleton was successful respectively. It is expected that the restoration technology of the deceased using the relative of relatives can restore the message of love and encouragement to the family in their future voice to the descendants.

1pSC16. Individual differences in the signaling of prosodic structure by changes in voice quality. Stefanie Shattuck-Hufnagel (MIT, 36-511 MIT, 77 Mass Ave., Cambridge, MA 02139, sshuf@mit.edu)

Earlier work has shown that speakers of American English often (although not always) produce irregular pitch periods (or other changes in voice quality) at prosodically significant locations, such as the onset of a new intonational phrase or a pitch-accented syllable, when those constituents begin with a [+voiced] phonemic segment (Pierrehumbert & Talkin 1992). This tendency varied substantially across 5 speakers of FM-Radio-News-style speech (Dilley *et al*. 1996). Further work with speakers of a west-coast dialect (Garellek 2012) raised questions about the appearance of this cue at phrase onsets, when the phrase begins with a prosodically weak (lexically unstressed) syllable; this suggests that it occurs only for phrasally-strong (pitch accented) syllables, whether they occur in phrase-initial position or not. In contrast, extensive analysis of voice quality changes in a corpus of imitated utterances produced by speakers of a mid-western dialect (Cole & Shattuck-Hufnagel 2011) clearly shows a tendency for voice-quality changes at the onsets of prosodically weak syllables when they occur at the onsets of intonational phrases. This raises the possibility that individual speakers or dialects may differ in the likelihood of using changes in voice-quality to mark different types of prosodic events, such as phrase onsets vs pitch accents.

1pSC17. Delayed auditory feedback induces slow oscillations in vocal fundamental frequency and intensity. Jieun Lee and Francois-Xavier Brajot (Commun. Sci. and Disord., Ohio Univ., Gover Ctr. W218, Athens, OH 45701, jl976314@ohio.edu)

A system with negative feedback tends toward oscillation or instability when delay is introduced into the feedback loop. Given current evidence for feedback control of pitch and loudness, we hypothesized that delaying auditory feedback during sustained phonation would result in oscillations in fundamental frequency and vocal intensity. Subject produced a sustained vowel across seven auditory feedback conditions: 0, 100, 200, 300, 400, 500, 600, and 700 ms delay. Increasingly large oscillations in fundamental frequency and intensity were observed as delay increased. The frequency of oscillations as a function of delay appear to be subject-specific, with a range between 0.3 and 0.8 Hz. These oscillations carried over into the prosodic contour of subjects' connected speech produced under the same delay conditions. These results suggest that disfluencies induced by delayed feedback (e.g., sound prolongations) may be the result of compensation for shifts in pitch and loudness, rather than discrepancies in articulatory or segmental timing.

1pSC18. Volume velocity field and acoustics measurements in a canine larynx model. Charles P. Farbos de Luzan, Liran Oren, Sid M. Khosla, Alexandra Maddox (Dept. of Otolaryngol. - HNS, Univ. of Cincinnati, Medical Sci. Bldg., Rm. 6303B, Cincinnati, OH 45267-0528, farboscs@ucmail.uc.edu), and Ephraim Gutmark (Aerosp. Eng., Univ. of Cincinnati, Cincinnati, OH)

In the classic source-filter theory, sound is produced at the glottis by a process known as flow modulation; in this case, "flow" specifically refers to the flow rate (Q) produced at the glottal exit during the phonation cycle. Flow modulation refers to the fact that Q is changing as the glottis opens and closes dQ/dt. Although dQ/dt is constantly changing from glottal opening and closing, the greatest rate of change happens during the latter part of closing, when Q rapidly decreases. This rapid deceleration is quantified by the maximum flow declination rate (MFDR). MFDR has been shown to highly correlate with acoustic intensity (loudness). The aim of this study is to measure changes in Q and the acoustic energy in a vibrating canine larynx model as a function of subglottal pressure and vocal tract constriction. Volume flow measurements are taken using time-resolved tomographic-PIV. Q at the glottal exit is extracted from PIV measurements and MFDR is computed from the waveform. Acoustic measurements (SPL) are taken simultaneously. Testing is done with and without vocal tract, which is place above the larynx. The constriction in the vocal tract is varied by changing the distance between the 2 false vocal folds (FVF). Each case above is tested at low and high subglottal pressures. Measurements show that the glottal exit flow is complex. The waveform of Q is skewed towards closing phase, even without a vocal tract. The skewing is affected further by the vocal tract constriction. Both MFDR and SPL increase with subglottal pressure.

1pSC19. Cooperation via communication: Influencing vocal alignment in conversation. Elliot Pollack and Elizabeth D. Casserly (Psych., Trinity College, 300 Summit St., Box 702353, Hartford, CT 06106, elliot.pollack@trincoll.edu)

Alignment of human vocal behavior is a well-documented phenomenon, however, the factors which influence its direction and magnitude are not firmly established. Components of speech which are subject to alignment include—but are not limited to—word choice, syntax, and rate of speech. In the present study, Speakers completed a puzzle task which required them to communicate with a Model whose fundamental frequency (F0) changed during the interaction. Additionally, naïve subjects (Listeners) assessed the overall similarity between the Speaker's speech over time and the Model using an AXB paradigm. Speakers were found to deviate from the Model in F0, however, were perceived by Listeners to mimic the Model in a holistic measure. Speakers were rated as becoming more like the Model when this partner diverged as opposed to converged. A personality factor survey showed that greater Openness predicted less perceived similarity. The discrepancy between divergence in the acoustic measure and convergence in the perceptual measure reveals a potential hierarchy of speech factors that we use to assess alignment.

Session 1pSP

Signal Processing in Acoustics, Underwater Acoustics, and Engineering Acoustics: Source Tracking with Microphone/Hydrophone Arrays II

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Invited Papers

1:00

1pSP1. Direction finding and inverse imaging with microphone arrays in a forest. Michael J. White, Michelle E. Swearingen, and Patrick J. Guertin (US Army ERDC/CERL, PO Box 9005, Champaign, IL 61826, michael.j.white@usace.army.mil)

Beneath the tree canopy, wind and temperature gradients are reduced, and so are wind noise and atmospheric fluctuations that cause acoustic scintillation. However, the large number of discrete scatting objects complicates the received signal in a repeatable manner. We placed three small arrays in a forest and operated a propane cannon in the area nearby to investigate the spatial and angular dependence of the received field and its correlations. As expected, scattering from trunks impairs the ability to find the source using the conventional beamformer. Several direction finding techniques were employed to evaluate the signal direction in this complex environment. Methods are compared against both measured data and a numerical scattering model for the same location.

1:20

1pSP2. Direction of arrival estimation using local series expansion evaluated on acoustic data. Gustaf Hendeby (Elec. Eng., Linköping Univ., LiU, Linköping, N/A 581 83, Sweden, gustaf.hendeby@liu.se), Thomas Lunner (Tech. Audiol., Dept. of Experimental and Clinical Res., Linköping Univ., Snekkersten, Denmark), and Fredrik Gustafsson (Elec. Eng., Linköping Univ., Linköping, N/A, Sweden)

Direction of arrival (DOA) estimation using a local series expansion of the signal is evaluated on acoustic data. It was recently proposed that DOA estimation using a sensor array could be performed using a local temporal/spatial series expansion. That is, to use the local phase of the signal rather than its estimated second order moments or correlation as in classical methods. The approach should provide easier handling of general sensor configurations, ability to deal with very low sampling rates and few samples, and benefits from miniature sensor configurations. Additionally, the signal is not required to be narrow-banded, as sufficient smoothness is enough. Here, the local series expansion method is benchmarked to a standard beam-forming method using experimental data, recorded in an anechoic chamber at the Technical University of Denmark (DTU). The dataset was recorded using 52 high quality microphones distributed on a sphere with radius 50 mm, and contains several recordings from one or two well defined sources from several different directions, with different levels of reverberation. The dataset is used to evaluate important aspects of the new method and its potential to improve on standard DOA methods.

1:40

1pSP3. Detection of reflections in different room acoustical conditions using generalized cross correlation phase transform. HyunIn Jo, Jong Gak Seo, and Jin Yong Jeon (Architectural Eng., Hanyang Univ., Seoul, Seongdong-gu 133-791, South Korea, best2012@naver.com)

A framework for localization of the received source emission was implemented in enclosures based on generalized cross correlation (GCC) weighting with phase transform (PHAT) function. It was to achieve the goal of capturing meaningful sounds including speech in actual environments. The effects of room reverberation and ambient noise on the performance of the system were investigated with a linear microphone array. Experiments were performed in different environments with different RT and background noise levels. The sound sources are consisting of pink noise and speech sources were used to observe the relationship between localization accuracy and room acoustical parameters. Results for evaluation indicated that the location of the reflections was noticeable for accurately detection of the sources in reverberant and noisy environments.

2:00

1pSP4. Array element localization using multidimensional scaling (MDS) and matrix completion, with comparisons to more conventional non-linear least squares optimization. Paul Hursky (Sonar-synesthetics, 4274 Pilon Point, San Diego, CA 92130, paul.hursky@gmail.com)

Multidimensional scaling (MDS) is an approach from the social sciences which was invented to project qualitative notions of distance onto 2D or 3D coordinate axes to provide visual displays for clustering purposes. As it turns out, this provides a viable algorithm for array element localization. When we calibrate an array using acoustic or RF ranging signals we measure inter-element distances and form a Euclidean Distance Matrix (EDM), and then convert them to 3D orthogonal coordinates. MDS converts the EDM to AEL coordinates. This process has some drawbacks. We may not get all the inter-element distances, because our ranging signals may have limited detection ranges, or because of obstructions to the line of sight. Also, it is not clear how to incorporate knowledge of unequal measurement errors among the EDM entries. Matrix completion is an approach from compressed sensing that allows us to perform the MDS paradigm, despite missing values in the EDM. We will present simulations for typical AEL scenarios and compare AEL results from MDS and matrix completion with results obtained using non-linear least squares optimization.

2:20

1pSP5. Subarray processing with coprime and minimum redundancy arrays. Guen-Soo Jo and Jung-Woo Choi (School of Elec. Eng., KAIST, Daehak-ro 291 KAIST, LG Innovation Hall(N24) 4106, Daejeon KS015, South Korea, thig1399@kaist.ac.kr)

A coprime sensor array (CSA) refers to a non-uniform linear array consisting of two undersampled uniform linear arrays (ULAs) with coprime numbers of elements, respectively. Its beam pattern can be enhanced by multiplying beampatterns of individual subarrays. This technique, called product processing, provides improved resolution because its sum coarray length is extended from the product of beampatterns. Nevertheless, the extended coarray inevitably includes holes that result in higher peak sidelobes than the full ULA case. In this paper, we propose a technique to reduce peak sidelobes through the combination with minimum redundancy arrays (MRAs) and an additional subarray. Two ULAs comprising the coprime array are thinned by the MRA technique, and the reduced number of sensors is then utilized to construct the third subarray. The third array suppresses peak sidelobes by placing nulls at the peak sidelobe locations. The characteristics of the proposed technique are analyzed in terms of white noise gain (WNG) and maximum sidelobe level (MSL). It is demonstrated that the proposed technique can improve the MSL with similar number of sensors, thereby reducing the estimation error in the presence of multiple sources.

2:40–2:55 Break

2:55

1pSP6. Recording and post-processing speech signals from magnetic resonance imaging experiments. Juha Kuortti and Jarmo Malinen (Dept. of Mathematics and Systems Anal., Aalto Univ., P.O. Box 11100, Helsinki FI-00076, Finland, juha.kuortti@aalto.fi)

Speech recordings during an Magnetic Resonance Imaging (MRI) experiment yield valuable but noisy acoustic data for modelling purposes. Despite recent improvements in optical microphones for MRI, using an acoustic sound recording system in dipole configuration, based on shielded electret microphones and waveguides, has some inherent advantages: For example, the bandwidth can be made very wide, and the extremely linear behaviour of all components facilitates the numerical post-processing of signals. Here, the full measurement chain and the critical design decisions are reviewed. In particular, one must take into account the resonant behaviour of the necessarily non-optimally terminated waveguides as well as the environment acoustics within the MRI coils. Two competing approaches for the signal post-processing were developed: (i) Optimal subtraction of the noise and speech signals, and (ii) spectral peak removal by adaptively fitted comb filters. The main objective is the high-quality spectral identification, estimation, and classification of source features, e.g., vowel formants. Approach (i) preserves the phase behaviour of the original signal. Approach (ii) produces excellent speech sound quality when the noise consists of only few harmonic sources; a salient property of MRI acoustic noise. Both of the approaches must compensate the frequency dependent damping of the measurement chain.

3:15

1pSP7. Enhancing angular resolution by coherent beamforming of full non-uniformly spaced linear hydrophone array. Delin Wang and Purnima R. Makris (Elec. and Comput. Eng., Northeastern Univ., 360 Huntington Ave., 302 Stearns Ctr., Rm. 312, Boston, MA 02115, wang.del@husky.neu.edu)

Uniformly-spaced apertures or subapertures of discrete linear receiver arrays are often used in remote sensing to increase signal-to-noise ratio (SNR) by coherent beamforming which reduces noise coming from directions outside the signal beam. To avoid grating lobes in real spatial directions, the uniformly-spaced array inter-element spacing d determines the maximum or cut-off frequency $f<c/2d$ of signals suitable for beamforming with the array, where c is the wave speed. Here we show that a non-uniformly spaced array, for instance formed by combining multiple uniformly-spaced subapertures of a nested linear array, can significantly improve the angular resolution while simultaneously avoiding dominant grating lobes in real angular space, even for signals with frequencies beyond the cut-off. The array gain, beamwidth, and maximum grating lobe height are calculated for the ONR Five Octave Research Array (FORA) for various combinations of its uniformly-spaced subapertures, including the full non-uniformly spaced array. Examples are provided of angular resolution enhancement with resulting non-uniformly spaced array or subarray in both active and passive ocean acoustic waveguide remote sensing over that of the uniformly spaced counterpart, including continental-shelf scale imagery of fish populations, and marine mammal vocalization SNR enhancement which improves detection and bearing-time estimation for passive localization.

3:35

1pSP8. Underwater target detection based on fourth-order cumulants beamforming. Xiukun Li, Hongjian Jia, and Mei Yang (College of Underwater Acoust. Eng., Harbin Eng. Univ., Rm. 1301, Shuisheng Bldg., No.145 Nantong St., Nangang District, Harbin, Heilongjiang 150001, China, jiahongjian_hrbeu@outlook.com)

Obtaining the target echo data of high signal to reverberation ratio (SRR) or signal to noise ratio (SNR) in the complex ocean environment is a key problem to improve the detection capability of underwater targets. In this paper, combined with the expansion characteristics of fourth-order cumulants to the array aperture and the structure of the smallest redundant array, a robust and high-resolution fourth-order cumulants beamforming method based on the smallest redundant array is presented. The target direction-of-arrival estimation (DOA) is achieved by forming a narrow space beam, while the reverberation interference is reduced. For the broadband detection system, combined with fractional Fourier transform (FRFT), the fourth-order cumulants beamforming method is extended to the DOA estimation of LFM signal. According to the signal characteristics difference between the target echo and the reverberation, the reverberation is filtered out in the FRFT domain, which achieves the secondary suppression of reverberation interference in the spatial domain and the transform domain, and further improves the SRR of the target echo signal. The validity of the research method is verified by the processing results of the active sonar target detection experiment.

3:50

1pSP9. Research on distributed beam forming technology in underwater communication. Hai X. Sun, Qi Jie, and liao C. Qiang (Communications Eng., Xiamen Univ. School of Information Sci. and Eng., Xiamen, FuJian 361005, China, hxsun@xmu.edu.cn)

In underwater collaborative network communication, the use of distributed beam forming technology can greatly improve the communication speed and reliability of the network, thus improving the performance of the system. Currently, a vector code based on the feedback information from the destination node to the relay node is used to design the vector code base of the beam, which is widely used in wireless communication. Because of the complex multi-diameter effect of the sound channel and the severity of the variability, the probability of the feedback error is very high. This can be fatal to the system of designing the beam forming technology through feedback information. This paper analyzes the method of feedback error of distributed beam forming system, and gives the approximate expression of feedback error probability and system performance. This paper presents a scheme to improve the feedback information by using the numerical search to improve the performance of feedback information. Through the simulation experiment analysis, the proposed scheme can obviously improve the data rate of the system.

4:05

1pSP10. Design of an unmanned aerial vehicle based on acoustic navigation algorithm. Meng Y. Gong, Hu P. Xu (Logistics Eng. College, Wuhan Univ. of Technol., Yujiayou Campus, Wuhan, Hubei 430063, China, 476793382@qq.com), and Yu H. Hu (Univ. of Wisconsin-Madison, Madison, American Samoa)

Autonomous unmanned aerial vehicles (UAV) navigate using Global Positioning System (GPS). However, in places where GPS signals are interfered, or blocked (e.g., tunnel, downtown with high-rise buildings), auxiliary navigation systems are desirable. In this work, an acoustic based near field ($<$ 1km) UAV guidance system is proposed. This system utilizes acoustic phase array mounted on UAV to provide direction of arrival estimates of narrow band acoustic sources at known locations. As such, the positions of UAV may be easily tracked. Through sensor fusion, this acoustic navigation system may also work together with GPS to provide even more accurate estimates.

Session 1pUW

Underwater Acoustics: Underwater Acoustic Propagation: Models, Methods, and Statistics

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Contributed Papers

1:00

1pUW1. Estimation of the probability density function of transmission loss in the ocean using area statistics. Brandon Patterson and David R. Dowling (Univ. of Michigan, 1231 Beal Ave., Rm. 2016, Ann Arbor, MI 48109, awesome@umich.edu)

Predictions of acoustic transmission loss (TL) in the ocean and its uncertainty are useful for a variety of ocean and naval engineering applications. Previously, an ad-hoc technique, area statistics (AS), was developed as a computationally efficient method to estimate the probability density function (PDF) of TL in any uncertain ocean environment from a single range-depth TL-field calculation in that environment. Here, the performance of AS is reported for ten ocean environments having uncertain sound speed profile, bathymetry, and seabed properties. In each environment, a parabolic-equation-based prediction of TL from a point source was generated as a function of range and depth at frequencies of 100, 200, and 300 Hz using the most probable values of the uncertain environmental parameters. From these calculations, AS was used to estimate the PDF of TL at more than 3000 locations within the ten environments. These estimated PDFs are then quantitatively compared, via an L1-error norm, to PDFs generated from traditional Monte Carlo techniques for the same locations and frequencies. Current results show that the L1-error of the AS PDF is less than 0.5, indicating 75% or greater overlap between the AS and Monte Carlo PDFs, at 90% of the test locations. [Sponsored by ONR.]

1:15

1pUW2. Statistics of bottom-diffracted surface-reflected arrivals in ocean acoustics. Ralph A. Stephen (RASCON Assoc. LLC, 360 Woods Hole Rd., Woods Hole, MA 02543-1592, rstephen@whoi.edu), S. T. Bolmer (Geology and Geophys., Woods Hole Oceanographic Inst., Woods Hole, MA), Peter F. Worcester, and Matthew A. Dzieciuch (Scripps Inst. of Oceanogr., La Jolla, CA)

Bottom-diffracted surface-reflected (BDSR) arrivals are a ubiquitous feature in long-range ocean acoustic propagation and are not predicted by existing forward models based on available bathymetric and bottom properties data. In a research cruise in the North Pacific in 2013 over 40 distinct bottom diffractor locations were identified within a 25 km radius survey (at least one diffractor location every 50 square kilometers). The BDSRs can be characterized in terms of: (a) grazing angle of the incident field, (b) in-plane or out-of-plane diffractors, (c) frequency (transmissions were made from 77.5 to 310 Hz), (d) receiver type (vertical or horizontal seismometer, hydrophone, etc.), (e) receiver location, (f) signal strength relative to direct and water multiple paths, and (g) location relative to bathymetric features. In this talk, we present the statistics of the BDSRs in terms of these criteria. This information will inform efforts to predict and exploit BDSRs and will be valuable in planning future cruises to identify the geologic features responsible for them. [Work supported by ONR.]

1:30

1pUW3. Analysis and comparison of spatial correlation and intensity distribution statistics. Chad M. Smith and Daniel C. Brown (Appl. Res. Lab., The Penn State Univ., State College, PA 16804, chad.smith@psu.edu)

Active sonar processing systems operating in shallow-water littoral environments are often plagued with excessive false alarms, generally referred to as sonar clutter. At times the effects of these clutter events can be mitigated through repeated measurements and/or careful sonar operator examination; however, this process can be time consuming and even experienced sonar operators may have difficulty identifying targets in cluttered environments. Additionally, increasing interest in autonomous vehicle platforms and the push for real-time decision and performance management introduces further requirements on sonar systems. This work provides an empirical analysis of the potential in using spatial correlation of the backscattered wavefront as a classification metric, and compares this metric to a commonly used probability density function (PDF) processing method. The shape of the PDF of the backscattered acoustic intensity is often used in sonar processing systems such as constant false alarm rate (CFAR) detectors. These PDFs generally provide an estimate of the Rayleigh-like (random phase scattering) or non-Rayleigh-like (generally finite bright scatterers) nature of temporal variations in acoustic returns. This talk will outline information that may be available about clutter events based on measurements of spatial correlation. [Work supported by Office of Naval Research.]

1:45

1pUW4. Sensitivity of acoustic propagation modeling to variations in ocean environmental conditions. Tetyana Margolina, John E. Joseph (Oceanogr., Naval Postgrad. School, 833 Dyer Rd., Monterey, CA 93943, tmargoli@nps.edu), and Mary Jordan (Meteorol., Naval Postgrad. School, Monterey, CA)

A parabolic equation acoustic model has been used to estimate transmission loss for a mid-frequency sound source to assist in marine mammal behavioral response studies. The sound propagation is modeled for a range-dependent environment using the Navy Digital Bathymetric Data Base and Bottom Sediment Type database, NOAA Global Ocean Sediment Thickness Dataset, and the High-Resolution Global Sea Surface Wind Speed monthly climatology. The sound speed along the propagation path is modeled using two ocean temperature/salinity fields of different spatial and temporal resolution: monthly 0.25 deg outputs of the Generalized Digital Environmental Model (GDEM), and the daily 1/12 deg outputs of the regional Hybrid Coordinate Ocean Model (HYCOM). The modeling was done for two operational areas with strikingly different ocean environments, the Southern California Bight and the Gulf Stream region off Cape Hatteras. The latter represents a much more dynamic ocean environment with profound temperature fronts, highly variable meso- and submesoscale eddy activity, and sharp bathymetry gradients along the shelf break. Received sound levels

measured on tagged marine mammals, and synchronized sound source/sound receiver tracklines are used to estimate uncertainty of the acoustic modeling results, and the non-linear sensitivity of the model to wide-range variations in the ocean environmental conditions.

2:00

1pUW5. Geoacoustic sediment model discrimination using acoustic color comparison. Anthony L. Bonomo and Marcia J. Isakson (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78713, anthony.bonomo@gmail.com)

Many competing geoacoustic models exist to represent sandy sediments. These models can be fluid, elastic, or poroelastic, depending on the number and types of waves the sediment is assumed to support. In this talk, a fully scattered field finite element approach is utilized to construct acoustic color templates for a buried spherical aluminum shell and a spherical bubble. The simulated results are then used to assess the ability of target strength measurements to discriminate between the predictions of the competing sediment models. Such discrimination may help determine the physical validity of a given geoacoustic model and aid in model selection. Five geoacoustic sediment models are considered: a simple fluid, constant-Q fluid model, the effective density fluid model of Williams, the viscous grain shearing model of Buckingham, the Biot-Stoll poroelastic model, and the extended Biot model of Chotiros. [Work supported by ONR, Ocean Acoustics.]

2:15

1pUW6. Finite impulse response of an ocean acoustic waveguide by compressed sensing. Richard S. Keiffer (Acoust. Div., Naval Res. Lab., Bldg. 1005, Rm. C-2, Stennis Space Ctr., MS 39529, richard.keiffer@nrlssc.navy.mil)

The usual approach to calculating the time domain acoustic response of an ocean waveguide is to Fourier synthesize it from a regularly sampled collection of frequency domain solutions. Depending on the details of the waveguide and the excitation signal, the resulting time series may be very sparse with significant time intervals of relative quiet between energetic multipath arrivals. This observed sparsity in the time domain suggests a computational advantage to synthesizing the time series by solving a convex optimization problem involving a much smaller set of solutions computed at random frequencies. The efficiency of the compressed sensing approach will depend on the sparsity of the time domain solution; the greater the sparsity in the time domain signal the greater the expected advantage of the compressed sensing approach. In this talk we examine the compressed sensing method of time series synthesis for a sequence of waveguides that range from extremely simple and ideal waveguides where the sparsity is expected to be great to the more realistic and complicated where the sparsity is expected to be reduced. [This work was supported by the U.S. Office of Naval Research.]

2:30

1pUW7. Discrete element method simulations for acoustic propagation through idealized seafloor sediments. Joseph Calantoni, Adam Seyfarth, Pedro M. Jordan, and Richard S. Keiffer (U.S. Naval Res. Lab., 1005 Balch Blvd., Stennis Space Ctr., MS 39529, joe.calantoni@nrlssc.navy.mil)

We performed discrete element method (DEM) simulations for the propagation of pressure and shear waves in idealized seafloor waveguides. Preliminary simulations used identical spheres configured in a random loose pack. DEM waveguides were essentially one-dimension with a 10 by 10 grain diameter cross section and varied in length from 20,000 to 100,000 grain diameters for grains ranging from 0.2 to 0.5 mm in diameter with the material properties of quartz. Rough walls were placed at the ends of the waveguide and periodic boundaries were used in the lateral direction. Pressure and shear wave propagation was simulated through normal and tangential sinusoidal oscillation of the boundary wall at frequencies ranging from 100 to 1,000 Hz. Sensitivity of propagation speed and attenuation to the specification of the material properties of the grains was investigated. For example, our simulations demonstrate the well-known dependence of pressure wave propagation speed on grain stiffness. Discussion will focus on creating links between simulation results and continuum models for acoustic propagation in seafloor sediments.

2:45–3:00 Break

3:00

1pUW8. Measurements of the acoustic-field autoproducts. Jessica E. Lipa (Mech. Eng., Univ. of Michigan, 1231 Beal Ave., Ann Arbor, MI 48109-2133), Brian M. Worthmann (Appl. Phys., Univ. of Michigan, Ann Arbor, MI), and David R. Dowling (Mech. Eng., Univ. of Michigan, Ann Arbor, MI, drd@umich.edu)

The frequency-difference and frequency-sum autoproducts are quadratic products of acoustic fields at different frequencies extracted from a frequency-domain acoustic field with nonzero bandwidth. In well-ensonified regions of ray-path environments, the autoproducts may—in theory—mimic genuine acoustic fields at frequencies below and above the original field's bandwidth (Worthmann and Dowling, 2017, JASA **141**, 4579-4590). For example, in a Lloyd's mirror environment, the autoproducts successfully mimic genuine out-of-band acoustic fields except near the surface or the source. In this presentation, results from a water tank experiment are shown that confirm predictions from theory. A 40- to 110-kHz pulse signal was broadcast from a source 20 cm below the surface to a hydrophone positioned at three different ranges between 20 and 50 cm and at depths from 0 to 40 cm. The measured frequency-difference autoproduct fields have cross correlations of greater than 90% with genuine acoustic fields between 0 and 60 kHz. Similarly, the measured frequency-sum autoproduct fields have cross correlations of greater than 90% with genuine acoustic fields between 110 and 190 kHz. Furthermore, the experimentally measured autoproducts correlate well with theoretically predicted autoproducts. Thus, these results serve to confirm the theory. [Sponsored by ONR and NSF.]

3:15

1pUW9. Autoproducts in refractive waveguides near shadow zones. Brian M. Worthmann (Appl. Phys., Univ. of Michigan, 1231 Beal Ave., Ann Arbor, MI 48109, bworthma@umich.edu) and David R. Dowling (Mech. Eng., Univ. of Michigan, Ann Arbor, MI)

The frequency-difference and frequency-sum autoproducts are quadratic products of nonzero-bandwidth frequency-domain acoustic fields at different frequencies but the same spatial location. The autoproducts have been shown to mimic genuine lower- or higher-frequency, out-of-band, acoustic fields within some limitations (Worthmann and Dowling, 2017, JASA **141**, 4579–4590). Previous analysis of the autoproducts considered only spatial regions of the environment that were well-ensonified by the in-band acoustic field, which itself is well-described by the ray-approximation. In this presentation, the necessity of this well-ensonified restriction is analyzed by comparing autoproduct fields with their out-of-band analogues in and near acoustic shadow zones created by refraction. The results presented are derived from exact or asymptotic solutions of the Helmholtz equation. The propagation environment chosen for this study is a horizontal, range-independent, stratified, unbounded, truncated-n^2-parabolic waveguide, which creates acoustic shadow zones in some regions of the acoustic field. By analyzing the autoproducts near the transition from well-ensonified region to shadow zone, the potentially detrimental effects of diffraction on the autoproducts' mimicry of out-of-band acoustic fields can be analyzed and quantified. [Sponsored by ONR and NSF.]

3:30

1pUW10. A higher-order approximation to a hybrid parabolic equation approach for density discontinuities. Kevin B. Smith (Dept. of Phys., Naval Postgrad. School, 833 Dyer Rd., Bldg. 232, Rm. 114, Monterey, CA 93943, kbsmith@nps.edu)

Parabolic equation models based on the highly efficient and stable split-step Fourier (SSF) algorithm have been known to suffer from accumulating phase errors in range when bottom interactions are prevalent. The primary cause has been identified as due to the treatment of the density discontinuity at the water/sediment interface, whereby a smoothing function is introduced to model the step function. The scale of this smoothing function is proportional to wavelength, and can distort the environmental representation at low frequencies. Yevick and Thomson (1997) introduced a hybrid approach to the problem whereby the density terms were treated separately from the

SSF approach via a finite-difference approximation. While this approach was shown to significantly reduce the aforementioned phase errors, the solutions were found to be highly sensitive to the choice of depth mesh size, making general usage of this approach less optimal. The finite difference approach of Yevick and Thomson was based on a 2nd order approximation of the differential operators. In this work, higher order approximations are applied in an attempt to stabilize the finite difference approximations. Solutions are presented for a simple shallow water waveguide utilizing the various approaches previously investigated as well as the higher order approximations. The results are compared with respect to both solution accuracy as well as numerical stability.

3:45

1pUW11. Exploration of acoustic spiral waves in an underwater environment. Grant Eastland (Test and Evaluation, Naval Undersea Warfare Ctr. Div. Keyport, 610 Dowell St., Keyport, WA 98345, grant.eastland@navy.mil)

The work to be presented is regarding a research project to investigate the viability and possible application of spiral or helical acoustic waves for use in underwater environments. The investigation encompasses studies into the properties and generation of spiral or helical acoustic waves. Spiral waves can be generated by precisely controlling the phase of the wave, which could have different physical properties due to what is known as "topological charge." This topological charge could be adjusted in some advantageous way, depending upon the desired application. In addition, reception of spiral or helical waves was investigated. The goal of the research is a proof of concept and presentation of the results of generation and reception of spiral acoustic waves. Spiral waves have been investigated for acoustic navigation and may lend itself well for use in other applications.

4:00

1pUW12. Using the argument of the Airy function as an analysis tool in ocean acoustics and oceanography. Stanley A. Chin-Bing (Phys. Dept., Univ. of New Orleans, 2000 Lakeshore Dr., New Orleans, LA 70148, chinbing@att.net) and Josie P. Fabre (Acoust. Div. Code 7180, Naval Res. Lab., Stennis Space Ctr., MS)

By assuming the index of refraction is linear between layers, it is possible to convert the depth dependent acoustic Helmholtz equation into an Airy equation. Decades ago this allowed the ocean acoustic pressure to be quickly calculated using tables of precalculated Airy functions with interpolation. High speed computers and the accurate parabolic equation model have rendered the Airy equation approach as a curiosity of history. However, the method can still be useful as an assessment tool. This has been demonstrated by using it to select similar ocean environmental parameters that have the same acoustic propagation characteristics. In this presentation, we apply the Airy equation method to evaluate the buoyancy frequency of similar ocean environments that support internal waves. A connection is made between the ocean acoustic propagation and the oceanography of the similar regions.

4:15

1pUW13. A fast calculation method of travel speed of pulse peak in convergence zone. Yun Ren, Renhe Zhang, Jun Wang, and Yonggang Guo (State Key Lab. of Acoust., Inst. of Acoust., Chinese Acad. of Sci., No. 21 North 4th Ring Rd., Haidian District, Beijing 100190, China, renyun@mail.ioa.ac.cn)

A long-range sound propagation experiment was conducted in the West Pacific Ocean in summer 2013. The signals received by a towed array indicate that the travel speed of pulse peak (TSPP) in the convergence zones is stable. Therefore, an equivalent sound speed can be used at all ranges in the convergence zones. A fast calculation method based on the beam-displacement ray-mode (BDRM) theory and convergence zone theory is proposed to calculate this equivalent sound speed. The computation speed of this proposed method is over 1000 times faster than that of the conventional calculation method based on the normal mode theory, with the computation error less than 0.4% compared with the experimental result. Also, the effect of frequency and sound speed profile on the TSPP is studied with the conventional and fast calculation methods, showing that the TSPP is almost independent of the frequency and sound speed profile in the ocean surface layer.

4:30

1pUW14. Analysis of sound propagation from the transitional area to deep water. Jixing Qin, Renhe Zhang, and Zhenglin Li (Inst. of Acoust., Chinese Acad. of Sci., No. 21 North 4th Ring Rd. Haidian District, Beijing 100190, China, qjx@mail.ioa.ac.cn)

Underwater acoustic propagation over the continental shelf and slope is complicated and is also an important issue. Motivated by a phenomenon in an experiment conducted in the South China Sea indicating that the energy of the received signal around the sound channel axis is significantly stronger than that at shallower depths, we study sound propagation from the transitional area to deep water. Numerical simulations with different source depths are first performed, from which we reach the following conclusions. When the source is placed near the sea surface, acoustic wave will be strongly attenuated by bottom losses in a range-independent deep-water environment, whereas it can propagate to a very long distance because of the continental slope. When the source is mounted on the slope bottom in shallow-water area, acoustic energy will be trapped near the sound channel axis, and it converges more evidently than the case where the source is located near the surface. Then, simulations with different source ranges are performed. By comparing the relative energy level in the vertical direction between the numerical and experimental results, the range of the unknown air-gun source is roughly determined. [Work supported by the National Natural Science Foundation of China under Grant Nos. 11434012 and 41561144006.]

Session 1eIDa

Interdisciplinary: Special Presentation on The Clarinet in Early New Orleans Jazz

David Woolworth, Chair
Roland, Woolworth & Associates, 365 CR 102, Oxford, MS

Chair's Introduction—5:00 p.m.

Invited Paper

5:05

leIDa1. The Clarinet in Early New Orleans Jazz: Style, Tone and Function. Michael White (Xavier Univ. of Louisiana, 1 Drexel Dr., New Orleans, LA 70125)

Very different from its sound in classical music, swing, modern jazz or other genres, the authentic New Orleans Jazz clarinet tradition remains among the most distinctive instrumental styles in American music. This program will focus on sound and tonal variety of the clarinet in early jazz in its role as an accompanying or featured instrument in dance ensembles and its high wailing voice in brass bands and jazz funerals. There will be a discussion of the instrument's distinguishing characteristics and tonal possibilities. The high level of diversity of sound among early jazz clarinetists will also be examined as affected by equipment, training and physical characteristics of individual players. Live musical examples will be provided by the Dr. Michael White Quartet. Dr. Michael White is a leading figure in traditional New Orleans jazz and one of only a few to creatively carry on the rich clarinet sound and style of that city. For over two decades he has been the main consultant for traditional jazz for the New Orleans Jazz & Heritage Festival. A relative of several first generation jazz musicians, he has distinguished himself as a jazz historian, writer, producer, and composer. In 2015 he received the Jazz Hero Award from the Jazz Journalists Association. Dr. White earned his PhD in Spanish at Tulane University and currently holds the Charles Keller Endowed Chair in the Humanities at Xavier University of Louisiana.

Payment of a separate registration fee is required to attend this session.

MONDAY EVENING, 4 DECEMBER 2017 SALON D, 7:00 P.M. TO 9:00 P.M.

Session 1eIDb

Interdisciplinary: Tutorial Lecture on Infrasound Phenomenology, Propagation, and Detection

Karim G. Sabra, Chair
Mechanical Engineering, Georgia Institute of Technology, 771 Ferst Drive, NW, Atlanta, GA 30332-0405

1p MON. PM

Chair's Introduction—7:00

Invited Paper

7:05

1eID1b. Infrasound phenomenology, propagation, and detection. Roger M. Waxler (NCPA, Univ. of MS, 1 Coliseum Dr., University, MS 38677, rwax@olemiss.edu)

Infrasound is generally understood to refer to acoustic disturbances in the atmosphere in frequency bands below the threshold of human hearing, but above the frequencies at which internal gravity waves propagate. The infrasonic band is nominally taken to be 0.05 Hz up to about 20 Hz. Infrasonic signals tend to be generated by large, violent events and propagate efficiently in sound ducts formed by wind jets and temperature gradients in the middle and upper atmosphere. Infrasonic signals can be detected at very large distances from the source, sometimes even globally. This tutorial will present an overview of the generation of infrasound by both natural and anthropomorphic sources and of the subsequent propagation and detection of infrasonic signals. Two particular sources will be discussed in detail: the signal generated by a large explosion and the so-called microbarom signal generated by colliding ocean waves. Signal propagation through the atmosphere will then be discussed. Available open source infrasound propagation packages and their use will be introduced, the significant atmospheric sound ducts will be identified, and the difficulties inherent in modeling propagation through a dynamic atmosphere will be emphasized. The current state and availability of atmospheric specification will be touched upon. Finally, the use of array processing for the extraction of infrasonic signals from the pressure fluctuations inherent to a turbulent atmosphere will be discussed.