# Session 1aAB

## Animal Bioacoustics: Topics in Animal Bioacoustics I

James A. Simmons, Chair
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**Chair's Introduction—8:25**

### *Contributed Papers*

**8:30**

**1aAB1. Spinner dolphins (*Stenella longirostris* GRAY, 1828) acoustic parameters recorded in the Western South Atlantic Ocean.** Juliana R. Moron, Artur Andriolo (Instituto de Ciências Biológicas, Universidade Federal de Juiz de Fora, Rua Batista de Oliveira 1110 apto 404 B, Juiz de Fora 36010520, Brazil, julianamoron@hotmail.com), and Marcos Rossi-Santos (Centro de Ciências Agrárias, Ambientais e Biológicas, Universidade Federal do Recôncavo da Bahia, Cruz das Almas, Brazil)

Spinner dolphins bioacoustics were study only in Fernando de Noronha Archipelago region in the Western South Atlantic Ocean. Our study aimed to describe the acoustic parameters of this species recorded approximately 3500 km south of Fernando de Noronha Archipelago. An one-element hydrophone was towed 250 m behind the vessel R/V Atlântico Sul over the continental shelf break. Continuous mono recording was performed with the hydrophone passing signals to a digital Fostex® FR-2 LE, recording at 96 kHz/24 bits. A group of approximate 400 dolphins were recorded on June 3, 2013, at 168.9 km shore distance (27o 24'29"S, 46o50'05"W). The wav-files were analyzed through the spectrogram configured as DFT 512 samples, 50% overlap and Hamming window of 1024 points generated by software Raven Pro 1.4. The preliminary results of 10 min recording allowed the extraction of 693 whistles that were classified in contours shapes as: upsweep (42%), chirp (17.3%), downsweep (14%), sinusoidal (10.5%), convex (5.9%), constant (5.4%), and concave (4.9%). Minimum frequencies ranged from 3.32 kHz to 23.30 kHz (mean = 10.88 kHz); maximum frequencies ranged from 6.61 kHz to 35.34 kHz (mean = 15.77 kHz); whistle duration ranged from 0.03 s to 2.58 s (mean = 0.68 s). These results are important to understand populations and/or species distributed in different ocean basins.

**8:45**

**1aAB2. A new method for detection of North Atlantic right whale up-calls.** Mahdi Esfahanian, Hanqi Zhuang, and Nurgun Erdol (Comput. and Elec. Eng. and Comput. Sci., Florida Atlantic Univ., 777 Glades Rd., Bldg: EE 96, Rm. 409, Boca Raton, FL 33431, mesfahan@fau.edu)

A study of detecting North Atlantic Right Whale (NARW) up-calls has been conducted with measurements from passive acoustic monitoring devices. Denoising and normalization algorithms are applied to remove local variance and narrowband noise in order to isolate the NARW up-calls in spectrograms. The resulting spectrograms, after binarization, are treated with a region detection procedure called the Moor-Neighbor algorithm to find continuous objects that are candidates of up-call contours. After selected properties of each detected object are computed, they are compared with a pair of low and high empirical thresholds to estimate the probability of the detected object being an up-call; therefore, those objects that are determined with certainty to be non up-calls are discarded. The final stage in the proposed call detection method is to separate true up-calls from the rest of potential up-calls with classifiers such as linear discriminate analysis (LDA), Naïve Bayes, and decision tree. Experimental results using the data set obtained by Cornell University show that the proposed method can achieve accuracy to 96%.

**9:00**

**1aAB3. Spatio-temporal distribution of beaked whales in southern California waters.** Simone Baumann-Pickering, Jennifer S. Trickey (Scripps Inst. of Oceanogr., Univ. of California San Diego, 9500 Gilman Dr., La Jolla, CA 92093, sbaumann@ucsd.edu), Marie A. Roch (Dept. of Comput. Sci., San Diego State Univ., San Diego, CA), and Sean M. Wiggins (Scripps Inst. of Oceanogr., Univ. of California San Diego, La Jolla, CA)

Cuvier's beaked whales are the dominant beaked whales offshore of southern California. Their abundance, distribution, and seasonality are poorly understood. Insights on the spatio-temporal distribution of both Cuvier's and a rare beaked whale with signal type BW43, likely Perrin's beaked whale, have been derived from long-term autonomous recordings of beaked whale echolocation clicks. Acoustic recordings were collected at 18 sites offshore of southern California since 2006, resulting in a total of ~26 years of recordings. About 23,000 acoustic encounters with Cuvier's beaked whales were detected. In contrast, there were ~100 acoustic encounters of the BW43 signal type. Cuvier's beaked whales were predominantly detected at deeper, more southern, and farther offshore sites, and there appears to be a seasonal pattern to their presence, with lower probability of detection during summer and early fall. The BW43 signal type had higher detection rates in the central basins, indicating a possible difference in habitat preference and niche separation between the two species. Further investigation is needed to reveal if these distribution patterns are purely based on bathymetric preference, driven by water masses that determine prey species composition and distribution, or possibly by anthropogenic activity.

**9:15**

**1aAB4. The acoustic characteristics of greater prairie-chicken vocalizations.** Cara Whalen, Mary Bomberger Brown (School of Natural Resources, Univ. of Nebraska - Lincoln, 3310 Holdrege St., Lincoln, NE 68583, cara-whalen@gmail.com), JoAnn McGee (Developmental Auditory Physiol. Lab., Boys Town National Res. Hospital, Omaha, NE), Larkin A. Powell, Jennifer A. Smith (School of Natural Resources, Univ. of Nebraska - Lincoln, Lincoln, NE), and Edward J. Walsh (Developmental Auditory Physiol. Lab., Boys Town National Res. Hospital, Omaha, NE)

Male Greater Prairie-Chickens (*Tympanuchus cupido pinnatus*) congregate in groups known as "leks" each spring to perform vocal and visual displays to attract females. Four widely recognized vocalization types produced by males occupying leks are referred to as "booms," "cackles," "whines," and "whoops." As part of a larger effort to determine the influence of wind turbine farm noise on lek vocal behavior, we studied the acoustic properties of vocalizations recorded between March and June in 2013 and 2014 at leks near Ainsworth, Nebraska. Although all four calls are produced by males occupying leks, the boom is generally regarded as the dominant call type associated with courtship behavior. Our findings suggest that the bulk of acoustic power carried by boom vocalizations is in a relatively narrow, low frequency band, approximately 100-Hz wide at 20 dB below the peak frequency centered on approximately 0.3 kHz. The boom vocalization is harmonic in character, has a fundamental frequency of

approximately 0.30±0.01 kHz, and lasts approximately 1.81±0.18 s. Understanding Greater Prairie-Chicken vocal attributes is an essential element in the effort to understand the influence of environmental sound, prominently including anthropogenic sources like wind turbine farms, on vocal communication success.

**9:30**

**1aAB5. Bioacoustics of *Trachymyrmex fuscus*, *Trachymyrmex tucumanus,* and *Atta sexdens rubropilosa* (Hymenoptera: Formicidae).** Amanda A. Carlos, Francesca Barbero, Luca P. Cassaci, Simona Bonelli (Life Sci. and System Biology, Univ. of Turin, Dipartimento di Biologia Animale e dell'Uomo Via Accademia Albertina 13, Turin 10123, Italy, amandacarlos@yahoo.com.br), and Odair C. Bueno (Centro de Estudos de Insetos Sociais (CEIS), Universidade Estadual Paulista Júlio de Mesquita Filho (UNESP), Rio Claro, Brazil)

The capability to produce species-specific sounds is common among ants. Ants of the genus *Trachymyrmex* occur in an intermediate phylogenetic position within the Attini tribe, between the leafcutters, such as *Atta sexdens rubropilosa*, and more basal species. The study of stridulations would provide important cues on the evolution of the tribe's diverse biological aspects. Therefore, in the present study, we described the stridulation signals produced by *Trachymyrmex fuscus*, *Trachymyrmex tucumanus,* and *A. sexdens rubropilosa* workers. Ant workers were recorded, and their stridulatory organs were measured. The following parameters were analyzed: chirp length [ms], inter-chirp (pause) [ms], cycle (chirp + inter-chirp) [ms], cycle repetition rate [Hz], and the peak frequency [Hz], as well as the number of ridges on the *pars stridens*. During the inter-chirp, there is no measurable signal for *A. sexdens rubropilosa*, whereas for *Trachymyrmex fuscus* and *Trachymyrmex tucumanus*, a low intensity signal was detected. In other words, the plectrum and the *pars stridens* of *A. sexdens rubropilosa* have no contact during the lowering of the gaster. Principal component analysis, to which mainly the duration of chirps contributed, showed that stridulation is an efficient tool to differentiate ant species at least in the case of the Attini tribe.

**9:45**

**1aAB6. Robustness of perceptual features used for passive acoustic classification of cetaceans to the ocean environment.** Carolyn Binder (Oceanogr. Dept., Dalhousie Univ., LSC Ocean Wing, 1355 Oxford St., PO Box 15000, Halifax, NS B3H 4R2, Canada, carolyn.binder@dal.ca) and Paul C. Hines (Dept. of Elec. and Comput. Eng., Dalhousie Univ., Halifax, NS, Canada)

Passive acoustic monitoring (PAM) is used to study cetaceans in their habitats, which cover diverse underwater environments. It is well known that properties of the ocean environment can be markedly different between regions, which can result in distinct propagation characteristics. These can in turn lead to differences in the time-frequency characteristics of a recorded signal and may impact the accuracy of PAM systems. To develop an automatic PAM system capable of operating under numerous environmental conditions, one must account for the impact of propagation conditions. A prototype aural classifier developed at Defence R&D Canada has successfully been used for inter-species discrimination of cetaceans. The aural classifier achieves accurate results by using perceptual signal features that model the features employed by the human auditory system. The current work uses a combination of at-sea experiments and pulse propagation modeling to examine the robustness of the perceptual features with respect to propagation effects. Preliminary results will be presented from bowhead and humpback vocalizations that were transmitted over 1–20 km ranges during a two-day sea trial in the Gulf of Mexico. Insight gained from experimental results will be augmented with model results. [Work supported by the U.S. Office of Naval Research.]

**10:00–10:15 Break**

**10:15**

**1aAB7. Passive acoustic monitoring on the seasonal species composition of cetaceans from a marine observatory.** Tzu-Hao Lin, Hsin-Yi Yu (Inst. of Ecology and Evolutionary Biology, National Taiwan Univ., No. 1, Sec. 4, Roosevelt Rd., Taipei 10617, Taiwan, schonkopf@gmail.com), Chi-Fang Chen (Dept. of Eng. Sci. and Ocean Eng., National Taiwan Univ., Taipei, Taiwan), and Lien-Siang Chou (Inst. of Ecology and Evolutionary Biology, National Taiwan Univ., Taipei, Taiwan)

Information on the species diversity of cetaceans can help us to understand the community ecology of marine top predators. Passive acoustic monitoring has been widely applied in the cetacean research, however, species identification based on tonal sounds remains challenging. In order to examine the seasonal changing pattern of species diversity, we applied an automatic detection and classification algorithm on acoustic recordings collected from the marine cable hosted observatory off the northeastern Taiwan. Representative frequencies of cetacean tonal sounds were detected. Statistical features were extracted based on the distribution of representative frequency and were used to classify four cetacean groups. The correct classification rate was 72.2% based on the field recordings collected from onboard surveys. Analysis on one-year recordings revealed that the species diversity was highest in winter and spring. Short finned pilot whales and Risso's dolphins were the most common species, they mainly occurred in winter and summer. False killer whales were mostly detected in winter and spring. Spinner dolphins, spotted dolphins, and Fraser's dolphins were mainly detected in summer. Bottlenose dolphins represent the least common species. In the future, the biodiversity, species-specific habitat use, and inter-specific interaction of cetaceans can be investigated through an underwater acoustic monitoring network.

**10:30**

**1aAB8. The effects of road noise on the calling behavior of Pacific chorus frogs.** Danielle V. Nelson (Dept. of Forest Ecosystems and Society, Oregon State Univ., Oregon State University, 321 Richardson Hall, Corvallis, OR 97331, danielle.nelson@oregonstate.edu), Holger Klinck (Fisheries and Wildlife, Oregon State Univ., Newport, OR), and Tiffany S. Garcia (Fisheries and Wildlife, Oregon State Univ., Corvallis, OR)

Fitness consequences of anthropogenic noise on organisms that have chorus-dependent breeding requirements, such as frogs, are not well understood. While frogs were thought to have innate and fixed call structure, species-specific vocal plasticity has been observed in populations experiencing high noise conditions. Adjustment to call structure, however, can have negative fitness implications in terms of energy expenditure and female choice. The Pacific chorus frog (*Pseudacris regilla*), a common vocal species broadly distributed throughout the Pacific Northwest, often breeds in waters impacted by road noise. We compared Pacific chorus frog call structure from breeding populations at 11 high- and low-traffic sites in the Willamette Valley, Oregon. We used passive acoustic monitoring and directional recordings to determine mean dominant frequency, amplitude, and call rate of breeding populations, individual frogs, and to quantify ambient road noise levels. Preliminary results indicate that while individuals do not differ in call rate or structure across noisy and quiet sites, high road noise levels decrease the effective communication distance of both the chorus and the individual. This research enhances our understanding of acoustic habitat in the Willamette Valley and the impacts of anthropogenic noise on a native amphibian species.

**10:45**

**1aAB9. Inter-individual difference of one type of pulsed sounds produced by beluga whales (*Delphinapterus leucas*).** Yuka Mishima (Tokyo Univ. of Marine Sci. and Technol., Konan 4-5-7, Minato-ku, Tokyo 108-8477, Japan, thank_you_for_email_5yuka@yahoo.co.jp), Tadamichi Morisaka (Tokai Univ. Inst. of Innovative Sci. and Technol., Shizuoka-shi, Japan), Miho Itoh (The Port of Nagoya Public Aquarium, Nagoya-shi, Japan), Ryota Suzuki, Kenji Okutsu (Yokohama Hakkeijima Sea Paradise, Yokohama-shi, Japan), Aiko Sakaguchi, and Yoshinori Miyamoto (Tokyo Univ. of Marine Sci. and Technol., Minato-ku, Japan)

Belugas often exchange one type of broadband pulsed sounds (termed PS1 calls) which possibly functions as a contact calls (Morisaka *et al.*,

2013). Here we investigate how belugas embed their signature information into the PS1 calls. PS1 calls were recorded from each of five belugas including both sexes and various ages at the Port of Nagoya Public Aquarium using a broadband recording system when in isolation. Temporal and spectral acoustic parameters of PS1 calls were measured and compared among individuals. Kruskal-Wallis test revealed that inter-pulse intervals (IPIs), the number of pulses, and pulse rates of PS1 calls had significant differences among individuals, but duration did not ($\chi 2=76.7$, $p<0.0001$; $\chi 2=26.2$, $p<0.0001$; $\chi 2=45.3$, $p<0.0001$; and $\chi 2=4.7$, $p=0.316$ respectively). The contours depicted by the IPIs as a function of pulse order were also individually different and only the contours of a calf fluctuated over time. Four belugas except a juvenile had individually distinctive power spectra. These results suggest that several acoustic parameters of PS1 calls may hold individual information. We found PS1-like calls from the other captive belugas (Yokohama Hakkeijima Sea Paradise) suggested that the PS1 call is not the specific call for one captive population but the basic call type for belugas.

**11:00**

**1aAB10. Numerical study of biosonar beam forming in finless porpoise (*Neophocaena asiaeorientalis*).** Chong Wei (College of Ocean & Earth Sci., Xiamen Univ., 1502 Spreckels St. Apt 302A, Honolulu, Hawaii 96822, weichong3310@foxmail.com), Zhitao Wang (Key Lab. of Aquatic Biodiversity and Conservation of the Chinese Acad. of Sci., Inst. of Hydrobiology of the Chinese Acad. of Sci., Wuhan, China), Zhongchang Song (College of Ocean & Earth Sci., Xiamen Univ., Xiamen, China), Whitlow Au (Hawaii Inst. of Marine Biology, Univ. of Hawaii at Manoa, Kaneohe, HI), Ding Wang (Key Lab. of Aquatic Biodiversity and Conservation of the Chinese Acad. of Sci., Inst. of Hydrobiology of the Chinese Acad. of Sci., Wuhan, China), and Yu Zhang (Key Lab. of Underwater Acoust. Commun. and Marine Information Technol. of the Ministry of Education, Xiamen Univ., Xiamen, China)

Finless porpoise (*Neophocaena asiaeorientalis*) is known to use the narrow band signals for echolocation living in the Yangtze River and in the adjoining Poyang and Dongting Lakes in China. In this study, the sound velocity and density of different tissues (including melon, muscle, bony structure, connective tissues, blubber, and mandibular fat) in the porpoise's head were obtained by measurement. The sound velocity and density were found out to have a linear relationship with Hounsfield unit (HU) obtained from the CT scan. The acoustic property of the head of the porpoise was reconstructed from the HU distribution. Numerical simulations of the acoustic propagation through finless porpoise's head were performed by a finite element approach. The beam formation was compared with those of the baiji, Indo-pacific humpback dolphin, and bottlenose dolphin. The role of the different structures in the head such as air sacs, melon, muscle, bony structure, connective tissues, blubber, and mandibular fat on biosonar beam was investigated. The results might provide useful information for better understanding of the sound propagation in finless porpoise's head.

**11:15**

**1aAB11. Evidence for a possible functional significance of horseshoe bat biosonar dynamics.** Rolf Müller, Anupam K. Gupta (Mech. Eng., Virginia Tech, 1075 Life Sci. Cir, Blacksburg, VA 24061, rolf.mueller@vt.edu), Uzair Gillani (Elec. and Comput. Eng., Virginia Tech, Blacksburg, VA), Yanqing Fu (Eng. Sci. and Mech., Virginia Tech, Blacksburg, VA), and Hongxiao Zhu (Dept. of Statistics, Virginia Tech, Blacksburg, VA)

The periphery of the biosonar system of horseshoe bats is characterized by a conspicuous dynamics where the shapes of the noseleaves (structures that surround the nostrils) and the outer ears (pinnae) undergo fast changes that can coincide with pulse emission and echo reception. These changes in the geometries of the sound-reflecting surfaces affect the device characteristics, e.g., as represented by beampatterns. Hence, this dynamics could give horseshoe bats an opportunity to view their environments through a set of different device characteristics. It is not clear at present whether horseshoe bats make use of this opportunity, but there is evidence from various sources, namely, anatomy, behavior, evolution, and information theory. Anatomical studies have shown the existence of specialized muscular actuation systems that are clearly directed toward geometrical changes. Behavioral observations indicate that these changes are linked to contexts where the bat is confronted with a novel or otherwise demanding situation. Evolutionary evidence comes from the occurrence of qualitatively similar ear deformation patterns in mustached bats (*Pteronotus*) that have independently evolved a biosonar for Doppler-shift detection. Finally, an information-theoretic analysis demonstrates that the capacity of the biosonar system for encoding sensory information is enhanced by these dynamic processes.

**11:30**

**1aAB12. Analysis of some special buzz clicks.** Odile Gerard (DGA, Ave. de la Tour Royale, Toulon 83000, France, odigea@gmail.com), Craig Carthel, and Stefano Coraluppi (Systems & Technol. Res., Woburn, MA)

Toothed whales are known to click regularly to find prey. Once a prey has been detected, the repetition rate of the clicks increases; these sequences are called buzzes. Previous work shows that the buzz clicks spectrum slowly varies from click to click for various species. This spectrum similarity allows buzz clicks association as a sequence using multi-hypothesis tracking (MHT) algorithms. Thus buzz classification follows automatic click tracking. The use of MHT reveals that in some rare cases a variant of this property has been found, whereby sub-sequences of clicks exhibit slowly varying characteristics. In 2010 and 2011, NATO Undersea Research Centre (NURC, now CMRE Centre for Maritime Research and Experimentation) conducted sea-trials with the CPAM (compact Passive Acoustic Monitoring), a volumetric towed array comprised of four or six hydrophones. This configuration allows for a rough estimate of clicking animal localization. Some buzzes with sub-sequences of slowly varying characteristics were recorded with the CPAM. Localization may help to understand this new finding from a physiological point of view. The results of this analysis will be presented.

## Session 1aNS

## Noise, Physical Acoustics, Structural Acoustics and Vibration, and Engineering Acoustics: Metamaterials for Noise Control I

Keith Attenborough, Cochair
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Olga Umnova, Cochair
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**Chair's Introduction—7:55**

### *Invited Papers*

**8:00**

**1aNS1. Recent results on sonic crystals for sound guiding and acoustic absorption.** Jose Sanchez-Dehesa, Victor M. García-Chocano, and Matthew D. Guild (Dept. of Electron. Eng., Universitat Politecnica de Valencia, Camino de vera s.n., Edificio 7F, Valencia, Valencia ES-46022, Spain, jsdehesa@upv.es)

We report on different aspects of the behavior of sonic crystals with finite size. First, at wavelengths on the order of the lattice period we have observed the excitation of deaf modes, i.e., modes with symmetry orthogonal to that of the exciting beam. Numerical simulations and experiments performed with samples made of three rows of cylindrical scatterers demonstrate the excitation of sound waves guided along a direction perpendicular to the incident beam. Moreover, the wave propagation inside the sonic crystal is strongly dependent on the porosity of the building units. This finding can be used to enhance the absorbing properties of the crystal. Also, we will discuss the properties of finite sonic crystals at low frequencies, where we have observed small period oscillations superimposed on the well-known Fabry-Perot resonances appearing in the reflectance and transmittance spectra. It will be shown that the additional oscillations are due to diffraction in combination with the excitation of the transverse modes associated with the finite size of the samples. [Work supported by ONR.]

**8:20**

**1aNS2. Acoustic metamaterial absorbers based on multi-scale sonic crystals.** Matthew D. Guild, Victor M. García-Chocano (Dept. of Electronics Eng., Universitat Politecnica de Valencia, Camino de vera s/n (Edificio 7F), Valencia 46022, Spain, mdguild@utexas.edu), Weiwei Kan (Dept. of Phys., Nanjing Univ., Nanjing, China), and Jose Sanchez-Dehesa (Dept. of Electronics Eng., Universitat Politecnica de Valencia, Valencia, Spain)

In this work, thermoviscous losses in single- and multi-scale sonic crystal arrangements are examined, enabling the fabrication and characterization of acoustic metamaterial absorbers. It will be shown that higher filling fraction arrangements can be used to provide a large enhancement in the complex mass density and loss factor, and can be combined with other sonic crystals of different sizes to create multi-scale structures that further enhance these effects. To realize these enhanced properties, different sonic crystal lattices are examined and arranged as a layered structure or a slab with large embedded inclusions. The inclusions are made from either a single solid cylinder or symmetrically arranged clusters of cylinders, known as magic clusters, which behave as an effective fluid. Theoretical results are obtained using a two-step homogenization process, by first homogenizing each sonic crystal to obtain the complex effective properties of each length scale, and then homogenizing the effective fluid structures to determine the properties of the ensemble structure. Experimental data from acoustic impedance tube measurements will be presented and shown to be in excellent agreement with the expected results. [Work supported by the US ONR and Spanish MINECO.]

**8:40**

**1aNS3. Quasi-flat acoustic absorber enhanced by metamaterials.** Abdelhalim Azbaid El Ouahabi, Victor V. Krylov, and Daniel J. O'Boy (Dept. of Aeronautical and Automotive Eng., Loughborough Univ., Loughborough University, Loughborough, Leicestershire LE11 3TU, United Kingdom, A.Azbaid-El-Ouahabi@lboro.ac.uk)

In this paper, the design of a new quasi-flat acoustic absorber (QFAA) enhanced by the presence of a graded metamaterial layer is described, and the results of the experimental investigation into the reflection of sound from such an absorber are reported. The matching metamaterial layer is formed by a quasi-periodic array of brass cylindrical tubes with the diameters gradually increasing from the external row of tubes facing the open air towards the internal row facing the absorbing layer made of a porous material. The QFAA is placed in a wooden box with the dimensions of $569 \times 250 \times 305$ mm. All brass tubes are of the same length (305 mm) and fixed between the opposite sides of the wooden box. Measurements of the sound reflection coefficients from the empty wooden box, from the box with an inserted porous absorbing layer, and from the full QFAA containing both the porous absorbing layer and the array of brass tubes have

been carried out in an anechoic chamber at the frequency range of 500–3000 Hz. The results show that the presence of the metamaterial layer brings a noticeable reduction in the sound reflection coefficients in comparison with the reflection from the porous layer alone.



**9:00**

**1aNS4. The influence of thermal and viscous effects on the effective properties of an array of slits.** John D. Smith (Physical Sci., DSTL, Porton Down, Salisbury SP4 0JQ, United Kingdom, jdsmith@dstl.gov.uk), Roy Sambles, Gareth P. Ward, and Alastair R. Murray (Dept. of Phys. and Astronomy, Univ. of Exeter, Exeter, United Kingdom)

A system consisting of an array of thin plates separated by air gaps is examined using the method of asymptotic homogenization. The effective properties are compared with a finite element model and experimental results for the resonant transmission of both a single slit and an array of slits. These results show a dramatic reduction in the frequency of resonant transmission when the slit is narrowed to below around one percent of the wavelength due to viscous and thermal effects reducing the effective sound velocity through the slits. These effects are still significant for slit widths substantially greater than the thickness of the boundary layer.

**9:20**

**1aNS5. Atypical dynamic behavior of periodic frame structures with local resonance.** Stéphane Hans, Claude Boutin (LGCB / LTDS, ENTPE / Université de Lyon, rue Maurice Audin, Vaulx-en-Velin 69120, France, stephane.hans@entpe.fr), and Céline Chesnais (IFSTTAR GER, Université Paris-Est, Paris, France)

This work investigates the dynamic behavior of periodic unbraced frame structures made up of interconnected beams or plates. Such structures can represent an idealization of numerous reticulated systems, as the microstructure of foams, plants, bones, the sandwich panels. As beams are much stiffer in tension-compression than in bending, the propagation of waves with wavelengths much greater than the cell size and the bending modes of the elements can occur in the same frequency range. Thus, frame structures can behave as metamaterials. Since the condition of scale separation is respected, the homogenization method of periodic discrete media is used to derive the macroscopic behavior. The main advantages of the method are the analytical formulation and the possibility to study the behavior of the elements at local scale. This provides a clear understanding of the mechanisms governing the dynamics of the material. In the presence of the local resonance, the form of the equations is unchanged but some macroscopic parameters depend on the frequency. In particular, this applies to the mass leading to a generalization of the Newtonian mechanics. As a result, there are frequency bandgaps. In that case, the same macroscopic modal shape is also associated with several resonant frequencies.

**9:40**

**1aNS6. Design of sound absorbing metamaterials by periodically embedding three-dimensional resonant or non-resonant inclusions in rigidly backed porous plate.** Jean-Philippe Groby (LAUM, UMR6613 CNRS, LAUM, UMR 6613 CNRS, AV. Olivier Messiaen, Le Mans F-72085, France, Jean-Philippe.Groby@univ-lemans.fr), Benoit Nennig (LISMMA, Supmeca, Saint Ouen, France), Clément Lagarrigue, Brunuo Brouard, Dazel Olivier (LAUM, UMR6613 CNRS, Le Mans, France), Olga Umnova (Acoust. Res. Ctr., Univ. of Salford, Salford, United Kingdom), and Vincent Tournat (LAUM, UMR6613 CNRS, Le Mans, France)

Air saturated porous materials, namely, foams and wools, are often used as sound absorbing materials. Nevertheless, they suffer from a lack of absorption efficiency at low frequencies, which is inherent to their absorption mechanisms (viscous and thermal losses), even when used as optimized multilayer or graded porous materials. These last decades, several solutions have been proposed to avoid this problem. Among them, metaporous materials consist in exciting modes trapping the energy between the periodic rigid inclusions embedded in the porous plate and the rigid backing or in the inclusions themselves. The absorption coefficient of different foams is enhanced both in the viscous and inertial regimes by periodically embedding 3D inclusions, possibly resonant, i.e., air filled Helmholtz resonators. This enhancement is due to different mode excitation: a Helmholtz resonance in the viscous regime and a trap mode in the inertial regime. In particular, a large absorption coefficient is reached for wavelengths in the air 27 times larger than the sample thickness. The absorption amplitude and bandwidth is then enlarged by removing porous material in front of the neck, enabling a lower impedance radiation, and by adjusting the resonance frequencies of the Helmholtz resonator.

**10:00–10:20 Break**

**10:20**

**1aNS7. Seismic metamaterials: Shielding and focusing surface elastic waves in structured soils.** Sebastien R. Guenneau, Stefan Enoch (Phys., Institut Fresnel, Ave. Escadrille Normandie Niemen, Marseille 13013, France, sebastien.guenneau@fresnel.fr), and Stephane Brule (Menard Co., Nozay, France)

Phononic crystals and metamaterials are man-made structures (with periodic heterogeneities typically a few micrometers to centimeters) that can control sound in ways not found in nature. Whereas the properties of phononic crystals derive from the periodicity of their structure, those of metamaterials arise from the collective effect of a large array of small resonators. These effects can be used to manipulate acoustic waves in unconventional ways, realizing functions such as invisibility cloaking, subwavelength focusing, and unconventional refraction phenomena (such as negative refractive index and phase velocity). Recent work has started to explore another intriguing domain of application: using similar concepts to control the propagation of seismic waves within the surface of the Earth. Our research group at the Aix-Marseille University and French National Center for Scientific Research (CNRS) has teamed up with civil engineers at an industrial company, Ménard, in Nozay, also in France, and carried out the largest-scale tests to date of phononic crystals. Arrays of boreholes in soil which are a few centimeters to a few meters in diameter are encouraging, thereafter called seismic metamaterials, can be used to deflect incoming acoustic waves at a frequency relevant to earthquake protection, or bring them to a focus. These preliminary successes could one day translate into a way of mitigating the destructive effects of earthquakes.

10:40

**1aNS8. Tunable resonator arrays—Transmission, near-field interactions, and effective property extraction.** Dmitry Smirnov and Olga Umnova (Acoust. Res. Ctr., Univ. of Salford, The Crescent, Salford, Greater Manchester m5 4wt, United Kingdom, d.smirnov@edu.salford.ac.uk)

Periodic arrays of slotted cylinders have been studied with a focus on analytical and semi-analytical techniques, observing near-field interactions and their influence on reflection and transmission of acoustic waves by the array. Relative orientation of the cylinders within a unit cell has been shown to strongly affect the array behavior, facilitating tunable transmission gaps. An improved homogenization method is proposed and used to determine effective properties of the array, allowing accurate and computationally efficient prediction of reflection and transmission characteristics of any number of rows at arbitrary incidence.

11:00

**1aNS9. Tunable cylinders for sound control in water.** Andrew Norris and Alexey Titovich (Mech. and Aerosp. Eng., Rutgers Univ., 98 Brett Rd., Piscataway, NJ 08854, norris@rutgers.edu)

Long wavelength effective medium properties are achieved using arrays of closely spaced tunable cylinders. Thin metal shells provide the starting point: for a given shell thickness h and radius a, the effective bulk modulus and density are both proportional to h/a. Since the metal has large impedance relative to water it follows that there is a unique value of h/a at which the shell is effectively impedance matched to water. The effective sound speed cannot be matched by the thin shell alone (except for impractical metals like silver). However, simultaneous impedance and speed matching can be obtained by adding an internal mass, e.g., an acrylic core in aluminum cylindrical tubes. By varying the shell thickness and the internal mass, a range of effective properties is achievable. Practical considerations such as shell thickness, internal mass material, and fabrication will be discussed. Arrays made of a small number of different tuned shells will be described using numerical simulations: example applications include focusing, lensing, and wave steering. [Work supported by ONR.]

11:20

**1aNS10. Sound waves over periodic and aperiodic arrays of cylinders on ground surfaces.** Shahram Taherzadeh, Ho-Chul Shin, and Keith Attenborough (Eng. & Innovation, The Open Univ., Walton Hall, Milton Keynes MK7 6AA, United Kingdom, shahram.taherzadeh@open.ac.uk)

Propagation of audio frequency sound waves over periodic arrays of cylinders placed on acoustically hard and soft surfaces has been studied through laboratory measurements and predictions using a point source. It is found that perturbation of the position of the cylinders from a regular array results in a higher insertion loss than completely periodic or random cylinder arrangements.

11:40

**1aNS11. Ground effect due to rough and resonant surfaces.** Keith Attenborough (Eng. and Innovation, Open Univ., 18 Milebush, Linslade, Leighton Buzzard, Bedfordshire LU7 2UB, United Kingdom, Keith.Attenborough@open.ac.uk), Ho-Chul Shin, and Shahram Taherzadeh (Eng. and Innovation, Open Univ., Milton Keynes, United Kingdom)

Particularly if the ground surface between noise source and receiver would otherwise be smooth and acoustically hard, structured low-rise ground roughness can be used as an alternative to conventional noise barriers. The techniques of periodic-spacing, absorptive covering, and local resonance can be used, as when broadening metamaterial stop bands, to achieve a broadband ground effect. This has been demonstrated both numerically and through laboratory experiments. Computations have employed multiple scattering theory, the Finite Element Method and the Boundary Element Method. The experiments have involved measurements over cylindrical and rectangular roughness elements and over their resonant counterparts created by incorporating slit-like openings. Resonant elements with slit openings have been found numerically and experimentally to add a destructive interference below the first roughness-induced destructive interference and thereby mitigate the adverse effects of the low-frequency surface waves generated by the presence of roughness elements. A nested configuration of slotted hollow roughness elements is predicted to produce multiple resonances and this idea has been validated through laboratory experiments.

## Session 1aPA

## Physical Acoustics and Noise: Jet Noise Measurements and Analyses I

Richard L. McKinley, Cochair
*Battlespace Acoustics, Air Force Research Laboratory, 2610 Seventh Street, Wright-Patterson AFB, OH 45433-7901*

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

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

**Chair's Introduction—8:15**

### *Invited Papers*

**8:20**

**1aPA1. F-35A and F-35B aircraft ground run-up acoustic emissions.** Michael M. James, Micah Downing, Alexandria R. Salton (Blue Ridge Res. and Consulting, 29 N Market St., Ste. 700, Asheville, NC 28801, michael.james@blueridgeresearch.com), Kent L. Gee, Tracianne B. Neilsen (Phys. and Astronomy, Brigham Young Univ., Provo, UT), Richard L. McKinley, Alan T. Wall, and Hilary L. Gallagher (Air Force Res. Lab., Dayton, OH)

A multi-organizational effort led by the Air Force Research Laboratory conducted acoustic emissions measurements on the F-35A and F 35B aircraft at Edwards Air Force Base, California, in September 2013. These measurements followed American National Standards Institute/Acoustical Society of America S12.75-2012 to collect noise data for community noise models and noise exposures to aircraft personnel. In total, over 200 unique locations were measured with over 300 high fidelity microphones. Multiple microphone arrays were deployed in three orientations: circular arcs, linear offsets from the jet-axis centerline, and linear offsets from the jet shear layer. The microphone arrays ranged from distances 10 ft outside the shear layer to 4000 ft from the aircraft with angular positions ranging from 0° (aircraft nose) to 160° (edge of the exhaust flow field). A description of the ground run-up acoustic measurements, data processing, and the resultant data set is provided.

**8:40**

**1aPA2. Measurement of acoustic emissions from F-35B vertical landing operations.** Micah Downing, Michael James (Blue Ridge Res. and Consulting, 29 N. Market St., Ste. 700, Asheville, NC 28801, micah.downing@blueridgeresearch.com), Kent Gee, Brent Reichman (Brigham Young Univ., Provo, UT), Richard McKinley (Air Force Res. Lab., Wright-Patterson AFB, OH), and Allan Aubert (Naval Air Warfare Ctr., Patuxent River, MD)

A multi-organizational effort led by the Air Force Research Laboratory conducted acoustic emissions measurements from vertical landing operations of the F-35B aircraft at Marine Corps Air Station Yuma, Arizona, in September 2013. These measurements followed American National Standards Institute/Acoustical Society of American S12.75-2012 to collect noise data from vertical landing operations for community noise models and noise exposures to aircraft personnel. Three circular arcs and two vertical microphone arrays were deployed for these measurements. The circular microphone arrays ranged from distances from 250 ft to 1000 ft from touch down point. A description of the vertical landing acoustic measurements, data processing, preliminary data analysis, the resultant dataset, and a summary of results will be provided.

**9:00**

**1aPA3. Acoustic emissions from flyover measurements of F-35A and F-35B aircraft.** Richard L. McKinley, Alan T. Wall, Hilary L. Gallagher (Battlespace Acoust. Branch, Air Force Res. Lab., 711 HPW/RHCB, 2610 Seventh St., Bldg 441, Wright-Patterson AFB, OH, richard.mckinley.1@us.af.mil), Christopher M. Hobbs, Juliet A. Page, and Joseph J. Czech (Wyle Labs., Inc., Arlington, VA)

Acoustic emissions of F-35A and F-35B aircraft flyovers were measured in September 2013, in a multi-organizational effort led by the Air Force Research Laboratory. These measurements followed American National Standards Institute/Acoustical Society of America S12.75-2012 guidance on aircraft flyover noise measurements. Measurements were made from locations directly under the flight path to 12,000 ft away with microphones on the ground, 5 ft, and 30 ft high. Vertical microphone arrays suspended from cranes measured noise from on the ground up to 300 ft above the ground. A linear ground-based microphone array measured noise directly along the flight path. In total, data were collected at more than 100 unique locations. Measurements were repeated six times for each flyover condition. Preliminary results are presented to demonstrate the repeatability of noise data over measurement repetitions, assess data quality, and quantify community noise exposure models.

**9:20**

**1aPA4. Three-stream jet noise measurements and predictions.** Brenda S. Henderson (Acoust., NASA, MS 54-3, 21000 Brookpark Rd., Cleveland, OH 44135, brenda.s.henderson@nasa.gov) and Stewart J. Leib (Ohio Aerosp. Inst., Cleveland, OH)

An experimental and numerical investigation of the noise produced by high-subsonic three-stream jets was conducted. The exhaust system consisted of externally mixed-convergent nozzles and an external plug. Bypass- and tertiary-to-core area ratios between 1 and 1.75, and 0.4 and 1.0, respectively, were studied. Axisymmetric and offset tertiary nozzles were investigated for heated and unheated conditions. For axisymmetric configurations, the addition of the third stream was found to reduce mid- and high-frequency acoustic levels in the peak-jet-noise direction, with greater reductions at the lower bypass-to-core area ratios. The addition of the third stream also decreased peak acoustic levels in the peak-jet-noise direction for intermediate bypass-to-core area ratios. For the offset configurations, an s-duct was found to increase acoustic levels relative to those of the equivalent axisymmetric-three-stream jet while half-duct configurations produced acoustic levels similar to those for the axisymmetric jet for azimuthal observation locations of interest. Comparisons of noise predictions with acoustic data are presented for selected unheated configurations. The predictions are based on an acoustic analogy approach with mean flow interaction effects accounted for using a Green's function, computed in terms of its coupled azimuthal modes, and a source model previously used for round and rectangular jets.

**9:40**

**1aPA5. Acoustic interaction of turbofan exhaust with deflected control surface for blended wing body airplane.** Dimitri Papamoschou (Mech. and Aerosp. Eng., Univ. of California, Irvine, 4200 Eng. Gateway, Irvine, CA 92697-3975, dpapamos@uci.edu) and Salvador Mayoral (Mech. and Aerosp. Eng., Univ. of California, Irvine, Irvine, Armed Forces Pacific)

Small-scale experiments simulated the elevon-induced jet scrubbing noise of the Blended-Wing-Body platform with a bypass ratio ten turbofan nozzle installed above the wing. The elevon chord length at the interaction zone was similar to the exit fan diameter of the nozzle. The study encompassed variable nozzle position, variable elevon deflection, removable inboard fins, and two types of nozzles—plain and chevron. Far-field microphone surveys were conducted underneath the wing. The interaction between the jet and the elevon produces excess noise that intensifies with increasing elevon deflection. When the elevon trailing edge is near the edge of the jet, excess noise is manifested as a low-frequency bump on the sound pressure level spectrum. An empirical model for this excess noise is presented. The interaction noise becomes severe, and elevates the entire spectrum, when the elevon intrudes significantly into the jet flow. The increase in effective perceived noise level (EPNL) falls on well-defined trends when correlated versus the penetration of the elevon trailing edge into the flow field of the isolated jet. The cumulative takeoff EPNL can increase by as much as 19 dB, underscoring the potentially detrimental effects of jet-elevon interaction on noise compliance.

**10:00–10:20 Break**

**10:20**

**1aPA6. Comparison of upside-down microphone with flush mounted microphone configuration.** Per Rasmussen (G.R.A.S. Sound & Vib. A/S, Skovlytoften 33, Holte 2840, Denmark, pr@gras.dk)

Measurement of fly-over aircraft noise is often performed using the microphones mounted in an upside-down configuration, with the microphone placed 7 mm above a hard reflecting surface. This method assumes that most of the sound is coming from the back of the microphone within an angle of + -60 degrees. The same microphone configuration is proposed for installed and un-installed jet-engine test in which case, however, the incidence angle for the microphone may be in the range of 60–85 degrees. The response of the upside-down microphone configuration is compared with flush mounted microphones as reference. The influence of microphone diameter (ranging from 1/8 in. to 1/2 in.) is compared in the different configurations and the effect of windscreens is investigated.

**10:40**

**1aPA7. Active control of noise from hot, supersonic turbulent jets.** Tim Colonius, Aaron Towne (Mech. Eng., Caltech, 1200 E. California Blvd., Pasadena, CA 91125, colonius@caltech.edu), Robert H. Schlinker, Ramons A. Reba, and Dan Shannon (Thermal and Fluid Sci. Dept., United Technologies Res. Ctr., East Hartford, CT)

We report on an experimental and reduced-order modeling study aimed at reducing mixing noise in hot supersonic jets relevant to military aircraft. A spinning valve is used to modulate four injection nozzles near the main jet nozzle lip over a range of frequencies and mass flow rates. Diagnostics include near-, mid-, and far-field microphone arrays aimed at measuring the effect of actuation on the near-field turbulent wavepacket structures and their correlation with mixing noise. The actuators provide more than 4 dB noise reduction at peak frequencies in the aft arc, and up to 2 dB reduction in OASPL. Experiments are performed to contrast the performance of steady and unsteady blowing with different amplitudes. The results to date suggest that the noise reduction is primarily associated with attenuated wave packet activity associated with the rapidly thickened shear layers that occur with both steady and unsteady blowing. Mean flow surveys are also preformed and serve as inputs to reduced-order models for the wave packets based on parabolized stability equations. These models are in turn used to corroborate the experimental evidence suggesting mechanisms of noise suppression in the actuated flow.

## *Contributed Papers*



**11:00**

**1aPA8. Efficient jet noise models using the one-way Euler equations.**
Aaron Towne and Tim Colonius (Dept. of Mech. and Civil Eng., California Inst. of Technol., 1200 E California Blvd., MC 107-81, Pasadena, CA 91125, atowne@caltech.edu)

Experimental and numerical investigations have correlated large-scale coherent structures in turbulent jets with acoustic radiation to downstream angles, where sound is most intense. These structures can be modeled as linear instability modes of the turbulent mean flow. The parabolized stability equations have been successfully used to estimate the near-field evolution of these modes, but are unable to properly capture the acoustic field. We have recently developed an efficient method for calculating these linear modes that properly captures the acoustic field. The linearized Euler equations are modified such that all upstream propagating acoustic modes are removed from the operator. The resulting equations, called one-way Euler equations, can be stably and efficiently solved in the frequency domain as a spatial initial value problem in which initial perturbations are specified at the flow inlet and propagated downstream by integrating the equations. We demonstrate the accuracy and efficiency of the method by using it to model sound generation and propagation in jets. The results are compared to accurate large-eddy-simulation data for both subsonic and supersonic jets.

**11:15**

**1aPA9. A new method of estimating acoustic intensity applied to the sound field near a military jet aircraft.**
Trevor A. Stout, Kent L. Gee, Tracianne B. Neilsen, Derek C. Thomas, Benjamin Y. Christensen (Phys. and Astronomy, Brigham Young Univ., 688 north 500 East, Provo, UT 84606, tstout@byu.edu), and Michael M. James (Blue Ridge Res. and Consulting LLC, Asheville, NC)

Intensity probes are traditionally made up of closely spaced microphones, with the finite-difference method used to approximate acoustic intensity. This approximation is not reliable approaching the Nyquist frequency limit determined by microphone spacing. However, the new phase and amplitude estimation (PAGE) method allows for accurate intensity approximation far above this limit. The PAGE method is applied to measurements from a three-dimensional intensity probe, which took data to the sideline and aft of a tethered F-22A Raptor. It is shown that the PAGE method produces physically meaningful intensity approximations for frequencies up to about 6 kHz, while the finite-difference method is only reliable up to about 2 kHz. [Work supported by ONR.]

**11:30**

**1aPA10. Three transformations of a crackling jet noise waveform and their potential implications for quantifying the "crackle" percept.**
S. Hales Swift (School of Aeronautics and Astronautics, Purdue Univ., 2286 Yeager Rd., West Lafayette, IN 47906, hales.swift@gmail.com), Kent L. Gee, and Tracianne B. Neilsen (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT)

In the 1975 paper by Ffowcs-Williams *et al.* on jet "crackle," there are several potentially competing descriptors—including a qualitative description of the sound quality or percept, a statistical measure, and commentary on the relation of the presence of shocks to the sound's quality. These descriptors have led to disparate conclusions about what constitutes a crackling jet, waveform, or sound quality. This presentation considers three modifications of a jet noise waveform that exhibits a crackling sound quality and initially satisfies all three definitions. These modifications alter the statistical distributions of primarily the pressure waveform or its first time difference in order to demonstrate how these modifications do or do not correspond to changes in the sound quality of the waveform. The result, although preliminary, demonstrates that the crackle percept is tied to the statistics of the pressure difference waveform instead of the pressure waveform itself.

MONDAY MORNING, 27 OCTOBER 2014 — MARRIOTT 5, 9:30 A.M. TO 12:00 NOON

# Session 1aSC

## Speech Communication: Speech Processing and Technology (Poster Session)

Michael Kiefte, Chair
*Human Communication Disorders, Dalhousie University, 1256 Barrington St., Halifax, NS B3J 1Y6, Canada*

All posters will be on display from 9:30 a.m. to 12:00 noon. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 9:30 a.m. to 10:45 a.m. and contributors of even-numbered papers will be at their posters from 10:45 a.m. to 12:00 noon.

## *Contributed Papers*

**1aSC1. Locus equations estimated form a corpus of running speech.**
Michael Kiefte (Human Commun. Disord., Dalhousie Univ., 1256 Barrington St., Halifax, NS B3J 1Y6, Canada, mkiefte@dal.ca) and Terrance M. Nearey (Linguist, Univ. of AB, Edmonton, NS, Canada)

Locus equations, or the linear relationship between onset and vowel second-formant frequency F2 in terms of slope and y-intercept, have been presented as possible invariant correlates to consonant place of articulation [e.g., Sussman *et al.* (1998). Behav. Brain Sci. 21, 241–299]. In the current study, formant measurements were extracted from both stressed and unstressed vowels taken from a database of spontaneous and read speech. Locus equations were estimated for several places of articulation of the preceding consonant. In addition, optimal time frames for estimating locus equations are determined with reference to automatic classification of consonant place of articulation as well as vowel identification. Formant frequencies are first measured at multiple time frames—both before and after voicing onset in the case of voiceless plosives—to find the pair of time frames that best estimates place of articulation via discriminant analysis and

other classification methods. In addition, locus-equation slopes are compared between stressed and unstressed vowels as well as between spontaneous and read speech samples. In addition, the role of total vowel duration across these contexts is described. The evaluation of several strategies for optimizing the automatic extraction of formant frequencies from running speech are also reported. [Work supported by SSHRC.]

**1aSC2. Formant trajectory analysis using dynamic time warping: Preliminary results.** Kirsten T. Regier (Linguist, Indiana Univ., 3201 W Woodbridge Dr., Muncie, IN 47304, krtodt@indiana.edu)

In English, there are at least two mechanisms that affect vowel duration—vowel identity and postvocalic consonant voicing. Previous studies have shown that these two mechanisms have independent effects on vowel duration (Port 1981, Todt 2010). This study presents preliminary results on the use of dynamic time warping to distinguish between the effects of vowel identity and postvocalic consonant voicing on the formant trajectories of English front vowels. Using PraatR (Albin 2014), formant trajectories are extracted from sound files in Praat and imported into R, where the dynamic time warping analysis is conducted using the dtw package (Giorgino 2009). Albin, A. L. (2014). PraatR: An architecture for controlling the phonetics software "Praat" with the R programming language. JASA 135, 2198. Giorgino T. (2009). "Computing and Visualizing Dynamic Time Warping Alignments in R: The dtw Package,"J. Stat. Software, 31(7), pp. 1–24. Port, R. F. (1981). Linguistic timing factors in combination. JASA 69(1), 262–274. R Core Team (2014). R: A language and environment for statistical computing. R Foundation for Statistical Computing, Vienna, Austria. Todt, K. R. (2010). The production of English front vowels by Spanish speakers: A study of vowel duration based on vowel tenseness and consonant voicing, JASA 128, 2489.

**1aSC3. A "pivot" model for extracting formant measurements based on vowel trajectory dynamics.** Aaron L. Albin and Wil A. Rankinen (Dept. of Linguist, Indiana Univ., Memorial Hall 322, 1021 E 3rd St., Bloomington, IN 47405-7005, aaalbin@indiana.edu)

Formant measurements are commonly extracted at fixed fractions across a vowel's duration (e.g., the 1/2 point for a monophthong and the 1/3 and 2/3 points for a diphthong). This approach tacitly relies on the convenience assumption that a speaker always maximally approximates the intended acoustic target at roughly the same point across a vowel's duration. The present paper proposes an alternate method whereby every formant point sampled within a vowel is considered as a possible "pivot" (i.e., turning point), with monophthongs modeled as having one pivot and diphthongs modeled as having two pivots. The optimal pivot for the vowel is then determined by fitting regression lines to the formant trajectory and comparing the goodness-of-fit of these lines to the raw formant data. When applied to a corpus of an American English dialect, the resulting measurements were found to be significantly correlated with previous methods. This suggests that the aforementioned convenience assumption is unnecessary and that the proposed model, which is more faithful to our understanding of articulatory dynamics, is a viable alternative. Moreover, rather than being assumed a priori, the location of the measurement can be treated as an empirical question in its own right.

**1aSC4. Exploiting second-order statistics improves statistical learning of vowels.** Fernando Llanos (School of Lang. and Cultures, Purdue Univ., 220 FERRY ST APT 6, Lafayette, IN 45901, fllanos@purdue.edu), Yue Jiang, and Keith R. Kluender (Dept. of Speech, Lang. and Hearing Sci., Purdue Univ., West Lafayette, IN)

Unsupervised clustering algorithms were used to evaluate three models of statistical learning of minimal contrasts between English vowel pairs. The first two models employed only first-order statistics with assumptions of uniform [M1] or Gaussian [M2] distributions of vowels in an F1-F2 space. The third model [M3] employed second-order statistics by encoding covariance between F1 and F2. Acoustic measures of F1/F2 frequencies for 12 vowels spoken by 139 men, women, and children (Hillendrand *et al.* 1995) were used as input to the models. Effectiveness of each model was tested for each minimal-pair contrast across 100 simulations. Each simulation consisted of two centroids that adjusted on a trial-by-trial basis as 1000 F1/F2 pairs were input to the models. With addition of each pair, centroids were reallocated by a k-means algorithm, an unsupervised clustering algorithm that provides an optimal partition of the space into uniformly-sized convex cells. The first-order Gaussian model [M2] performed better than a uniform distribution [M2] for six of seven minimal pairs. The second-order model [M3] was significantly superior to both first-order models for every pair. Results have implications for optimal perceptual learning of phonetic differences in ways that respect lawful covariance across vocal tract lengths that vary across talkers.

**1aSC5. Analysis of acoustic to articulatory speech inversion for natural speech.** Ganesh Sivaraman (Elec. & Comput. Eng., Univ. of Maryland College Park, 7704 Adelphi Rd., Apt 11, Hyattsville, MD 20783, ganesa90@umd.edu), Carol Espy-Wilson (Elec. & Comput. Eng., Univ. of Maryland College Park, College Park, MD), Vikramjit Mitra (SRI Int.., Menlo Park, CA), Hosung Nam (Korea Univ., Seoul, South Korea), and Elliot Saltzman (Physical Therapy & Athletic Training, Boston Univ., New Haven, Connecticut)

Speech inversion is a technique to estimate vocal tract configurations from speech acoustics. We constructed two such systems using feedforward neural networks. One was trained using natural speech data from the XRMB database and the second using synthetic data generated by the Haskins Laboratories TADA model that approximated the XRMB data. XRMB pellet trajectories were first converted into vocal tract constriction variables (TVs), providing a relative measure of constriction kinematics (location and degree) and synthetic TV data was obtained directly using TADA. The natural and synthetic speech inversion systems were trained as TV estimators using these respective sets of acoustic and TV data. TV-estimators were first tested using previously collected acoustic data on the utterance "perfect memory" spoken at slow, normal, and fast rates. The TV estimator trained on XRMB data (but not on TADA data) was able to recover the tongue tip gesture for /t/ in the fast utterance despite the gesture occurring partly during the acoustic silence of the closure. Further, the XRMB system (but not the TADA system) could distinguish between bunched and retroflexed /r/. Finally, we compared the performance of the XRMB system with a set of independently trained speaker-dependent systems (using the XRMB database) to understand the role of speaker-specific differences in the partitioning of variability across acoustic and articulatory spaces.

**1aSC6. Testing AutoTrace: A machine-learning approach to automated tongue contour data extraction.** Gustave V. Hahn-Powell (Linguist, Univ. of Arizona, 2850 N Alvernon Way, Apt 17, Tucson, AZ 85712, hahnpowell@email.arizona.edu) and Diana Archangeli (Linguist, Univ. of Hong Kong, Tucson, Arizona)

While ultrasound provides a remarkable tool for tracking the tongue's movements during speech, it has yet to emerge as the powerful research tool it could be. A major roadblock is that the means of appropriately labeling images is a laborious, time-intensive undertaking. In earlier work, Fasel and Berry (2010) introduced a "translational" deep belief network (tDBN) approach to automated labeling of ultrasound images of the tongue, and tested it against a single-speaker set of 3209 images. This study tests the same methodology against a much larger data set (about 40,000 images), using data collected for different studies with multiple speakers and multiple languages. Retraining a "generic" network with a small set of the most erroneously labeled images from language-specific development sets resulted in an almost three-fold increase in precision in the three test cases examined.

**1aSC7. Usability of SpeechMark® landmark analysis system for teaching speech acoustics.** Marisha Speights and Suzanne E. Boyce (Dept. of Commun. Sci. and Disord., Univ. of Cincinnati, PO Box 670379, Cincinnati, OH 45267-0379, speighma@mail.uc.edu)

Learning about the intersection of articulation and acoustics, and particularly acoustic measurement techniques, is challenging for students in Linguistics, Psychology and Communication Sciences and Disorders curricula. There is a steep learning curve before students can apply the material to an interesting research question; for those in more applied programs such as

Communication Disorders or ESL, there is an additional challenge in envisioning how the knowledge can be applied in changing behavior. The availability of software tools such as Wavesurfer, Praat, Audacity, TF32, the University College of London software suite, among others, has made it possible for instructors to design laboratory experiences in visualization, manipulation, and measurement of speech acoustics. Many students have found them complex for their first exposure to taking scientific measurements. The SpeechMark® acoustic landmark analysis system has been developed to automate the detection of specific acoustic events important for speech, such as voicing offset and onset, stop bursts, fricative noise, and vowel midpoints, and to provide automated formant frequency measurement used for vowel space analysis. This paper describes a qualitative multiple case study in which seven teachers of speech acoustics were interviewed to explore whether such pre-analysis of the acoustic signal could be useful for teaching.

**1aSC8. Surveying the nasal peak: A1 and P0 in nasal and nasalized vowels.** Will Styler and Rebecca Scarborough (Linguist, Univ. of Colorado, 295 UCB, Boulder, CO 80309, william.styler@colorado.edu)

Nasality can be measured in the acoustical signal using A1-P0, where A1 is the amplitude of the harmonic under F1, and P0 is the amplitude of a low-frequency nasal peak (~250 Hz) (Chen 1997). In principle, as nasality increases, P0 goes up and A1 is damped, yielding lower A1-P0. However, the details of the relationship between A1 and P0 in natural speech have not been well described. We examined 4778 vowels in French and English elicited words, measuring A1, P0, and the surrounding harmonic amplitudes, and comparing oral and nasal tokens (phonemic nasal vowels in French, and coarticulatorily nasalized vowels in English). Linear mixed-effects regressions confirmed that A1-P0 is predictive of nasality: 4.16 dB lower in English nasal contexts relative to oral and 5.73 dB lower in French (both $p<0.001$). In English, as expected, P0 increased 1.42 dB and A1 decreased 3.93 dB ($p<0.001$). In French, however, both A1 and P0 lowered with nasality (5.73 and 0.93 dB, respectively, $p<0.001$). Even so, in both languages, P0 became more prominent relative to adjacent harmonics in nasal vowels. These data reveal cross-linguistic differences in the acoustic realization of nasal vowels and suggest P0 prominence as a potential perceptual cue to be investigated.

**1aSC9. Impact of mismatch conditions between mobile phone recordings on forensic voice comparison.** Balamurali B T Nair, Esam A. Alzqhoul, and Bernard J. Guillemin (Dept. of Elec. and Comput., The Univ. of Auckland, Bldg. 303, Rm. 240, Level 2, Sci. Ctr., 38 Princes St., Auckland, Auckland, Auckland 1142, New Zealand, bbah005@aucklanduni.ac.nz)

Mismatched conditions between the recordings of suspect, offender and relevant background population represent a typical scenario in real forensic casework. In this paper, we investigate the impact of mismatch conditions associated with mobile phone speech recordings on forensic voice comparison (FVC). The two major mobile phone technologies currently in use today are the Global System for Mobile Communications (GSM) and Code Division Multiple Access (CDMA). These are fundamentally different in the way in which they handle the speech signal, which in turn will lead to significant mismatch between speech recordings. Our results suggest that the resulting degradation on the accuracy of a FVC analysis can be very significant (as high as 150%). Surprisingly, though, our results also suggest that the reliability of a FVC analysis may actually improve. We propose a strategy for lessening this impact by passing the suspect speech data through the GSM or CDMA codecs, depending on the network origin of the offender data, prior to the FVC analysis. Though this goes a long way to mitigating the impact (a reduction in loss of accuracy from 150% to 80%), it is still not as good as analysis under matched conditions.

**1aSC10. 99.8 percent accuracy achieved on Peterson and Barney (1952) acoustic measurements.** Michael A. Stokes (R & D, Waveform Commun., 3929 Graceland Ave., Indianapolis, IN 46208, waveform.model@yahoo.com)

In 2012, a paper was presented (Reetz, 2012) discussing the lack of working phonemic models, which was an acknowledgment to an earlier presentation (Ladefoged, 2004) discussing 50+ years of phonetics and phonology. These presentations highlighted the successes in phonological research over the last 60 and 50 years, respectively, but both concluded that there is still no recognized working model of phoneme identification. This presentation will discuss the Waveform Model of Vowel Perception (Stokes, 2009) achieving 99.8% accuracy on the Peterson and Barney (1952) dataset using 30 conditional statements across all ten vowels produced by the 33 males (509/510 for the vowels identified by humans at 100%). These results replicate and improve on the 99.2% achieved across the vowels produced by the males in the Hillenbrand (1995) dataset (Stokes, 2011). As a logical progression, ELBOW was developed in 2013 using the algorithm developed for static data to identify streaming vowel productions achieving over 91% before introducing improvements. Beyond ELBOW, it was essential to replicate earlier results on the most cited dataset in the literature. The Waveform Model has now replicated human performance across multiple datasets and is being successfully introduced into automatic speech recognition applications.

**1aSC11. Lombard effect based speech analysis across noisy environments for voice communications with cochlear implant subjects.** Jaewook Lee, Hussnain Ali, Ali Ziaei, and Jonh H. Hansen (Elec. Eng., Univ. of Texas at Dallas, 800 West Campbell Rd., EC33, Office ECSN 4.414, Richardson, TX 75080, jaewook@utdallas.edu)

Changes in speech production including vocal effort based on auditory feedback are an important research domain for improved human communication. For example, in the presence of environmental noise, a speaker experiences the well-known phenomenon known as Lombard effect. Lombard effect has been studied for normal hearing listeners as well as for automatic speech/speaker recognition systems, but not for cochlear implant (CI) recipients. The objective of this study is to analyze the speech production of CI users with respect to environmental change. We observe and study this effect using mobile personal audio recordings from continuous single-session audio streams collected over an individual's daily life. Prior advancements in this domain include the "Prof-Life-Log" longitudinal study at UTDallas. Four CI speakers participated by producing read and spontaneous speech in six naturalistic noisy environments (e.g., office, car, outdoor, cafeteria, etc.). A number of speech production parameters (e.g., short-time log-energy, fundamental frequency, etc.) known to be sensitive to Lombard speech were measured for both communicative and non-communicative speech as a function of environment. Results indicate that variability in the speech production parameters were found in the upward direction with an increase in background noise level. Overall higher values in acoustic variables were observed in the inter-personal conversations related to the non-conversational speech.

## Session 1aSP

# Signal Processing in Acoustics: Sampling Methods for Bayesian Signal Processing

Cameron J. Fackler, Cochair
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### *Invited Papers*

**8:40**

**1aSP1. Statistical sampling and Bayesian illumination waveform design for multiple-hypothesis target classification in cognitive signal processing.** Grace A. Clark (Grace Clark Signal Sci., 532 Alden Ln., Livermore, CA 94550, clarkga1@comcast.net)

Statistical sampling algorithms are widely used in Bayesian signal processing for drawing real-valued independent, identically distributed (i.i.d.) samples from a desired distribution. This paper focuses on the more difficult problem of how to draw complex correlated samples from a distribution specified by both an arbitrary desired probability density function and a desired power spectral density. This problem arises in cognitive signal processing. A cognitive signal processing system (for example, in radar or sonar) is one that observes and learns from the environment; then uses a dynamic closed-loop feedback mechanism to adapt the illumination waveform so as to provide system performance improvements over traditional systems. Current cognitive radar algorithms focus only on target impulse responses that are Gaussian distributed to achieve mathematical tractability. This research generalizes the cognitive radar target classifier to deal effectively with arbitrary non-Gaussian distributed target responses. The key contribution lies in the use of a kernel density estimator and an extension of a new algorithm by Nichols *et al.* for drawing complex correlated samples from target distributions specified by both an arbitrary desired probability density function and a desired power spectral density. Simulations using non-Gaussian target impulse response waveforms demonstrate very effective classification performance.

**9:00**

**1aSP2. Bayesian inversion and sequential Monte Carlo sampling techniques applied to nearfield acoustic sensor arrays.** Mingsian R. Bai (Power Mech. Eng., Tsing Hua Univ., 101 sec.2, Kuang_Fu Rd., Hsinchu 30013, Taiwan, msbai63@gmail.com), Amal Agarwal (Power Mech. Eng., Tsing Hua Univ., Mumbai, India), Ching-Cheng Chen, and Yen-Chih Wang (Power Mech. Eng., Tsing Hua Univ., Taipei, Taiwan)

This paper demonstrates that inverse source reconstruction can be performed using a methodology of particle filters that relies primarily on the Bayesian approach of parameter estimation. The proposed approach is applied in the context of nearfield acoustic holography based on the equivalent source method (ESM). A state-space model is formulated in light of the ESM. The parameters to estimate are amplitudes and locations of the equivalent sources. The parameters constitute the state vector which follows a first-order Markov process with the transition matrix being the identity for every frequency-domain data frame. The implementation of recursive Bayesian filters involves a sequential Monte Carlo sampling procedure that treats the estimates as point masses with a discrete probability mass function (PMF) which evolves with iteration. It is evident from the results that the inclusion of the appropriate prior distribution is crucial in the parameter estimation.

**9:20**

**1aSP3. Bayesian sampling for practical design of multilayer microperforated panel absorbers.** Cameron J. Fackler and Ning Xiang (Graduate Program in Architectural Acoust., Rensselaer Polytechnic Inst., 110 8th St, Greene Bldg., Troy, NY 12180, facklc@rpi.edu)

Bayesian sampling is applied to produce practical designs for microperforated panel acoustic absorbers. Microperforated panels have the capability to produce acoustic absorbers with very high absorption coefficients, without the use of porous materials. However, the absorption produced by a single panel is limited to a narrow frequency range, particularly at high absorption coefficient values. To provide broadband absorption, multiple microperforated panel layers may be combined into a multilayer absorber. To design such an absorber, the necessary number of layers must be determined and four design parameters must be specified for each layer. Using Bayesian model selection and parameter estimation, this work presents a practical method for designing multilayer microperforated panel absorbers. Particular attention is paid to aspects of the underlying sampling method that enable automatic handling of design constraints such as limitations of the manufacturing process and availability of raw materials.

9:40

**1aSP4. Particle filtering for robust modal identification and sediment sound speed estimation.** Nattapol Aunsri and Zoi-Heleni Michalopoulou (Mathematical Sci., New Jersey Inst. of Technol., 323 ML King Blvd., Newark, NJ 07102, michalop@njit.edu)

Bayesian methods provide a wealth of information on acoustic features of a propagation medium and the uncertainty surrounding their estimation. In previous work, we showed how sequential Bayesian (particle) filtering can be used to extract dispersion characteristics of a waveguide. Here, we utilize these characteristics for the estimation of geoacoustic properties of sediments. As expected, the method relies on accurate identification of modes. The effect of correct/erroneous mode identification on geoacoustic estimates is quantified and approaches are developed for robust modal recognition in conjunction with the particle filter. Additionally, the statistical behavior of the noise present in the data measurements is further investigated with more complex noise modeling leading to improved results. The approaches are validated with both synthetic and real data collected during the Gulf of Mexico Experiment. [Work supported by ONR.]



10:00–10:20 Break

10:20

**1aSP5. Efficient trans-dimensional Bayesian inversion for geoacoustic profile estimation.** Stan E. Dosso, Jan Dettmer, Gavin Steininger (School of Earth & Ocean Sci, Univ. of Victoria, PO Box 1700, Victoria, BC V8W 3P6, Canada, sdosso@uvic.ca), and Charles W. Holland (Appl. Res. Lab., The Penn State Univ., State College, PA)

This paper considers sampling efficiency of trans-dimensional (trans-D) Bayesian inversion based on the reversible-jump Markov-chain Monte Carlo (rjMCMC) algorithm, with application to seabed acoustic reflectivity inversion. Trans-D inversion is applied to sample the posterior probability density over geoacoustic parameters for an unknown number of seabed layers, providing profile estimates with uncertainties that include the uncertainty in the model parameterization. However, the approach is computationally intensive. The efficiency of rjMCMC sampling is largely determined by the proposal schemes applied to perturb existing parameters and to assign values for parameters added to the model. Several proposal schemes are examined, some of which appear new for trans-D geoacoustic inversion. Perturbations of existing parameters are considered in a principal-component space based on an eigen-decomposition of the unit-lag parameter covariance matrix (computed from successive models along the Markov chain, a diminishing adaptation). The relative efficiency of proposing new parameters from the prior versus a Gaussian distribution focused near existing values is considered. Parallel tempering, which employs a sequence of interacting Markov chains with successively relaxed likelihoods, is also considered to increase the acceptance rate of new layers. The relative efficiency of various proposal schemes is compared through repeated inversions with a pragmatic convergence criterion.

10:40

**1aSP6. Bayesian tsunami-waveform inversion with trans-dimensional tsunami-source models.** Jan Dettmer (Res. School of Earth Sci., Australian National Univ., 3800 Finnerty Rd., Victoria, Br. Columbia V8W 3P6, Canada, jand@uvic.ca), Jakir Hossen, Phil R. Cummins (Res. School of Earth Sci., Australian National Univ., Canberra, ACT, Australia), and Stan E. Dosso (School of Earth and Ocean Sci., Univ. of Victoria, Victoria, BC, Canada)

This paper develops a self-parametrized Bayesian inversion to infer the spatio-temporal evolution of tsunami sources (initial sea state) due to megathrust earthquakes. To date, tsunami-source uncertainties are poorly understood, and the effect of choices such as discretization have not been studied. The approach developed here is based on a trans-dimensional self-parametrization of the sea surface, avoids regularization constraints and provides rigorous uncertainty estimation that accounts for model-selection ambiguity associated with the source discretization. The sea surface is parametrized using self-adapting irregular grids, which match the local resolving power of the data and provide parsimonious solutions for complex source characteristics. Source causality is ensured by including rupture-velocity and obtaining delay times from the Eikonal equation. The data are recorded on ocean-bottom pressure and coastal wave gauges and predictions are based on Green-function libraries computed from ocean-basin scale tsunami models for cases that include/exclude dispersion effects. The inversion is applied to tsunami waveforms from the great 2011 Tohoku-Oki (Japan) earthquake. The tsunami source is strongest near the Japan trench with posterior mean amplitudes of ~5 m. In addition, the data appear sensitive to rupture velocity, which is part of our kinematic source model.

## *Contributed Paper*

11:00

**1aSP7. Model selection using Bayesian samples: An introduction to the deviance information criterion.** Gavin Steininger, Stan E. Dosso, Jan Dettmer (SEOS, U Vic, 201 1026 Johnson St., Victoria, BC v7v 3n7, Canada, gavin.amw.steininger@gmail.com), and Charles W. Holland (SEOS, U Vic, State College, Pennsylvania)

This paper presents the deviance information criterion (DIC) as a metric for model selection based on Bayesian sampling approaches, with examples from seabed geoacoustic and/or scattering inversion. The DIC uses all samples of a distribution to approximate Bayesian evidence, unlike more common measures such as the Bayesian information criterion, which only use point estimates. The DIC uses distribution samples to approximate Bayesian evidence, unlike more common measures such as the Bayesian information criterion based on point estimates. Hence the DIC is more appropriate for non-linear Bayesian inversions utilizing posterior sampling. Two examples are considered: determining the dominant seabed scattering mechanism (interface and/or volume scattering), and choosing between seabed profile parameterizations based on smooth gradients (polynomial splines) or discontinuous homogeneous layers. In both cases, the DIC is applied to trans-dimensional inversions of simulated and measured data, utilizing reversible jump Markov chain Monte Carlo sampling. For the first case, the DIC is found to correctly select the true scattering mechanism for simulations, and its choice for the measured data inversion is consistent with sediment cores extracted at the experimental site. For the second case, the DIC selects the polynomial spline parameterization for soft seabeds with smooth gradients. [Work supported by ONR.]

# Session 1aUW

## Underwater Acoustics: Understanding the Target/Waveguide System–Measurement and Modeling I

Kevin L. Williams, Chair
*Applied Physics Lab., University of Washington, 1013 NE 40th St., Seattle, WA 98105*

**Chair's Introduction—8:45**

### *Invited Papers*

**8:50**

**1aUW1. Very-high-speed 3-dimensional modeling of littoral target scattering.** David Burnett (Naval Surface Warfare Ctr., Code CD10, 110 Vernon Ave., Panama City, FL 32407, david.s.burnett@navy.mil)

NSWC PCD has developed a high-fidelity 3-D finite-element (FE) modeling system that computes acoustic color templates (target strength vs. frequency and aspect angle) of single or multiple realistic objects (e.g., target + clutter) in littoral environments. High-fidelity means that 3-D physics is used in all solids and fluids, including even thin shells, so that solutions include not only all propagating waves but also all evanescent waves, the latter critically affecting the former. Although novel modeling techniques have accelerated the code by several orders of magnitude, it takes about one day to compute an acoustic color template. However, NSWC PCD wants to be able to compute thousands of templates quickly, varying target/environment features by small amounts, in order to develop statistically robust classification algorithms. To accomplish this, NSWC PCD is implementing a radically different FE technology that has already been developed and verified. It preserves all the 3-D physics but promises to accelerate the code another two to three orders of magnitude. Porting the code to an HPC center will accelerate it another one to two orders of magnitude, bringing performance to seconds per template. The talk will briefly review the existing system and then describe the new technology.

**9:10**

**1aUW2. Modeling three-dimensional acoustic scattering from targets near an elastic bottom using an interior-transmission formulation.** Saikat Dey, William G. Szymczak (Code 7131, NRL, 4555 Overlook Ave. SW, Washington, DC 20375, saikat.dey@nrl.navy.mil), Angie Sarkissian (Code 7130, NRL, Washington, DC), Joseph Bucaro (Excet Inc., Springfield, VA), and Brian Houston (Code 7130, NRL, Washington, DC)

For targets near the sediment–fluid interface, the scattering response is fundamentally influenced by the characterization of the sediment in the model. We show that if the model consists of a three-dimensional elastic sediment with acoustic fluid on top, then the use of perfectly matched-layer (PML) approximation for the truncation of the infinite exterior domain for scattering applications has fundamental problems and gives erroneous results. We present a novel formulation using the an interior-transmission representation of the scattering problem where the exterior truncation with PML does not induce errors in the result. Numerical examples will be presented to verify the application of this formulation to scattering from elastic targets near a fluid–sediment interface.

**9:30**

**1aUW3. The fluid–structure interaction technique specialized to axially symmetric targets.** Ahmad T. Abawi (HLS Res., 3366 North Torrey Pines Court, Ste. 310, La Jolla, CA 92037, abawi@hlsresearch.com) and Petr Krysl (Structural Eng., Univ. of California, San Diego, La Jolla, CA)

The fluid–structure interaction technique provides a paradigm for solving scattering from elastic targets embedded in a fluid by a combination of finite and boundary element methods. In this technique, the finite element method is used to compute the target's impedance matrix and the Helmholtz–Kirchhoff integral with the appropriate Green's function is used to represent the field in the exterior medium. The two equations are coupled at the surface of the target by imposing the continuity of pressure and normal displacement. This results in a Helmholtz–Kirchhoff boundary element equation that can be used to compute the scattered field anywhere in the surrounding environment. This method reduces a finite element problem to a boundary element one with drastic reduction in the number of unknowns, which translates to a significant reduction in numerical cost. This method was developed and tested for general 3D targets. In this paper, the method is specialized to axially symmetric targets, which provides further reduction in numerical cost, and validated using benchmark solutions.

## Contributed Papers

**9:50**

**1aUW4. A new T matrix for acoustic target scattering by elongated objects in free-field and in bounded environments.** Raymond Lim (Code X11, NSWC Panama City Div., 110 Vernon Ave., Code X11, Panama City, FL 32407-7001, raymond.lim@navy.mil)

The transition (T) matrix of Waterman has been very useful for computing fast, accurate acoustic scattering predictions for axisymmetric elastic objects but this technique is usually limited to fairly smooth objects that are not too aspherical unless complex basis functions or stabilization schemes are used. To remove this difficulty, a spherical-basis formulation adapted from approaches proposed recently by Waterman [J. Acoust. Soc. Am. 125, 42–51 (2009)] and Doicu, *et al*. [*Acoustic & Electromagnetic Scattering Analysis Using Discrete Sources*, Academic Press, London, 2000] is suggested. The new method is implemented by simply transforming the high-order outgoing spherical basis functions within standard T-matrix formulations to low-order functions distributed along the object's symmetry axis. A free-field T-matrix is produced in a nonstandard form but computations with it become much more stable for aspherical shapes. Some advantages of this approach over Waterman's and Doicu, *et al*.'s approaches are noted and, despite its nonstandard form, the feasibility of extension to objects in a plane-stratified environment is demonstrated. Sample calculations for an elongated spheroid demonstrate the enhanced stability.

**10:05–10:20 Break**

**10:20**

**1aUW5. Kirchhoff approximation for spheres and cylinders partially exposed at flat surfaces and application to the interpretation of backscattering.** Aaron M. Gunderson, Anthony R. Smith, and Philip L. Marston (Phys. and Astronomy Dept., Washington State Univ., Pullman, WA 99164-2814, aaron.gunderson01@gmail.com)

For cylinders partially exposed at flat surfaces, the Kirchhoff approximation was previously evaluated analytically and compared with measured backscattering at a free surface as a function of exposure [K. Baik and P. L. Marston, IEEE J. Ocean. Eng. 33, 386–396 (2008)]. In the present research, this approach is extended to the cases of numerical integration for high frequency backscattering by partially exposed spheres and cylinders. The cylinder case was limited to broadside illumination at grazing incidence for which one-dimensional integration is sufficient and the limits of integration were previously discussed by Baik and Marston. In the corresponding sphere case, however, two-dimensional integration is required and the corresponding limits of integration become complicated functions of the amount of exposure and the grazing angle of the illumination. These approximations of the backscattering, while they omit Franz wave and elastic contributions, are useful for modeling the evolution of how the reflected scattering contributions depend on the target exposure. They are also useful for understanding the time evolution of specular scattering contributions. The sphere case was compared with the exact analysis of backscattering by a half exposed rigid sphere at a free surface that also displays partially reflected Franz wave contributions. [Work supported by ONR.]

## Invited Papers

**10:35**

**1aUW6. Acoustic ray model for the scattering from an object on the sea floor.** Steven G. Kargl, Aubrey L. Espana, and Kevin L. Williams (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, kargl@uw.edu)

Target scattering within a waveguide is recast into a ray model, where time-of-flight wave packets are tracked. The waveguide is replaced by an equivalent set of image sources and receivers, where rays are associated with these images and interactions with the waveguide's boundaries are taken into account. By transforming wave packets into the frequency domain, scattering becomes a multiplication of a wave packet's spectrum at the target location and the target's free-field scattering amplitude. Data- and model-model comparisons for an aluminum replica of a 100-mm unexploded ordnance will be discussed. For the data-model comparisons, synthetic aperture sonar (SAS) data were collect during Pond Experiment 2010 from this replica, where it was placed on a water-sand sediment boundary. The model-model comparisons use the results from a hybrid 2-D/3-D model. The hybrid model combines a 2D finite-element model to predict the scattered pressure and its derivatives in the near-field of the target, and then a 3D Helmholtz integral to propagate the pressure to the far field. The data- and model-model comparisons demonstrate the viability of using the ray model to quickly generate realistic pings suitable for both SAS and acoustic color template processing. [Research supported by SERDP and ONR.]

**10:55**

**1aUW7. Orientation dependence for backscattering from a solid cylinder near an interface: Imaging and spectral properties.** Daniel Plotnick, Philip L. Marston (Washington State Univ., 1510 NW Turner Dr., Apt. 4, Pullman, WA 99163, dsplotnick@gmail.com), Aubrey Espana, and Kevin L. Williams (Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

When a solid cylinder lies proud on horizontal sand sediment significant contributions to backscattering, specular and elastic, involve multipath reflections from the cylinder and interface. The scattering structure and resulting spectrum versus azimuthal angle, the "acoustic template," may be understood using a geometric model [K. L. Williams *et al*., J. Acoust. Soc. Am. 127, 3356–3371 (2010)]. If the cylinder is tilted such that the cylinder axis is no longer parallel to the interface the multipath structure is modified. Some changes in the acoustic template can be approximately modeled using a combination of geometric and physical acoustics. For near broadside scattering the analysis gives a simple expression relating certain changes in the template to the orientation of the cylinder and the source geometry. These changes are useful for inferring the cylinder orientation from the scattering. Changes to the template at end-on and intermediate angles are also examined. The resulting acoustic images show strong dependence on the cylinder orientation in agreement with this model. A similar model applies to a metallic cylinder adjacent to a flat free surface and was confirmed in tank experiments. The effect of vertical tilt on the acoustic image was also investigated. [Work supported by ONR.]

**11:15**

**1aUW8. Acoustic scattering enhancements for partially exposed cylinders in sand and at a free surface caused by Franz waves and other processes.** Anthony R. Smith, Aaron M. Gunderson, Daniel S. Plotnick, Philip L. Marston (Phys. and Astronomy Dept., Washington State Univ., Pullman, WA, spacetime82@gmail.com), and Grant C. Eastland (NW Fisheries Sci. Ctr., Frank Orth & Assoc. (NOAA Affiliate), Seattle, WA)

Creeping waves on solid cylinders having slightly subsonic phase velocities and large radiation damping are described as Franz waves because of association with complex poles investigated by Franz. For free-field high frequency broadside backscattering in water, the associated echoes are weak due to radiation damping. It was recently demonstrated, however, that for partially exposed solid metal cylinders at a free surface viewed at grazing incidence, the Franz wave echo can be large relative to the specular echo when the grazing angle is sufficiently small [G. C. Eastland and P. L. Marston, J. Acoust. Soc. Am. 135, 2489–2492 (2014)]. The Fresnel zone associated with the specular echo is occluded making it weak while the Franz wave is partially reflected at the interface behind the cylinder. This hypothesis is also supported by calculating the exact backscattering by half-exposed infinitely long rigid cylinders viewed over a range of grazing angles. Additional experiments concern the high frequency backscattering by cylinders partially buried in sand viewed at small grazing angles. From the time evolution of the associated backscattering by short tone bursts, situations have been identified for which partially reflected Franz wave contributions become significant. Franz waves may contribute to sonar clutter from rocks. [Work supported by ONR.]

**11:35**

**1aUW9. Pressure gradient coupling to an asymmetric cylinder at an interface.** Christopher Dudley (NSWC PCD, 110 Vernon Ave., Panama City, FL 32407, mhhd@hotmail.com)

Invited Abstract Special session: "Investigation of target response near interfaces, where coupling between target and environmental properties are important." Acoustic scattering results from solid and hollow notched aluminum cylinders are presented as a function of the incident angle. This flat machined into the circular cylinder resembles the topography(geometry) of an finned unexploded ordnance (UXO). Prior experiments have shown selective coupling to modes of a flat ended cylinder and the effect of pressure nodes to coupling to a similar notched cylinder [Espana et al., J. Acoust. Soc. Am. 126, 2187 (2009) and Marston & Marston, J. Acoust. Soc. Am. 127, 1750 (2010)]. The wavefront crossing the flat face of the notch in the paddle has a pressure gradient when not co-linear with the normal to the flat face of the notch. This pressure gradient applies a torque to the cylinder. Torsional modes can be setup in multiple scaled version of the pseudo-UXOs. Analysis of scattering experiments in the Gulf of Mexico and laboratory scale water tanks indicate robust returns form these fin like targets.

MONDAY AFTERNOON, 27 OCTOBER 2014 — MARRIOTT 7/8, 1:00 P.M. TO 5:15 P.M.

# Session 1pAA

## Architectural Acoustics: Computer Auralization as an Aid to Acoustically Proper Owner/Architect Design Decisions

Robert C. Coffeen, Cochair
*Architecture, University of Kansas, 4721 Balmoral Drive, Lawrence, KS 66047*

Kevin Butler, Cochair
*Henderson Engineers, Inc., 8345 Lenexa Dr., #300, Lenexa, KS 66214*

**Chair's Introduction—1:00**

### *Invited Papers*

**1:05**

**1pAA1. The impact of auralization on design decisions for the House of Commons of the Canadian Parliament.** Ronald Eligator (Acoustic Distinctions, 145 Huguenot St., New Rochelle, NY 10801, religator@ad-ny.com)

The House of Commons of the Canadian Parliament will be temporary relocated to a 27,000 m$^3$ glass-enclosed atrium with stone and glass walls while their home Chamber is being renovated and restored. Acoustic goals include excellent speech intelligibility for Members and guests in the room, and production of high-quality audio recordings of all proceedings for live and recorded streaming and broadcast. Room modeling and auralization using CATT Acoustic has been used to evaluate the acoustic environment of the temporary

location during design. Modeling and testing of the current House Chamber has also been performed to validate the results and conclusions drawn from the model of the new space. The use of auralizations has helped the Owner and Architect understand the impact of design choices on the achievement of the acoustic performance goals, and smoothed the path for the integration of design features that might otherwise have been difficult for them to accept. Measured and calculated data as well as audio examples will be presented.

**1:25**

**1pAA2. Cost effective auralizations to help architects and owners make informed decisions for sound isolating assemblies.** David Manley and Ben Bridgewater (D.L. Adams Assoc., Inc., 1536 Ogden St., Denver, CO 80218, dmanley@dlaa.com)

As an acoustical consultant, subjective descriptions of noise environments only get you so far. For example, it can be difficult for an Architect to qualify the difference between STC 35 and STC 40 windows on a given office space next to a highway. Often, justifying the increased cost for the increased sound isolation performance is at the forefront of the decision making process for the Owner and Architect. To help them understand the relative difference in performance, DLAA uses a simplified auralization process to create audio demonstrations of the difference between sound isolating assemblies. This presentation will discuss the process of creating the auralizations and review case studies where the auralizations helped the client make a more informed decision.



**1:45**

**1pAA3. Using auralization to aid in decision making to meet customer requirements for room response and speech intelligibility.** Thomas Tyson (Professional Systems Div., Bose, 5160 South Deborah Ct., Springfield, MO 65810, Tom_Tyson@bose.com)

To meet specific design goals such as a high degree of speech intelligibility along with targeted reverberation time; the presenter will show how the use of auralization can help determine the effectiveness of acoustic treatments and loudspeaker directivity types, beyond just the use of predicted numerical data.

**2:05**

**1pAA4. Bridging the gap between eyes and ears with auralization.** Robin S. Glosemeyer Petrone, Scott D. Pfeiffer (Threshold Acoust..com, 53 W Jackson Blvd., Ste. 815, Chicago, IL 60604, robin@thresholdacoustics.com), and Marcus Mayell (Judson Univ., Elgin, IL)

Ray trace animation, level plots, and impulse responses, while all useful tools in providing a visual representation of sound, do not always bridge the gap between the eye and ear. Threshold utilized auralization to inform decisions for an upcoming theater renovation with the goal of improving the room's acoustic support of orchestral performance. To achieve the desired acoustic response, the renovation will require major modifications to the shaping of a hall with a very distinctive architectural vernacular; a distinctive vernacular that will need to be preserved in some form to maintain the facility's identity. Along with other modeling tools, auralization provided useful support, reassuring both the client and the design team of the validity of the concepts.

**2:25**

**1pAA5. Extended tools for simulated impulse responses.** Wolfgang Ahnert and Stefan Feistel (Ahnert Feistel Media Group, Arkonastr. 45-49, Berlin D-13189, Germany, wahnert@ada-amc.eu)

To calculate impulse responses is already done since more than 25 years. The routines did allow simple calculations without and now always with scattered sound components. Today, sophistic routines calculate frequency-dependent full impulse responses comparable with measured ones. Parallel to this development, auralization routines have been developed first for monaural and binaural reproduction and nowadays ambisonic signals are created in B-Format of first and second order. These signals make audible during the reproduction in an ambisonic playback configuration the distribution of wall and ceiling reflections in computer models in EASE. Beside the acoustic detection of desired or unwanted reflections, which always is asking for the correct reproduction of the ambisonic signals the visualization of the reflection distribution is desired. In EASE, a new tool has been implemented to correlate the reflections in an impulse response with their position in a 3D presentation. This new hedgehog presentation of full impulse responses correlates angle-dependent with the view position of the model. So, any wanted or unwanted reflections may be identified quickly. A comparison with ambisonic signals via auralization is possible.

**2:45**

**1pAA6. Auralization as an aid in decision-making: Examples from professional practice.** Benjamin Markham, Robert Connick, and Jonah Sacks (Acentech Inc., 33 Moulton St., Cambridge, MA 02138, bmarkham@acentech.com)

The authors and our colleagues have presented dozens of auralizations in the service of our architectural acoustics consulting work, on projects ranging from large atriums to classrooms to sound isolation between nightclubs and surrounding facilities (and many others). The aim of most of these presentations is to communicate the relative efficacy of design alternatives or acoustical treatment options. In some cases, the effects are profound; in others, the acoustical impact may be rather subtle. Without perfect correlation, we have noted a general trend: when the observable change in acoustical attributes presented in the auralization is substantial, so too is the interest on the part of the owner to invest in significant or even aggressive acoustical design alternatives; by contrast, subtler changes in perceived acoustical character often leave owners and architects less inclined to dedicate design resources to pursue alternatives that differ from the architect or owner's original vision. Examples of auralizations following (and contradicting) this trend will be presented, along with descriptions of the design direction taken following meetings and discussions that accompanied the auralizations.

**3:05–3:20 Break**

3:20

**1pAA7. Auralization and the real world.** Shane J. Kanter (Threshold Acoust., 53 W. Jackson Blvd., Ste. 815, Chicago, IL 60604, skanter@thresholdacoustics.com), Ben Bridgewater (D.L. Adams, Denver, CO), and Robert C. Coffeen (School of Architecture, Design & Planning, The Univ. of Kansas, Lawrence, KS)

Architects value their senses and strive to design spaces that are engaging all five of them. However, architects typically make design decisions based primarily on how spaces appear and feel, as opposed to acousticians who normally justify design intent with the use of numbers, graphs, and charts. Although the data are clear to acousticians, auralizations are a useful tool to engage architects, building owners, and other clients and their sense of hearing to help them make informed decisions. If auralizations are used to demonstrate the effect of design decisions based on acoustics, there must be confidence in the accuracy and realism of these audio simulations. In order to better understand the accuracy and realism of auralizations, a study was conducted comparing auralizations created from models of an existing facility to listening within the facility. Listeners were asked to compare the "real world" sound to the auralizations of this sound by completing a survey with questions focusing on such comparisons. By presenting the actual sound and the auralizations in the same space, a direct comparison can be made and the accuracy and realism of the auralizations can be determined. Results and observations from the study will be presented.

3:40

**1pAA8. Directing room acoustic decisions for a college auditorium renovation by using auralization.** Robert C. Coffeen (Architecture, Univ. of Kansas, 4721 Balmoral Dr., Lawrence, KS 66047, rcoffeen@ku.edu)

From an acoustical viewpoint, the renovation of a multipurpose college auditorium was predicted by music and theater faculty to be a compromise not suitable for either music or theater. It was obvious that either variable sound absorption or active acoustics would be required to satisfy the multipurpose uses of the auditorium. Active acoustics was rejected by the college due to cost and an experience by one faculty member. And the faculty committee was not familiar with variable sound absorption. Using a computer model of the auditorium it was determined that the volume of the venue could be established to produce the desired maximum reverberation time for music and that vertical rising drapery could produce the desired reverberation time for drama. Auralization was used to demonstrate to the faculty committee that with variable sound absorption the auditorium could properly accommodate music of various types and theatrical performances including drama.

## *Contributed Papers*

4:00

**1pAA9. "Illuminating" reflection orders in architectural acoustics using SketchUp and light rendering.** J. Parkman Carter (Architectural Acoust., Rensselaer Polytechnic Inst., 32204 Waters View Circle, Cohoes, NY 12047, cartej8@rpi.edu)

The conventional architecture workflow tends to—quite literally—"overlook" matters of sound, given that the modeling tools of architectural design are almost exclusively visual in nature. The modeling tools used by architectural acousticians, however, produce visual representations, which are, frankly, less than inspirational for the design process. This project develops a simple scheme to visualize acoustic reflection orders using light rendering in the freely available and widely used Trimble SketchUp 3D modeling software. In addition to allowing architectural designers to visualize acoustic reflections in a familiar modeling environment, this scheme also works easily with complex geometry. The technique and examples will be presented.

4:15

**1pAA10. Using auralization to evaluate the decay characteristics that impact intelligibility in a school auditorium.** Bruce C. Olson (Ahnert Feistel Media Group, 8717 Humboldt Ave. North, Brooklyn Park, MN 55444, bcolson@afmg.eu) and Bruce C. Olson (Olson Sound Design, Brooklyn Park, MN)

Auralization was used to evaluate the effectiveness of the loudspeaker design in a high school auditorium to provide good speech intelligibility when used for lectures. The goals of this project where to offer an aural impression that enhances the visual printouts of the simulation results from the 3D model of the space in EASE using the Analysis Utility for Room Acoustics. The process used will be described and some of the results will be presented.

4:30

**1pAA11. Vibrolization: Simulating whole-body structural vibration for clients and colleagues with the Motion Platform.** Clemeth Abercrombie (Acoust., Arup, New York, NY), Tom Wilcock (Adv. Tech. and Res., Arup, New York, NY), and Andrew Morgan (Acoust., Arup, 77 Water St., Arup, New York, NY 10005, andrew.morgan@arup.com)

Arup has recently introduced an experiential design tool for demonstrating whole-body vibration. The Motion Platform, a bespoke simulator, moves vertically and can reproduce structural vibration in buildings, transport, and any other situations that involve shaking. Beyond humans, the platform can also shake objects—opening the door for developing new vibration criteria for devices such as video cameras and projectors. We will share our experience in developing the platform and how it has helped us communicate design ideas to clients and design team members.

4:45

**1pAA12. The role of auralization utilizing the end user source signal in determining final material finishes for the Chapel at St. Dominics.** David S. Woolworth (Oxford Acoust., 356 CR 102, Oxford, MS 38655, dave@oxfordacoustics.com)

The Chapel at St. Dominics Hospital in Jackson, Mississippi, was created for religious services, prayer time, and serve other spiritual needs of the hospital's patients, employees, medical staff, hospital visitors, and the greater community. It is an intimate space seating up to 100 people and is used daily by the Dominican Sisters, who first started the Jackson Infirmary in 1946. This paper outlines the process used to record the voices of the sisters and then use them to generate auralizations, which helped drive decisions regarding acoustic finishes.

5:00

**1pAA13. The construction and implementation of a multichannel loudspeaker array for accurate spatial reproduction of sound fields.** Matthew T. Neal, Colton D. Snell, and Michelle C. Vigeant (Graduate Program in Acoust., Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, mtn5048@psu.edu)

The spatial distribution of sound has a strong impact upon a listener's overall impression of a room and must be reproduced accurately for auralization. In concert hall acoustics, directionally independent metrics such as reverberation time and clarity index simply do not predict this impression. Late lateral energy level, lateral energy fraction, and the interaural correlation coefficient are measures of spatial impression, but more work is needed before we fully understand how the directional distribution of sound should influence architectural design decisions. A three-dimensional array of 28 loudspeakers and two subwoofers has been constructed in a hemi-anechoic chamber at PSU, allowing for accurate reproduction of sound fields. For the array, closed-box loudspeakers were built and digitally equalized to ensure a flat frequency response. With this facility, subjective studies investigating spatial sound in concert halls can be conducted using measured sound fields and perceptually motivated auralizations, not tied to a physical room. Such a facility is instrumental in understanding and communicating subtle differences in sound fields to listeners, whether they be musicians, architects, or clients. The flexibility and versatility of this system will facilitate room acoustics research at Penn State for years to come. [Work supported by NSF Award 1302741.]



MONDAY AFTERNOON, 27 OCTOBER 2014 LINCOLN, 1:00 P.M. TO 5:00 P.M.

## Session 1pAB

## Animal Bioacoustics and Signal Processing in Acoustics: Array Localization of Vocalizing Animals

Michelle Fournet, Cochair
*College or Earth Ocean and Atmospheric Sciences, Oregon State University, 425 SE Bridgeway Ave., Corvallis, OR 97333*

David K. Mellinger, Cochair
*Coop. Inst. for Marine Resources Studies, Oregon State University, 2030 SE Marine Science Dr., Newport, OR 97365*

**Chair's Introduction—1:00**

### *Invited Papers*

**1:05**

**1pAB1. Exploiting the sound-speed minimum to extend tracking ranges of vertical arrays in deep water environments.** Aaron Thode, Delphine Mathias (SIO, UCSD, 9500 Gilman Dr., MC 0238, La Jolla, CA 92093-0238, athode@ucsd.edu), Janice Straley (Univ. of Alaska, Southeast, Sitka, AK), Russel D. Andrews (Alaska SeaLife Ctr., Seward, AK), Chris Lunsford, John Moran (Auke Bay Labs., NOAA, Juneau, AK), Jit Sarkar, Chris Verlinden, William Hodgkiss, and William Kuperman (SIO, UCSD, La Jolla, CA)

Underwater acoustic vertical arrays can localize sounds by measuring the vertical elevation angles of various multipath arrivals generated by reflections from the ocean surface and bottom. This information, along with measurements of the relative arrival times of the multipath, can be sufficient for obtaining the range and depth of an acoustic source. At ranges beyond a few kilometers ray refraction effects add additional multipath possibilities; in particular, the existence of a sound-speed minimum in deeper waters permits purely refracted ray arrivals to be detected and distinguished on an array, greatly extending the tracking range for short-aperture systems. Here, two experimental vertical array deployments are presented. The first is a simple two-element system, deployed using longline fishing gear off Sitka, AK. By tracking a tagged sperm whale, this system demonstrated an ability to localize this species out to 35 km range, and provide estimates of the detection range of these animals as a function of sea state. The second deployment—a field trial of an 128-element, mid-frequency vertical array system off Southern California—illustrates how multi-element array gain can further extend the detection and tracking ranges of sperm and humpback whales in deep-water environments. [Work supported by NPRB, NOAA, and ONR.]

**1:25**

**1pAB2. Arrayvolution—An overview of array systems to study bats and toothed whales.** Jens C. Koblitz (German Oceanographic Museum, Katharinenberg 14-20, Stralsund 18439, Germany, Jens.Koblitz@meeresmuseum.de), Magnus Wahlberg (Dept. of Biology, RMIT Univ., Odense, Denmark), Peter Stilz (Freelance Biologist, Hechingen, Germany), Jamie MacAulay (Sea Mammal Res. Unit, Univ. of St Andrews, St. Andrews, United Kingdom), Simone Götze, Anna-Maria Seibert (Animal Physiol., Inst. for Neurobiology, Univ. of Tübingen, Tübingen, Germany), Kristin Laidre (Polar Sci. Ctr., Appl. Phys. Lab, Univ. of Washington, Seattle, WA), Hans-Ulrich Schnitzler (Animal Physiol., Inst. for Neurobiology, Univ. of Tübingen, Zübingen, Germany), and Harald Benke (German Oceanographic Museum, Stralsund, Germany)

Some echolocation signal parameters can be studied using a single receiver. However, studying parameters such as source level, directionality, and direction of signal emission require the use of multi-receiver arrays. Acoustic localization allows for determination of the position of echolocators at the time of signal emission, and when multiple animals are present, calls can be assigned to individuals based on their location. This combination makes large multi-receiver arrays a powerful tool. Here we present an overview of different array configurations used to study both toothed whales and bats, using a suite of systems ranging from semi-3D-minimum receiver number-number-arrays (3D-MINNAs), linear-2-D-over determined arrays (2D-ODAs), to 3-D-over-determined-arrays (3D-ODAs). We discuss approaches to process and summarize the usually large amounts of data. In some studies, the absolute position of an echolocator and not only relative to the array is crucial. Combining acoustic localizations from a source with geo-referenced receivers allows for determining geo-referenced movements of an echolocator. Combining these animal tracks with other geo-referenced data such as hydrographic parameters will allow new insights into habitat use.

**1:45**

**1pAB3. Tracking Cuvier's beaked whales using small aperture arrays.** Martin Gassmann, Sean M. Wiggins, and John Hildebrand (Scripps Inst. of Oceanogr., Univ. of California San Diego, 9152 Regents Rd., Apt. L, La Jolla, CA 92037, mgassmann@ucsd.edu)

Cuvier beaked whales are deep-diving animals that produce strongly directional sounds using high frequencies (>30 kHz) at which attenuation due to absorption and scattering is high (>8 dB/km). This makes it difficult to track beaked whales in three dimensions with standard large-aperture hydrophone arrays. By embedding two volumetric small-aperture (~1 m element spacing) arrays into a large-aperture (~1 km element spacing) array of five nodes, individuals and even groups of Cuvier beaked whales were tracked in three dimensions continuously up to one hour within an area of 10 km$^2$ in the Southern California Bight. This passive acoustic tracking technique provides a tool to study the characteristics of beaked whale echolocation, and their behavior during deep-diving.

**2:05**

**1pAB4. Using ocean bottom seismometer networks to better understand fin whale distributions at different spatial scales.** Michelle Weirathmueller, William SD Wilcock, and Dax C. Soule (Univ. of Washington, 1503 NE Boat St., Seattle, WA 98105, michw@uw.edu)

Ocean bottom seismometers (OBSs) are designed to monitor ground motion caused by earthquakes, but they also record low frequency vocalizations of fin and blue whales. Seismic networks used for opportunistic whale datasets are rarely optimized for acoustic localization of marine mammals. We demonstrate the use of OBSs for studying fin whales using two different networks. The first example is a small, closely spaced network of 8 OBSs deployed on the Juan de Fuca Ridge from 2003 to 2006. An automated method for identifying arrival times and locating fin whale calls using a grid search was applied to obtain 154 individual fin whale tracks over one year, revealing information on swimming patterns and spatial distribution in the vicinity of a mid ocean ridge. The second example is a network with widely spaced OBSs, such that a given call can only be detected on one instrument. The Cascadia Initiative Experiment is a sparse array of 70 OBSs covering the Juan de Fuca Plate from 2011 to 2015. Localization methods based on differential arrival times are not possible but techniques to locate the range and bearing to fin whales with a single OBS can be applied to constrain larger scale spatial distributions by comparing call densities in different regions.

**2:25**

**1pAB5. Baleen whale localization using hydrophone streamers during seismic reflection surveys.** Shima H. Abadi (Lamont–Doherty Earth Observatory, Columbia Univ., 122 Marine Sci. Bldg., University of Washington 1501 NE Boat St., Seattle, Washington 98195, shimah@ldeo.columbia.edu), Maya Tolstoy (Lamont–Doherty Earth Observatory, Columbia Univ., Palisades, NY), William S. D. Wilcock (School of Oceanogr., Univ. of Washington, Seattle, WA), Timothy J. Crone, and Suzanne M. Carbotte (Lamont–Doherty Earth Observatory, Columbia Univ., Palisades, NY)

Seismic reflection surveys use acoustic energy to image the structure beneath the seafloor, but concern has been raised about their potential impact on marine animals. Most of the energy from seismic surveys is low frequency, so the concern about their impact is focused on Baleen whales that communicate in the same frequency range. To better mitigate against this impact, safety radii are established based on the criteria defined by the National Marine Fisheries Service. Marine mammal observers use visual and acoustic techniques to monitor safety radii during each experiment. However, additional acoustic monitoring, in particular, locating marine mammals, could demonstrate the effectiveness of the observations, and help us understand animal responses to seismic experiments. A novel sound source localization technique using a seismic streamer has been developed. Data from seismic reflection surveys conducted with the R/V Langseth are being analyzed with this method to locate baleen whales and verify the accuracy of visual detections during experiments. The streamer is 8 km long with 636 hydrophones sampled at 500 Hz. The work focuses on time intervals when only a mitigation gun is firing because of marine mammal sightings. [Sponsored by NSF.]

**2:45**

**1pAB6. Faster than real-time automated acoustic localization and call association for humpback whales on the Navy's Pacific Missile Range Facility.** Tyler A. Helble (SSC-PAC, 2622 Lincoln Ave., San Diego, CA 92104, tyler.helble@gmail.com), Glenn Ierley, Gerald D'Spain (Scripps Inst. of Oceanogr., San Diego, CA), and Stephen Martin (SSC-PAC, San Diego, CA)

Optimal time difference of arrival (TDOA) methods for acoustically localizing multiple marine mammals have been applied to the data from the Navy's Pacific Missile Range Facility in order to localize and track humpback whales. Modifications to established methods were necessary in order to simultaneously track multiple animals on the range without the need for post-processing and in a fully automated way, while minimizing the number of incorrect localizations. The resulting algorithms were run with no human intervention at computational speeds faster than the data recording speed on over 40 days of acoustic recordings from the range, spanning several years and multiple seasons. Spatial localizations based on correlating sequences of units originating from within the range produce estimates having a standard deviation typically 10 m or less (due primarily to TDOA measurement errors), and a bias of 20 m or less (due to sound speed mismatch). Acoustic modeling and Monte Carlo simulations play a crucial role in minimizing both the variance and bias of TDOA localization methods. These modeling and simulation techniques will be discussed for optimizing array design, and for maximizing the quality of localizations from existing data sets.

3:05

**1pAB7. Applications of an adaptive back-propagation method for passive acoustic localizations of marine mammal sounds.** Ying-Tsong Lin, Arthur E. Newhall, and James F. Lynch (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., Bigelow 213, MS#11, WHOI, Woods Hole, MA 02543, ytlin@whoi.edu)

An adaptive back-propagation localization method utilizing the dispersion relation of the acoustic modes of low-frequency sound signals is reviewed in this talk. This method employs an adaptive array processing technique (the maximum a posteriori mode filter) to extract the acoustic modes of sound signals, and it is capable of separating signals from noisy data. The concept of the localization algorithm is to back-propagate modes to a location where the modes align with each other. Gauss-Markov inverse theory is applied to make the normal mode back-propagator adaptive to the signal-to-noise ratio (SNR). When the SNR is high, the localization procedure will push the algorithm to achieve high resolution. On the other hand, when the SNR is low, the procedure will try to retain its robustness and reduce the noise effects. Examples will be shown in the talk to demonstrate the localization performance with comparisons to other methods. Applications to baleen whale sounds collected in Cape Cod Bay, Massachusetts, will also be presented. Lastly, population density estimation using this passive acoustic localization method will be discussed.

**3:20–3:45 Break**

3:45

**1pAB8. Tracking porpoise underwater movements in tidal rapids using drifting hydrophone arrays.** Jamie D. Macaulay, Doug Gillespie, Simon Northridge, and Jonathan Gordon (SMRU, Univ. of St Andrews, 15 Crichton St., Anstruther, Fife KY103DE, United Kingdom, jdjm@st-andrews.ac.uk)

The growing interest in generating electrical power from tidal currents using tidal turbine generators raises a number of environmental concerns, including the risk that cetaceans might be injured or killed through collision with rotating turbine blades. To understand this risk we need better information on how cetaceans use tidal rapid habitats and in particular their underwater movements and dive behavior. Focusing on harbor porpoises, a European protected species, we have developed an approach which uses time of arrival differences of narrow band high frequency (NBHF) clicks detected on large aperture hydrophone arrays drifting in tidal rapids, to determine dive tracks of porpoises underwater. Probabilistic localization algorithms have been developed to filter echoes and provide accurate 2D or geo-referenced 3D locations. Calibration trials have been carried out that show that the system can provide depth and location data with submeter errors. Data collected over three seasons in tidal races around Scotland has provided new insights into how harbor porpoises are using these unique habitats, information vital for assessing the risk tidal turbines may pose.

4:00

**1pAB9. Using a coherent hydrophone array for observing sperm whale range, classification, and shallow-water dive profiles.** Duong D. Tran, Wei Huang, Alexander C. Bohn, Delin Wang (Elec. and Comput. Eng., Northeastern Univ., 006 Hayden Hall, 370 Huntington Ave., Boston, MA 02115, wang.del@husky.neu.edu), Zheng Gong, Nicholas C. Makris (Mech. Eng., Massachusetts Inst. of Technol., Cambridge, MA), and Purnima Ratilal (Elec. and Comput. Eng., Northeastern Univ., Boston, MA)

Sperm whales in the New England continental shelf and slope were passively localized, in both range and bearing, and classified using a single low-frequency (<2500 Hz), densely sampled, towed horizontal coherent hydrophone array system. Whale bearings were estimated using time-domain beamforming that provided high coherent array gain in sperm whale click signal-to-noise ratio. Whale ranges from the receiver array center were estimated using the moving array triangulation technique from a sequence of whale bearing measurements. Multiple concurrently vocalizing sperm whales, in the far-field of the horizontal receiver array, were distinguished and classified based on their horizontal spatial locations and the inter-pulse intervals of their vocalized click signals. The dive profile was estimated for a sperm whale in the shallow waters of the Gulf of Maine with 160 m water-column depth located close to the array's near-field where depth estimation was feasible by employing time difference of arrival of the direct and multiply reflected click signals received on the horizontal array. By accounting for transmission loss modeled using an ocean waveguide-acoustic propagation model, the sperm whale detection range was found to exceed 60 km in low to moderate sea state conditions after coherent array processing.



4:15

**1pAB10. Testing the beam focusing hypothesis in a false killer whale using hydrophone arrays.** Laura N. Kloepper (Dept. of Neurosci., Brown Univ., 185 Meeting St. Box GL-N, Providence, RI 02912, laura_kloepper@brown.edu), Paul E. Nachtigall, Adam B. Smith (Zoology, Univ. of Hawaii, Honolulu, HI), John R. Buck (Elec. and Comput. Eng., Univ. of Massachusetts Dartmouth, Dartmouth, MA), and Jason E. Gaudette (Neurosci., Brown Univ., Providence, RI)

The odontocete sound production system is complex and composed of tissues, air sacs, and a fatty melon. Previous studies suggested that the emitted sonar beam might be actively focused, narrowing depending on target distance. In this study, we further tested this beam focusing hypothesis in a false killer whale (*Pseudorca crassidens*) in a laboratory setting. Using three linear arrays, we recorded the same emitted click at 2, 4, and 7 m distance while the animal performed a target detection task with the target distance varying between 2, 4, and 7 m. For each click, we calculated the beamwidth, intensity, center frequency, and bandwidth as recorded on each array. As the distance from the whale to the array increased, the received click intensity was higher than predicted by spreading loss. Moreover, the beamwidth varied with range as predicted by the focusing model and contrary to a piston model or spherical spreading. These results support the hypothesis that the false killer whale adaptively focuses its sonar beam according to target range. [Work supported by ONR and NSF.]

4:30

**1pAB11. Sei whale localization and tracking using a moored, combined horizontal and vertical line array near the New Jersey continental shelf.** Arthur E. Newhall, Ying-Tsong Lin, James F. Lynch (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., 210 Bigelow Lab. MS11, Woods Hole, MA 02543, anewhall@whoi.edu), and Mark F. Baumgartner (Biology, Woods Hole Oceanographic Inst., Woods Hole, MA)

In 2006, a multidisciplinary experiment was conducted in the Mid-Atlantic continental shelf off the New Jersey coast. During a 2 day period in mid-September 2006, more than 200, unconfirmed but identifiable, sei whale (*Balaenoptera borealis*) calls were collected on a moored, combined horizontal and vertical line hydrophone array. Sei whale movements were tracked over long distances (up to tens of kilometers) using a normal mode back propagation method. This approach uses low-frequency, broadband passive sei whale call receptions from a single-station, two-dimensional hydrophone array to perform long distance localization and tracking by exploiting the dispersive nature of propagating acoustic modes in a shallow water environment. Source depth information and the source signal can also be determined from the localization application. This passive whale tracking, combined with the intensive oceanography measurements performed during the experiment, was also used to examine sei whale movements in relation to oceanographic features observed in this region.

4:45

**1pAB12. Obtaining underwater acoustic impulse responses via blind channel estimation.** Brendan P. Rideout, Eva-Marie Nosal (Dept. of Ocean and Resources Eng., Univ. of Hawaii at Manoa, 2540 Dole St., Holmes Hall 402, Honolulu, HI 96822, bprideou@hawaii.edu), and Anders Høst-Madsen (Dept. of Elec. Eng., Univ. of Hawaii at Manoa, Honolulu, HI)

Blind channel estimation is the process of obtaining the impulse responses between a source and multiple (arbitrarily placed) receivers without prior knowledge about the source characteristics or the environment. This approach could simplify localization of non-impulsive submerged sound sources (e.g., pinnipeds or cetaceans); the process of picking arrivals (direct and reflected) could be carried out on the estimated impulse responses rather than on the recorded waveforms, thus facilitating the use of time of arrival-based localization approaches. Blind channel estimation could also be useful in estimating the original source signal of a vocalizing animal through deconvolution of the estimated channel impulse responses and the recorded waveforms. In this paper, simulation and controlled pool studies will be used to explore requirements on source and environment characteristics and to quantify blind channel estimation performance for underwater passive acoustic applications.

MONDAY AFTERNOON, 27 OCTOBER 2014 INDIANA A/B, 1:15 P.M. TO 5:30 P.M.

## Session 1pBA

## Biomedical Acoustics: Medical Ultrasound

Robert McGough, Chair

*Department of Electrical and Computer Engineering, Michigan State University, 2120 Engineering Building, East Lansing, MI 48824*

### *Contributed Papers*

1:15

**1pBA1. Investigation of fabricated 1 MHz lithium niobate transfer standard ultrasonic transducer.** Patchariya Petchpong (Acoust. and Vib. Dept., National Inst. Metrology of Thailand, 75/7 Rama VI Rd., Thungphayathai, Rajthevi, Bangkok 10400, Thailand, patchariya@nimt.or.th) and Yong Tae Kim (Div. of Convergence Technol., Korea Res. Inst. of Standards and Sci., Daejeon, South Korea)

The fabrication of a single element transducer made from Lithium Niobate (LiNbO3) operating at 1 MHz is focused on this paper. The air-backed $LiNbO_3$ transducer is developed to be used as the standard transfer ultrasonic transducer to calibrate the ultrasound power-meter, which is measured the total emitted acoustic power radiated from the medical equipment. To clarify the precision of the acoustic power, the primary standard calibration measurement (radiation force balance, RFB) based on IEC 61161 is used to investigate the fabricated transducer. The geometry of the piezoelectric active element was first designed by the prediction of Krimholtz, Leedom, and Matthaei (KLM) simulation technique. The electrical impedance measurements of the $LiNbO_3$ element, before and after assembling into the transducer, were checked and compared. The results of electrical impedance show that the operating frequency is in the range from 1 MHz to 10 MHz by forming harmonics. The evaluations of total emitted power and radiation conductance of fabricated transducer were also revealed. Results of acoustic power have been responding up to 2.1 W, which can be assessed within 6% of expanded uncertainty ($k=2$).

1:30

**1pBA2. Sustained acoustic medicine for stimulation of wound healing: A translational research report.** Matthew D. Langer and George K. Lewis (ZetrOZ, 56 Quarry Rd., Trumbull, CT 06611, mlanger@zetroz.com)

The healing of both acute and chronic wounds is a challenging clinical issue affecting more than 6.5 million Americans. The regeneration phase of wound healing is critical to restoration of function, but is often prolonged because of the adverse environment for cell growth. Therapeutic ultrasound increases nutrient absorption by cells, accelerates cellular metabolism, and stimulates production of ECM proteins, which all increase the rate of wound healing. To test the effect of long duration ultrasound exposure, an initial study of wound healing was conducted in a rat model, with wounds sutured to prevent closure via contraction. In this study, a 6 mm wound healed in $9\pm2$ days when exposed to 6 hours of ultrasound therapy, and $15\pm1$ days with a placebo device ($p<0.01$). Following IRB approval of a similar protocol for use in humans, a case study was performed on the wound closure of a chronic wound. Four weeks of daily LITUS therapy reduced the wound size by 90% from its size after 21 days of treatment with standard of care. These results demonstrate the efficacy of long duration LITUS for healing wounds in an animal model and an initial case of healing in a human subject.

1:45

**1pBA3. Long duration ultrasound facilitates delivery of a therapeutic agent.** Kelly Stratton, Rebecca Taggart, and George K. Lewis (ZetrOZ, 56 Quarry Rd., Trumbull, CT 06611, george@zetroz.com)

The ability for ultrasound to enhance drug delivery through the skin has been established in an animal model. This research tested the delivery of a therapeutic agent into human skin using sustained ultrasonic application over multiple hours. An IRB-approved pilot study was conducted using hyaluronan, a polymer found in the skin and associated with hydration. To assess the effectiveness of the delivery, a standard protocol was applied to measure moisture of the volar forearm with a corneometer. Fifteen subjects applied the hyaluronan to their forearms daily. One location was then treated with a multi-hour ultrasonic treatment, and the other was not. Baseline skin hydration measurements were taken for one week, followed by daily treatments with moisturizer and corneometer measurements twice per week for three weeks. Subjects experienced double the increase in sustained moisture when ultrasound was used in conjunction with a moisturizer when compared to moisturizer alone ($p<0.001$) over the four weeks. This study successfully demonstrated ultrasound treatment enhanced delivery of a therapeutic agent into the skin.

**2:00**

**1pBA4. Characterizing the pressure field in a modified flow cytometer quartz flow cell: A combined measurement and model approach to validate the internal pressure.** Camilo Perez (BioEng. and Ctr. for Industrial and Medical Ultrasound - Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105-6698, camipiri@uw.edu), Chenghui Wang (Inst. of Acoust., College of Phys. & Information Technol., Shaanxi Normal Univ., Xi'an, Shaanxi, China), Brian MacConaghy (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., Univ. of Washington, Seattle, WA), Juan Tu (Key Lab. of Modern Acoust., Nanjing Univ., Nanjing, Jiangsu, China), Jarred Swalwell (Oceanogr., Univ. of Washington, Seattle, WA), and Thomas J. Matula (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

We incorporated an ultrasound transducer into a flow cytometer to "activate" microbubbles passing the laser interrogation zone (J. Acoust. Soc. Am. 126, 2954–2962, (2009)). This system allows high throughput recording of the volume oscillations of microbubbles, and has led to a new bubble dynamics model that incorporates shear thinning (Phys. Med. Biol. 58, 985–998 (2013)). Important parameters in the model include the ambient microbubble size, $R_0$, driving pressure, $P_A$, and the shell parameters $\chi$ and $\kappa$, the shell elasticity and viscosity, respectively. $R_0$ is obtained by calibrating the cytometer. Pressure calibration is difficult because the flow channel width (<200μm) is too small to insert a hydrophone. The objective of this study was to develop a calibration method for a 20-cycle, 1 MHz transient pressure field. The pressure field propagating through the channel and into water was compared to a 3-D FEM model. After validation, the model was used to simulate the driving pressure as input for the bubble dynamics model, leaving only $\chi$ and $\kappa$ variables. This approach was used to determine the mechanical properties for different bubbles (albumin, lipid, and lyzozyme shells). Excellent fits were obtained in many cases, but not all, suggesting heterogeneity in microbubble shell parameters.

**2:15**

**1pBA5. Entropy based detection of molecularly targeted nanoparticle ultrasound contrast agents in tumors.** Michael Hughes (Int. Med./Cardiology, Washington Univ. School of Medicine, 1632 Ridge Bend Dr., St. Louis, MO 63108, mshatctrain@gmail.com), John McCarthy (Dept. of Mathematics, Washington Univ., St. Louis, MO), Jon Marsh, and Samuel Wickline (Int. Med./Cardiology, Washington Univ. School of Medicine, Saint Louis, MO)

In this study, we demonstrate the use of "joint entropy" of two random variables (X,Y) can be applied to markedly improve tumor conspicuity (where X = f(t) =backscattered waveform and Y = g(t) = a reference waveform; both differentiable functions). Previous studies have shown that a good initial choice of reference is a reflection of the original insonifying pulse taken from a stainless-steel reflector. Using this choice, joint entropy analysis is more sensitive to accumulation of targeted contrast agents than conventional gray-scale or signal energy analysis by roughly a factor of 2 [Hughes, M. S., *et al.*, J. Acoust. Soc. Am., 133(1), p 283, 2013]. We now derive an improved reference that is applied to three groups of (MDA-435, breast tumor) flank tumor-implanted athymic nude mice to identify tumor vasculature after binding perfluorocarbon nanoparticles (~250 nm) to neovascular avb3 integrins. Five mice received i.v.avb3-targeted nanoparticles, five received nontargeted nanoparticles, and five received saline at a dose of 1 ml/kg, which was allowed to circulate for up to two hours prior to imaging. Three analogous groups of nonimplanted mice were imaged in the same region following the same imaging protocol. Our results indicate an improvement in contrast by a factor of 2.5 over previously published results. Thus, judicious selection of the reference waveform is critical to improving contrast-to-noise in tumor environments when attempting to detect targeted nanostructures for molecular imaging of sparse features.

**2:30**

**1pBA6. Effects of fluid medium flow and spatial temperature variation on acoustophoretic motion of microparticles in microfluidic channels.** Zhongzheng Liu and Yong-Joe Kim (Texas A&M Univ., 3123 TAMU, College Station, TX 77843, liuzz008@tamu.edu)

Current, state-of-the-art models of acoustophoretic forces, applied to microparticles suspended in fluid media inside microfluidic channels, and acoustic streaming velocities inside the microfluidic channels have been mainly derived with the assumption of "static" fluid media with uniform temperature distributions. Therefore, it has been challenging to understand the effects of "moving" fluid media and fluid medium temperature variation on acoustophoretic microparticle motion in the microfluidic channels. Here, a numerical modeling method to accurately predict the acoustophoretic motion of compressible microparticles in the microfluidic channels is presented to address the aforementioned challenge. In the proposed method, the Mass, Momentum, and Energy Conservation Equations and the State Equation are decomposed by using a perturbation method into the zeroth- to the second-order equations. Here, the fluid medium flow and temptation variation are considered in the zeroth-order equations and the solutions of the zeroth-order equations (i.e., the zeroth-order fluid medium velocities and temperature distribution) are propagated into the higher-order equations, ultimately affecting the second-order acoustophoretic forces and acoustic streaming velocities. The effects of the viscous fluid medium flow and the medium temperature variation on the acoustophoretic forces and the acoustic streaming velocities were then studied in this article by using the proposed numerical modeling method.

**2:45**

**1pBA7. Thrombolytic efficacy and cavitation activity of rt-PA echogenic liposomes versus Definity exposed to 120-kHz ultrasound.** Kenneth B. Bader, Guillaume Bouchoux, Christy K. Holland (Internal Medicine, Univ. of Cincinnti, 231 Albert Sabin Way, CVC 3933, Cincinnati, OH 45267-0586, Kenneth.Bader@uc.edu), Tao Peng, Melvin E. Klegerman, and David D. McPherson (Internal Medicine, Univ. of Texas Health Sci. Ctr., Houston, TX)

Echogenic liposomes can be used as a vector for co-encapsulation of the thrombolytic drug rt-PA and microbubbles. These agents can be acoustically activated for localized cavitation-enhanced drug delivery. The objective of our study was to characterize thrombolytic efficacy and sustained cavitation nucleation and activity from rt-PA-loaded echogenic liposomes (t-ELIP). A spectrophotometric method was used to determine the enzymatic activity of rt-PA released from t-ELIP and compared to unencapsulated rt-PA. The thrombolytic efficacy of t-ELIP, rt-PA alone, or rt-PA and the commercial contrast agent Definity® exposed to sub-megahertz ultrasound was determined in an *in vitro* flow model. Ultraharmonic (UH) emissions from stable cavitation were recorded during insonation. Both UH emissions and thrombolytic efficacy were significantly greater for rt-PA and Definity® over either rt-PA alone or t-ELIP with equivalent rt-PA loading. Furthermore, the enzymatic activity of t-ELIP was significantly lower than free rt-PA. When the dosage of t-ELIP was adjusted to compensate for the lack of enzymatic activity, similar thrombolytic efficacy was found for t-ELIP and Definity® and rt-PA. However, sustained ultraharmonic emissions were not observed for t-ELIP in the flow phantom.

**3:00**

**1pBA8. Temporal stability evaluation of fluorescein-nanoparticles loaded on albumin-coated microbubbles.** Marianne Gauthier (Dept. of Elec. and Comput. Eng., BioAcoust. Res. Lab., Univ. of Illinois at Urbana-Champaign, 4223 Beckman Inst.,405 N. Mathews, Urbana, IL 61801, frenchmg@illinois.edu), Jamie R. Kelly (Dept. of BioEng., BioAcoust. Res. Lab., Univ. of Illinois at Urbana-Champaign, Urbana, IL), and William D. O'Brien (Dept. of Elec. and Comput. Eng., BioAcoust. Res. Lab., Univ. of Illinois at Urbana-Champaign, Urbana, IL)

Purpose: This study aims to evaluate the temporal stability of newly designed FITC-nanoparticles (NPs) loaded on albumin-coated microbubbles (MBs) to be used for future drug delivery purposes. Materials and Methods: MBs (3.6 108 MB/mL) were obtained by sonicating 5% bovine serum

albumin and 15% dextrose solution. NPs (5 mg/mL) were produced from fluorescein (FITC)-PLA polymers and functionalized using EDC/NHS. NP-loaded MBs resulted from the covalent linking between functionalized NPs and MBs via carbodiimide technique. Three parameters were quantitatively monitored over a 4-week duration at 8 time points: MB diameter was determined using a circle detection routine based on the Hough transform, MB number density was evaluated using a hemocytometer, and NP-loading yield was assessed based on the loaded-MB fluorescence uptake. Based on the hypotheses, analyses of variance or Kruskal Wallis test were run to evaluate the stability of these physical parameters over the time of the experiment. Results: Statistical analysis exhibited no significant differences in NP-loaded MB mean sizes, number densities, and loading yields over time ($p > 0.05$). Conclusion: Newly designed NP-loaded MBs are stable over at least a 4-week duration and can be used without extra precaution concerning their temporal stability. [This work was supported by NIH R37EB002641.]

**3:15–3:30 Break**

**3:30**

**1pBA9. Chronotropic effect in rats heart caused by pulsed ultrasound.** Olivia C. Coiado and William D. O'Brien Jr. (Dept. of Elec. and Comput. Eng., Univ. of Illinois at Urbana-Champaign, 405 N Mathews, 4223 Beckman Inst., Urbana, IL 61801, oliviacoiado@hotmail.com)

This study investigated the dependence of an increasing/decreasing sequence of pulse repetition frequencies (PRFs) on the chronotropic effect via the application of 3.5-MHz pulsed ultrasound (US) on the rat heart. The experiments were divided into three 3-month-old female rat groups ($n=4$ ea): control, PRF increase and PRF decrease. Rats were exposed to transthoracic ultrasonic pulses at ~0.50% of duty factor at 2.0-MPa peak rarefactional pressure amplitude. For the PRF increase group, the PRF started lower than that of the rat's heart rate and was increased sequentially in 1-Hz steps every 5 s (i.e., 4, 5, and 6 Hz) for a total duration of 15 s. For the PRF decrease group, the PRF started greater than that of the rat's heart rate and was decreased sequentially in 1-Hz steps every 5 s (i.e., 6, 5, and 4 Hz). For the PRF decrease and control groups, the ultrasound application resulted in a significant negative chronotropic effect (~11%) after ultrasound exposure. However, for the PRF increase group, a significant but less decrease of the heart rate (~3%) was observed after ultrasound exposure. The ultrasound application caused a negative chronotropic effect after US exposure for increase/decrease US group. [Support: NIH Grant R37EB002641.]

**3:45**

**1pBA10. Ultrasonic welding in orthopedic implants.** Kristi R. Korkowski and Timothy Bigelow (Mech. Eng., Iowa State Univ., 2201 Coover Hall, Ames, IA 50011, korkowsk@iastate.edu)

A critical event in hip replacement is the occurrence of osteolysis. Cemented hip replacements most commonly use polymethylmethacrylate (PMMA), not as an adhesive but rather a filler to limit micromotion and provide stability. PMMA, however, contributes to osteolysis through both a thermal response during curing and implant wear debris. In order to mitigate the occurrence of osteolysis, we are exploring ultrasonic welding as a means of attachment. Weld strength was assessed using ex vivo bovine rib and femur bones. A flat end mill provided 20 site locations for insertion of an acrylonitrile butadiene styrene, ABS pin. Each location was characterized on topography, porosity, discoloration, and any other notable features. Each site was welded using a Branson Ultrasonic Welder 2000iw; 20 kHz, 1100 W. Machine parameters include weld force, weld time, and hold time. The bond strength was determined using a tensile tester. Tensile testing showed a negative correlation between porosity and bond strength. Further evaluation and characterization of bone properties to bond strength will enable appropriate selection of welding properties to ensure a superior bond.

**4:00**

**1pBA11. Estimation of subsurface temperature profiles from infrared measurements during ultrasound ablation.** Tyler R. Fosnight, Fong Ming Hooi, Sadie B. Colbert, Ryan D. Keil, and T. Douglas Mast (Biomedical Eng., Univ. of Cincinnati, 3938 Cardiovascular Res. Ctr., 231 Albert Sabin Way, Cincinnati, OH 45267-0586, doug.mast@uc.edu)

Measurement of *in situ* spatiotemporal temperature profiles would be useful for developing and validating thermal ablation methods and therapy monitoring approaches. Here, finite difference and analytic solutions to Pennes' bio-heat transfer equation were used to determine spatial correlations between temperature profiles on parallel planes. Time delays and scale factors for correlated profiles were applied to infrared surface-temperature measurements to estimate subsurface temperatures. To test this method, *ex vivo* bovine liver tissue was sonicated by linear image-ablate arrays with 1–6 pulses of 5.0 MHz unfocused (7.5 s, 64.4–92.0 W/cm$^2$ *in situ* $I_{SPTP}$) or focused (1 s, 562.7–799.6 W/cm$^2$ *in situ* $I_{SPTP}$, focus depth 10 mm) ultrasound. Temperature was measured on the liver surface by an infrared camera at 1 fps and extrapolated to the imaging/ablation plane, 3 mm below the surface. Echo decorrelation maps were computed from pulse-echo signals captured at 118 fps during 5.0 s rest periods beginning 1.1 s after each sonication pulse. Tissue samples were frozen at $-80\ ^\circ$C, sectioned, vitally stained, imaged, and segmented for analysis. Estimated thermal dose profiles showed correspondence with segmented tissue histology, while thresholded temperature profiles corresponded with measured echo decorrelation. These results suggest utility of this method for thermal ablation research.

**4:15**

**1pBA12. Temperature dependence of harmonics generated by nonlinear ultrasound beam propagation in water.** Borna Maraghechi, Michael C. Kolios, and Jahan Tavakkoli (Phys., Ryerson Univ., 350 Victoria St., Toronto, ON M5B 2K3, Canada, borna.maraghechi@ryerson.ca)

Ultrasound thermal therapy is used for noninvasive treatment of cancer. For accurate ultrasound based temperature monitoring in thermal therapy, the temperature dependence of acoustic parameters is required. In this study, the temperature dependence of acoustic harmonics was investigated in water. The pressure amplitudes of the transmitted fundamental frequency (p1), and its harmonics (second (p2), third (p3), fourth (p4), and fifth (p5)) generated by nonlinear ultrasound propagation were measured by a calibrated hydrophone in water. The hydrophone was placed at the focal point of a focused 5-MHz transducer (f-number 4.5) to measure the acoustic pressure. Higher harmonics were generated by transmitting a 5-MHz 15-cycle pulse that resulted in a focal positive peak pressure of approximately 0.26 MPa in water. The water temperature was increased from 26 °C to 52 °C in increments of 2 °C. Due to this temperature elevation, the value of p1 decreased by 9%±1.5% (compared to its value at 26 °C) and values of p2, p3, p4, and p5 increased by %5±2%, 22%±8%, 44%±7%, and 55%±5%, respectively. The results indicate that the nonlinear harmonics are highly temperature dependent and their temperature sensitivity increase with the harmonic number. It is concluded that the nonlinear harmonics could potentially be used for ultrasound-based thermometry.

**4:30**

**1pBA13. Implementation of a perfectly matched layer in nonlinear continuous wave ultrasound simulations.** Xiaofeng Zhao and Robert McGough (Dept. of Elec. and Comput. Eng., Michigan State Univ., East Lansing, MI, zhaoxia6@msu.edu)

FOCUS, the "Fast Object-Oriented C++ Ultrasound Simulator" (http://www.egr.msu.edu/~fultras-web), simulates nonlinear ultrasound propagation by numerically evaluating the Khokhlov–Zabolotskaya–Kuznetsov (KZK) equation. For continuous-wave excitations, KZK simulations in FOCUS previously required that the simulations extend over large radial distances relative to the aperture radius, which reduced the effect of reflections from the boundary on the main beam. To reduce the size of the grid required for these calculations, a perfectly matched layer (PML) was recently added to the KZK simulation routines in FOCUS. Simulations of the linear pressure fields generated by a spherically focused transducer with an aperture radius of 1.5 cm and a radius of curvature of 6cm are evaluated for a peak surface pressure of 0.5 MPa and a 1 MHz fundamental frequency.

Results of linear KZK simulations with and without the PML are compared to an analytical solution of the linear KZK equation on-axis, and the results show that simulations without the PML require a radial boundary that is at least seven times the aperture radius, whereas the PML enables accurate simulations for a radial boundary that is only two times the aperture radius. [This work was supported in part by NIH Grant R01 EB012079.]

**4:45**

**1pBA14. An improved time-base transformation scheme for computing waveform deformation during nonlinear propagation of ultrasound.** Boris de Graaff, Shreyas B. Raghunathan, and Martin D. Verweij (Acoust. Wavefield Imaging, Delft Univ. of Technol., Lorentzweg 1, Delft 2628CJ, Netherlands, m.d.verweij@tudelft.nl)

Nonlinear propagation plays an important role in various applications of medical ultrasound, like higher harmonic imaging and high intensity focused ultrasound (HIFU) treatment. Simulation of nonlinear ultrasound fields can greatly assist in explaining experimental observations and in predicting the performance of novel procedures and devices. Many numerical simulations are based on the generic split-step approach, which takes the ultrasound field at the transducer plane and propagates this forward over successive parallel planes. Usually, the spatial steps between the planes are small and the diffraction, attenuation, and nonlinear deformation may be treated as separate substeps. For the majority of methods, e.g., for all KZK-type methods, the nonlinear substep relies on the implicit solution of the one-dimensional Burgers equation, which is implemented using a time-base transformation. This generally works fine, but when the shock wave regime is approached, reduced spatial steps are required to avoid time points to "cross over," and the method can become notoriously slow. This paper analyses the fundamental difficulty with the common time base transformation, and provides an alternative that does not suffer from the mentioned slowdown. Numerical results will be shown to demonstrate that this alternative will allow much larger spatial steps without compromising the numerical accuracy.

**5:00**

**1pBA15. An error reduction algorithm for numeric calculation of the spatial impulse response.** Nils Sponheim (Inst. of Industrial Dev., Faculty of Technol., Art and Design, Oslo and Akershus Univ. College of Appl. Sci., Pilestredet 35, P.O. Box 4, St. Olavs plass, Oslo NO-0130, Norway, nils.sponheim@hioa.no)

The most frequently used method for calculation of the pulsed pressure field of ultrasonic transducers is the spatial impulse response (SIR) method. This paper presents a new numeric approach that reduce the numeric error by weighting the contribution of each source element into the SIR time array, by considering the exact time of arrival of each contribution. The resolution of the time array $\Delta t$ must be finite. This results in an error in travel time of $\pm\Delta t/2$. However, we know the exact travel time and based on this, we can share the contribution from each source element between the two closest time elements so that the average time corresponds to the exact travel time and thereby reduce the numeric error. This study compares the old and the new numeric algorithm with the analytic solution for a planar circular disk because it has a simple analytic solution. The paper presents calculations of the SIR for selected points in space and calculations of the RMS-error between the numeric algorithms and the analytic solution. The proposed new numeric algorithm decreases the numeric noise or error with a factor of 5 compared to the old numeric algorithm.

**5:15**

**1pBA16. Teaching auscultation visually with low cost system, is it feasible?** Sergio L. Aguirre (Universidade Federal de Santa Maria, Rua Professor Heitor da Graça Fernandes, Avenida Roraima 1000 Centro de Tecnologia, Santa Maria, Rio Grande do Sul 97105-170, Brazil, sergio.aguirre@eac.ufsm.br), Ricardo Brum, Stephan Paul, Bernardo H. Murta, and Paula P. Jardin (Universidade Federal de Santa Maria, Santa Maria, RS, Brazil)

Cardiac auscultation can generate important information in the diagnosis of diseases. The sounds that the cardiac system provides are understood in the frequency range of human hearing, but in a region of low sensitivity. This project aims to build a low cost didactic software/hardware set for teaching cardiac auscultation technique in Brazilian universities. The frequencies of interest to describe the human cardiac cycle were found in the range of 20 Hz to 1 kHz which includes low frequencies where available low-cost transducers usually have large errors. To create the system, an optimization of the geometry of the chestpiece is being programmed with finite element simulations; meanwhile, digital filters for specific frequencies of interest and an interface based on MATLAB are being developed. There were needed filters for the gallops (20 to 70 Hz), heart beats (20 to 100 Hz), ejection murmurs (100 to 500 Hz), mitral stenosis (30 to 80 Hz), and regurgitations (200 to 900 Hz). The FEM simulation of a chestpiece demonstrates high signaling levels on the desired frequency range, which can be used with the filters to obtain specific information. Furthermore, the ideal signal recording equipments will be defined, implemented, and tested.

# Session 1pNS

## Noise and Physical Acoustics: Metamaterials for Noise Control II

Olga Umnova, Cochair
*University of Salford, The Crescent, Salford M5 4WT, United Kingdom*

Keith Attenborough, Cochair
*DDEM, The Open University, Walton Hall, Milton Keynes MK7 6AA, United Kingdom*

**Chair's Introduction—12:55**

### *Invited Paper*

**1:00**

**1pNS1. Sound propagation in the presence of a resonant surface.** Logan Schwan (Univ. of Salford, The Crescent, Salford m5 4wt, United Kingdom, logan.schwan@gmail.com) and Olga Umnova (Acoust. Res. Ctr., Univ. of Salford, Salford, United Kingdom)

The interactions between acoustic waves and an array of resonators are studied. The resonators are arranged periodically on an impedance surface so that the scale separation between sound wavelength and the array period is achieved. An asymptotic multi-scale model which accounts for viscous and thermal losses in the resonators is developed and is used to derive an effective surface admittance. It is shown that the boundary conditions at the surface are substantially modified around the resonance frequency. The pressure field on the surface is nearly canceled leading to a phase shift between the reflected and the incident waves. The array can also behave as an absorbing layer. The predictions of the homogenized model are compared with multiple scattering theory (MST) applied to a finite size array and the limitations of the former are identified. The influence of the surface roughness and local scattering on the reflected wave is discussed.

### *Contributed Papers*

**1:20**

**1pNS2. Flexural wave induced coherent scattering in arrays of cylindrical shells in water.** Alexey S. Titovich and Andrew N. Norris (Mech. and Aerosp. Eng., Rutgers Univ., 98 Brett Rd., Piscataway, NJ 08854, alexey17@eden.rutgers.edu)

A periodic array of elastics shells in water is a sonic crystal with local resonances in the form of flexural vibrations. This acoustic metamaterial has seen application in wave steering by grading the index in the array, as well as acoustic filters manifested by Bragg scattering. The primary reason for using shells is that they can be tuned quasi-statically to have water-like effective acoustic properties. The issue is that the modally dense flexural resonances can form pseudogaps in the frequency response resulting in total reflection from the array. Furthermore, if a flexural resonance falls in the Bragg band gap, total transmission is possible at that frequency. Although the scattered wave due to low order flexural vibration of a thin shell is evanescent, when several shells are closely spaced, the effect on the far-field response is dramatic. In this paper, the interaction of neighboring shells is investigated theoretically using the Love-Timoshenko shell theory and multiple scattering. A simple model is offered to describe the interaction of modes based on the analytical work. The directionality of the lowest flexural modes is also discussed as it can lead to phasing between neighboring shells.

**1:35**

**1pNS3. A thin-panel underwater acoustic absorber.** Ashley J. Hicks, Michael R. Haberman, and Preston S. Wilson (Mech. Eng. and Appl. Res. Labs, Univ. of Texas at Austin, 3607 Greystone Dr., Apartment 1410, Austin, TX 78731, a.jean.hicks@utexas.edu)

We present experimental results on the acoustic behavior of thin-panel underwater sound absorbers composed of a sub-wavelength layered structure. The panels are formed using an inner layer of Delrin or PLA plastic with circular air-filled holes sandwiched between two rubber outer layers. The panel structure mimics a planar encapsulated bubble screen exactly one bubble thick, but displays performance that is significantly more broadband than a comparable bubble screen, which is only useful near the resonance frequency of the bubble. Initial results indicate 10 dB of insertion loss in the frequency range 1 kHz to 5 kHz for a panel that is about 1/250th of a wavelength in thickness at the lowest frequency. The effect of air volume fraction and the use of a 3-D printed (porous) inner layer on insertion loss will be presented and discussed. [Work supported by ONR.]

**1:50**

**1pNS4. Micromechanical effective medium modeling of metamaterials of the Willis form.** Michael B. Muhlestein, Michael R. Haberman, and Preston S. Wilson (Appl. Res. Labs. and Dept. of Mech. Eng., Univ. of Texas at Austin, 3201 Duval Rd. #928, Austin, TX 78759, mimuhle@gmail.com)

The unique behavior of acoustic metamaterials (AMM) results from deeply sub-wavelength structures with hidden degrees of freedom rather than the inherent material properties of their constituents. This distinguishes AMM from classical composite or cellular materials and also complicates attempts to model their overall response. This is especially true when sub-wavelength structures yield anisotropic effective material response, a key feature of AMM devices designed using transformation acoustics. Further, previous work has shown that the dynamic response of heterogeneous materials must include coupling between the overall strain and momentum fields [Milton and Willis, Proc. R. Soc. A **463**, 855–880, (2007)]. A micromechanical homogenization model of the overall Willis constitutive equations is presented to address these difficulties. The model yields a low-volume-fraction estimate of anisotropic and frequency-dependent effective properties in the long-wavelength limit. The model employs volume averages of the dyadic Green's function calculating the particle displacement resulting from a unit force source. This Green's function is shown to be analogous to one that determines the particle velocity in a fluid resulting from a unit dipole moment. The predicted effective properties for isotropic materials with spherical inclusions fall within the Hashin-Shtrikman bounds and agree with self-consistent estimates. [Work supported by ONR.]

**2:05**

**1pNS5. Acoustic metamaterial homogenization based on equivalent fluid media with coupled field response.** Caleb F. Sieck (Appl. Res. Labs. and Dept. of Elec. & Comput. Eng., The Univ. of Texas at Austin, 4021 Steck Ave #115, Austin, TX 78759, cfsieck@utexas.edu), Michael R. Haberman (Appl. Res. Labs. and Dept. of Mech. Eng., The Univ. of Texas at Austin, Austin, TX), and Andrea Alù (Dept. of Elec. & Comput. Eng., The Univ. of Texas at Austin, Austin, TX)

Homogenization schemes for wave propagation in heterogeneous electromagnetic (EM) and elastic materials indicate that EM bianisotropy and elastic momentum-strain and stress-velocity field coupling is required to correctly describe the effective behavior of the medium [Alu, Phys. Rev. B, **84**, 075153 (2011); Milton and Willis, Proc. R. Soc. A, **463**, 855–880, (2007)]. Further, the determination of material coupling terms in EM resolves apparent violations of causality and passivity which is present in earlier models [A. Alù, Phys. Rev. B, **83**, 081102(R) (2011)]. These details have not received much attention in fluid acoustics, but they are important for a proper description of acoustic metamaterial behavior. We derive expressions for effective properties of a heterogeneous fluid medium from expressions for the conservation of mass, the conservation of momentum, and the equation of state and find a physically meaningful effective material response from first-principles. The results show inherent coupling between the ensemble averaged volume strain-momentum and pressure-velocity field. The approach is valid for an infinite periodic lattice of heterogeneities and employs zero-, first-, and second-order tensorial Green's functions to relate point-discontinuities in compressibility and density to far field pressure and particle velocity fields. [This work was supported by the Office of Naval Research.]

**2:20**

**1pNS6. Nonlinear behavior of a coupled multiscale material containing snapping acoustic metamaterial inclusions.** Stephanie G. Konarski, Michael R. Haberman, and Mark F. Hamilton (Appl. Res. Labs., The Univ. of Texas at Austin, P.O. Box 8029, Austin, TX 78713-8029, skonarski@utexas.edu)

Snapping acoustic metamaterial (SAMM) inclusions are engineered subwavelength structures that exhibit regimes of both positive and negative stiffness. Snapping is defined as large, rapid deformations resulting from the application of an infinitesimal change in externally applied pressure. This snapping leads to a large hysteretic response at the inclusion scale and is thus of interest for enhancing absorption of energy in acoustic waves. The research presented here models the forced dynamics of a multiscale material consisting of SAMM inclusions embedded in a nearly incompressible viscoelastic matrix material to explore the influence of small-scale snapping on enhanced macroscopic absorption. The microscale is characterized by a single SAMM inclusion, while the macroscale is sufficiently large to encompass a low volume fraction of non-interacting SAMM inclusions within the nearly incompressible matrix. A model of the forced dynamical response of this heterogeneous material is achieved by coupling the two scales in time and space using a generalized Rayleigh-Plesset analysis, which has been adapted from the field of bubble dynamics. A loss factor for the heterogeneous medium is examined to characterize energy dissipation due to the forced behavior of these metamaterial inclusions. [Work supported by the ARL:UT McKinney Fellowship in Acoustics and Office of Naval Research.]

**2:35**

**1pNS7. Cloaking of an acoustic sensor using scattering cancelation.** Matthew D. Guild (Dept. of Electronics Eng., Universitat Politecnica de Valencia, Camino de vera s/n (Edificio 7F), Valencia 46022, Spain, mdguild@utexas.edu), Andrea Alù (Dept. of Elec. and Comput. Eng., Univ. of Texas at Austin, Austin, TX), and Michael R. Haberman (Appl. Res. Labs. and Dept. of Mech. Eng., Univ. of Texas at Austin, Austin, TX)

Acoustic scattering cancelation (SC) is an approach enabling the elimination of the scattered field from an object, thereby cloaking it, without restricting the incident wave from interacting with the object. This aspect of an SC cloak lends itself well to applications in which one wishes to extract energy from the incident field with minimal scattering, such as for sensing and noise control. In this work, an acoustic cloak designed based on the scattering cancelation method, and made of two effective fluid layers, is applied to the case of an acoustic sensor consisting of a hollow piezoelectric shell with mechanical absorption, providing a 20–50 dB reduction in the scattering strength. The cloak is shown to increase the range of frequencies over which there is nearly perfect phase fidelity between the acoustic signal and the voltage generated by the sensor, while remaining within the physical bounds of a passive absorber. The feasibility of achieving the necessary fluid layer properties is demonstrated using sonic crystals with the use of readily available acoustic materials. [Work supported by the US ONR and Spanish MINECO.]

**2:50**

**1pNS8. Cloaking non-spherical objects and collections of objects using the scattering cancelation method.** Ashley J. Hicks (Appl. Res. Labs. and Dept. of Mech. Eng., The Univ. of Texas at Austin, Appl. Res. Labs., 10000 Burnet Rd., Austin, TX 78758, ahicks@arlut.utexas.edu), Matthew D. Guild (Wave Phenomena Group, Dept. of Electronics Eng., Universitat Politècnica de València, Valencia, Spain), Michael R. Haberman (Appl. Res. Labs. and Dept. of Mech. Eng., The Univ. of Texas at Austin, Austin, TX), Andrea Alù (Dept. of Elec. and Comput. Eng., The Univ. of Texas at Austin, Austin, TX), and Preston S. Wilson (Appl. Res. Labs. and Dept. of Mech. Eng., The Univ. of Texas at Austin, Austin, TX)

Acoustic cloaks can be designed using transformation acoustics (TA) to guide acoustic disturbances around an object. TA cloaks, however, require the use of exotic materials such as pentamode materials [Proc. R. Soc. A. **464**, pp. 2411–2434, (2008)]. Alternatively, the scattering cancelation (SC) method allows the cloaked object to interact with the acoustic wave and can be realized with isotropic materials [Phys. Rev. B., **86**, 104302 (2012)]. Unfortunately, SC cloaking performance may be degraded if the shape of the cloaked object diverges from the one for which the cloak was originally designed. This study investigates the design of two-layer SC cloaks for imperfect spherical objects. The cloaking material properties are determined by minimizing the scattered field from a model of the imperfect object approximated as a series of concentric shells. Predictions from this approximate analytical model are compared with three-dimensional finite element (FE) models of the cloaked and uncloaked non-spherical shapes. Analytical and FE results are in good agreement for $ka \leq 5$, indicating that the SC method is robust to object imperfections. Finally, FE models are used to explore SC cloak robustness to multiple-scattering by investigating linear arrays of cloaked objects for different incident angles. [Work supported by ONR.]

**3:05**

**1pNS9. Parity-time symmetric metamaterials and metasurfaces for loss-immune and broadband acoustic wave manipulation.** Romain Fleury, Dimitrios Sounas, and Andrea Alu (ECE Dept., The Univ. of Texas at Austin, 1 University Station C0803, Austin, TX 78712, romain.fleury@utexas.edu)

We explore the largely uncharted scattering properties of acoustic systems that are engineered to be invariant under a special kind of space-time

symmetry, consisting in taking their mirror image and running time backwards. Known as Parity-Time symmetry, this special condition is shown here to lead to acoustic metamaterials that possess a balanced distribution of gain (amplifying) and loss (absorbing) media, at the basis of ideal loss-compensation, and under certain conditions, unidirectional invisibility. We have designed and built the first acoustic metamaterial with parity-time symmetric properties, obtained by pairing the acoustic equivalent of a lasing system with a coherent perfect acoustic absorber, implemented using electro-acoustic resonators loaded with non-Foster electrical circuits. The active system can be engineered to be fully stable and, in principle, broadband. We discuss the underlying physics and present the realization of a unidirectional invisible acoustic sensor with unique sensing properties. We also discuss the potential of PT acoustic metamaterials and metasurfaces for a variety of metamaterial-related applications, which we obtain in a loss-immune and broadband fashion, including perfect cloaking of sensors, planar focusing, and unidirectional cloaking of large objects.

MONDAY AFTERNOON, 27 OCTOBER 2014 INDIANA C/D, 1:15 P.M. TO 4:45 P.M.

## Session 1pPA

## Physical Acoustics and Noise: Jet Noise Measurements and Analyses II

Richard L. McKinley, Cochair
*Battlespace Acoustics, Air Force Research Laboratory, 2610 Seventh Street, Wright-Patterson AFB, OH 45433-7901*

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

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

**Chair's Introduction—1:15**

### *Invited Papers*

**1:20**

**1pPA1. Considerations for array design and inverse methods for source modeling of full-scale jets.** Alan T. Wall (Battlespace Acoust. Branch, Air Force Res. Lab., Bldg. 441, Wright-Patterson AFB, OH 45433, alantwall@gmail.com), Blaine M. Harker, Trevor A. Stout, Kent L. Gee, Tracianne B. Neilsen (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT), Michael M. James (Blue Ridge Res. and Consulting, Asheville, NC) and Richard L. McKinley (Air Force Res. Lab., Boston, OH)

Microphone array-based measurements of full-scale jet noise sources necessitate the adaptation and incorporation of advanced array processing methodologies. Arrays for full-scale jets measurements can require large apertures, high spatial sampling densities, and strategies to account for partially coherent fields. Many approaches have been taken to sufficiently capture radiated noise in past jet noise investigations, including patch-and-scan measurements with a small dense array, one-dimensional measurements along the extent of the jet in conjunction with an axisymmetric assumption, and full two-dimensional source coverage with a large microphone set. Various measurement types are discussed in context of physical jet noise field properties, such as spatial coherence, source stationary, and frequency content.

**1:40**

**1pPA2. Toward the development of a noise and performance tool for supersonic jet nozzles: Experimental and computational results.** Christopher J. Ruscher (Spectral Energies, LLC, 2654 Solitaire Ln. Apt. #3, Beavercreek, OH 45431, cjrusche@gmail.com), Barry V. Kiel (RQTE, Air Force Res. Lab., Dayton, OH), Sivaram Gogineni (Spectral Energies, LLC, Dayton, OH), Andrew S. Magstadt, Matthew G. Berry, and Mark N. Glauser (Dept. of Mech. and Aerosp. Eng., Syracuse Univ., Syracuse, NY)

Modal decomposition of experimental and computational data for a range of two- and three-stream supersonic jet nozzles will be conducted to study the links between the near-field flow features and the far-field acoustics. This is accomplished by decomposing near-field velocity and pressure data using proper orthogonal decomposition (POD). The resultant POD modes are then used with the far-field sound to determine a relationship between the near-field modes and portions of the far-field spectra. A model will then be constructed for each of the fundamental modes, which can then be used to predict the entire far-field spectrum for any supersonic jet. The resultant jet noise model will then be combined with an existing engine performance code to allow parametric studies to optimize thrust, fuel consumption, and noise reduction.

**2:00**

**1pPA3. Finely resolved spatial variation in F-22 spectra.** Tracianne B. Neilsen, Kent L. Gee, Hsin-Ping C. Pope, Blaine Harker (Brigham Young Univ., N311 ESC, Provo, UT 84602, tbn@byu.edu), and Michael M. James (Blue Ridge Res. and Consulting, LLC, Asheville, NC)

Examination of the spatial variation in the spectrum from ground-based microphones near an F-22 Raptor has revealed spectral features at high engine power that are not seen at intermediate power or in laboratory-scale jet noise. At military and afterburner powers, a double peaked spectrum is detected around the direction of maximum radiation. In this region, there is not a continuous variation in peak frequency with downstream distance, as seen in lab-scale studies, but a transition between the relative levels for two discrete one-third octave bands. Previous attempts to match similarity spectra for turbulent mixing noise to a few of these measurements split the difference between the two peak frequencies [Neilsen *et al*., J. Acoust. Soc. Am. 133, 2116–2125 (2013)]. The denser spatial resolution afforded by examining the spectral variation on all 50 ground-based microphones, located 11.6 m to the sideline and spanning 30 m, provides the opportunity to further investigate this phenomenon and propose a more complete formulation of expected spectral shapes. Special care must be given to account for the relative amount of waveform steepening, which varies with level, distance, and angular position. [Work supported by ONR.]

1p MON. PM

**2:20**

**1pPA4. Experimental and computational studies of noise reduction for tactical fighter aircraft.** Philip Morris, Dennis K. McLaughlin, Russell Powers, Nidhi Sikarwar, and Matthew Kapusta (Aerosp. Eng., Penn State Univ., 233C Hammond Bldg., University Park, PA 16802, pjm@psu.edu)

The noise levels generated by tactical fighter aircraft can result in Noise Induced Hearing Loss for Navy personnel, particularly those involved in carrier deck operations. Reductions in noise source levels are clearly necessary, but these must be achieved without a loss in aircraft performance. This paper describes an innovative noise reduction technique that has been shown in laboratory scale measurements to provide significant reductions in both mixing as well as broadband shock-associated noise. The device uses the injection of relatively low pressure and low mass flow rate air into the diverging section of the military-style nozzle. This injection generates “fluidic inserts” that change the effective nozzle area ratio and generate streamwise vorticity that breaks up the large scale turbulent structures in the jet exhaust that are responsible for the dominant mixing noise. The paper describes noise measurements with and without forward flight that demonstrate the noise reduction effectiveness of the inserts. The experiments are supported by computations that help to understand the flow field generated by the inserts as well as help to optimize the distribution and strength of the flow injection.

**2:40**

**1pPA5. Detection and analysis of shock-like waves emitted by heated supersonic jets using shadowgraph flow visualization.** Nathan E. Murray (National Ctr. for Physical Acoust., The Univ. of MS, 1 Coliseum Dr., University, MS 38677, nmurray@olemiss.edu)

Shock-like waves in the acoustic field adjacent to the shear layer formed by a supersonic, heated jet are observed using the method of retro-reflective shadowgraphy. The two inch diameter jet issued from a converging–diverging nozzle at a pressure ratio of 3.92 with a temperature ratio of 3.3. Image sets were obtained near the jet exit and in the post-potential core region. In both locations, shock-like waves can be observed immediately adjacent to the jet shear layer. Each image is subdivided into a set of overlapping tiles. A radon transform is applied to the auto-correlation of each tile providing a quantitative measure of the dominant propagation direction of waves in each sub-region. The statistical distribution of propagation angles over the image space provides a measure of the distribution of source convection speeds and source locations in the jet shear layer. Results show general agreement with a convection speed on the order of 70 percent of the jet velocity.

**3:00–3:20 Break**

**3:20**

**1pPA6. Where are the nonlinearities in jet noise?** Charles E. Tinney (Ctr. for AeroMech. Res., The Univ. of Texas at Austin, ASE/EM, 210 East 24th St., Austin, TX 78712, cetinney@utexas.edu) and Woutijn J. Baars (Mech. Eng., The Univ. of Melbourne, Parkville, VIC, Australia)

For some time now it has been theorized that spatially evolving instability waves in the irrotational near-field of jet flows couple both linearly and nonlinearly to generate far-field sound [Sandham and Salgado, Philos. Trans. R. Soc. Am. 366 (2008); Suponitsky, J. Fluid Mech. 658 (2010)]. An exhaustive effort at The University of Texas of Austin was initiated in 2008 to better understand this phenomenon, which included the development of a unique analysis technique for quantifying their coherence [Baars *et al*., AIAA Paper 2010–1292 (2010); Baars and Tinney, Phys. Fluids 26, 055112 (2014)]. Simulated data have shown this technique to be effective, albeit, insurmountable failures arise when exercised on real laboratory measurements. The question that we seek to address is how might jet flows manifest nonlinearities? Both subsonic and supersonic jet flows are considered with simulated and measured data sets encompassing near-field and far-field pressure signals. The focus then turns to considering nonlinearities in the form of cumulative distortions, and the conditions required for them to be realized in a laboratory scale facility [Baars, *et al*., J. Fluid Mech. 749 (2014)].

**3:40**

**1pPA7. Characterization of supersonic jet noise and its control.** Ephraim Gutmark, Dan Cuppoletti, Pablo Mora, Nicholas Heeb, and Bhupatindra Malla (Aerosp. Eng. and Eng. Mech., Univ. of Cincinnati, 799 Rhodes Hall, Cincinnati, OH 45221, gutmarej@ucmail.uc.edu)

As supersonic aircraft and their turbojet engines become more powerful they emit more noise. The principal physical difference between the jets emanating from supersonic jets and those from subsonic jets is the presence of shocks in the supersonic one. This paper summarizes a study of noise reduction technologies applied to supersonic jets. The measurements are performed with a simulated

exhaust of a supersonic nozzle representative of supersonic aircraft. The nozzle has a design Mach number of 1.56 and is examined at design and off-design conditions. Several components of noise are present including mixing noise, screech, broadband shock associated noise, and crackle. Chevrons and fluidic injection by microjets and a combination of them are shown to reduce the noise generated by the main jet. These techniques provide significant reduction in jet noise. PIV provides detailed information of the flow and brings out the physics of the noise production and reduction process.

## *Contributed Papers*

**4:00**

**1pPA8. Influence of windscreen on impulsive noise measurement.** per rasmussen (G.R.A.S. Sound & Vib. A/S, Skovlytoften 33, Holte 2840, Denmark, pr@gras.dk)

The nearfield noise from jet engines may contain impulsive sound signals with high crest factors. Most jet engine noise measurements are performed outside in potentially windy conditions, and it may, therefore, be necessary to use windscreens on microphones to reduce the influence of wind induced noise on the microphone. The windscreen will, however, influence the frequency response of the microphone especially at high frequencies. This will change both the magnitude and the phase response and, therefore, change the measured impulse. The effect of different sizes of windscreen is investigated and the effect on impulsive type signals is evaluated both in the time domain and the frequency domain.

**4:15**

**1pPA9. Comparison of nonlinear, geometric, and absorptive effects in high-amplitude jet noise propagation.** Brent O. Reichman, Kent L. Gee, Tracianne B. Neilsen, Joseph J. Thaden (Brigham Young Univ., 453 E 1980 N, #B, Provo, UT 84604, brent.reichman@byu.edu), and Michael M. James (Blue Ridge Research and Consulting, LLC, Asheville, NC)

In recent years, understanding of nonlinearity in noise from high-performance jet aircraft has increased, with successful modeling of nonlinear propagation in the far field. However, the importance and characteristics of nonlinearity in the near field are still debated. An ensemble-averaged, frequency-domain version of the Burgers equation can be inspected to directly compare the effects of nonlinearity on the sound pressure level with the effects of atmospheric absorption and geometric spreading on a decibel scale. This nonlinear effect is calculated using the quadspectrum of the pressure and the squared pressure waveforms. Results from applying this analysis to F-22A data at various positions in the near field reveal that in the near field the nonlinear effects are of the same order of magnitude as geometric spreading and that both of these effects are significantly greater than absorption in the area of maximum radiation. [Work supported by ONR and an ORISE fellowship through AFRL.]

**4:30**

**1pPA10. Correlation lengths in deconvolved cross-beamforming measurements of military jet noise.** Blaine M. Harker, Kent L. Gee, Tracianne B. Neilsen (Dept. of Phys. and Astronomy, Brigham Young Univ., N283 ESC,, Provo, UT 84602, blaineharker@byu.net), Alan T. Wall (Battlespace Acoust. Branch, Air Force Res. Lab., Wright-Patterson Air Force Base, OH), and Michael M. James (Blue Ridge Research and Consulting, LLC, Asheville, NC)

Beamforming algorithms have been applied in multiple contexts in aeroacoustic applications, but difficulty arises when applying these to the partially correlated and distributed sources found in jet noise. To measure and more accurately distinguish correlated sources, cross-beamforming methods are employed to incorporate correlation information. Deconvolution methods such as DAMAS-C, an extension of the deconvolution approach for the mapping of acoustic sources (DAMAS), remove array effects from cross-beamforming applications and further resolve beamforming results. While DAMAS-C results provide insight to correlation between sources, the extent to which these results relate to source correlation remains to be analyzed. Numerical simulations of sources with varying degrees of correlation are provided to benchmark the DAMAS-C results. Finally, correlation lengths are established for DAMAS-C results from measurements for full-scale military jet noise sources. [Work supported by ONR.]

# Session 1pSCa

## Speech Communication and Biomedical Acoustics: Findings and Methods in Ultrasound Speech Articulation Tracking

Keith Johnson, Cochair
*Linguistics, University of California, Berkeley, 1203 Dwinelle Hall, Berkeley, CA 94720*

Susan Lin, Cochair
*UC Berkeley, 1203 Dwinelle Hall, UC Berkeley, Berkeley, CA 94720*

**Chair's Introduction—1:00**

### *Invited Papers*

**1:05**

**1pSCa1. Examining suprasegmental and morphological effects on constriction degree with ultrasound imaging.** Lisa Davidson (Linguist, New York Univ., 10 Washington Pl., New York, NY 10003, lisa.davidson@nyu.edu)

Two case studies of ultrasound imaging use tongue shape differences to investigate whether suprasegmental influences affect the articulatory implementation of otherwise equivalent phonemic sequences. First, we examine whether word-medial and word-final stop codas have the same degree of constriction (e.g., "blacktop" vs. "black top"). Previous research on syllable position effects on articulatory implementation have conflated syllable position with word position, and this study investigates whether each prosodic factor has an independent contribution. Results indicate that where consistent differences are found, they are due not to the prosodic position but to speaker-specific implementation. Second, we examine whether morphological status influences the darkness of American English /l/ in comparing words like "tallest" and "flawless." While the intervocalic /l/s in "tall-est" and "flaw-less" are putatively assigned the same syllabic status, the /l/ in "tallest" corresponds to the coda /l/ of the stem "tall" whereas that of "flawless" is the onset of the affix "-less." Results indicate that /l/ is darker—the tongue is lower and more retracted—when corresponding to the coda of the stem word. Data in both studies were analyzed with smoothing spline ANOVA, an effective statistical technique for examining differences between whole tongue curves.

**1:25**

**1pSCa2. Imaging dynamic lingual movements that we could previously only imagine.** Amanda L. Miller (Linguist, The Ohio State Univ., 222 Oxley Hall, 1712 Neil Ave., Columbus, OH 43210-1298, amiller@ling.osu.edu)

Pioneering lingual ultrasound studies of speech demonstrated that almost the entire tongue could be imaged (McKay 1957). Early studies contributed to our knowledge of tongue shape and tongue bracing in vowels (Morrish *et al.* 1984; Stone *et al.* 1987). However, until recently, lingual ultrasound studies have been limited to standard video frame rates of 30 fps, which are sufficient only for imaging stable speech sounds such as vowels and liquids. High frame rate lingual ultrasound (>100 fps) allows us to view the production of dynamic speech sounds, such as stop consonants, and even click consonants. The high sampling rate, which yields an image of the tongue every 8–9 ms, improves image quality, by decreasing temporal smear, allowing even tongue tip movements to be visualized to a greater extent than was previously possible. Results from several high frame rate ultrasound studies (114 fps) of consonants that were collected and analyzed using the CHAUSA method (Miller and Finch 2011) are presented. The studies elucidate (a) tongue dorsum and root gestures in velar and uvular pulmonic consonants; (b) tongue coronal, dorsal, and root gestures in four contrastive click consonants; and (c) lingual gestures in pulmonic fricatives.

**1:45**

**1pSCa3. Ultrasound evidence for place of articulation of the mora nasal /ɴ/ in Japanese.** Ai Mizoguchi (The Graduate Ctr., City Univ. of New York, 365 Fifth Ave., Rm. 7304, New York, NY 10016, amizoguchi@gc.cuny.edu) and Douglas H. Whalen (Haskins Labs., New Haven, CT)

The Japanese mora nasal /ɴ/, which occurs in syllable-final position, takes its place of articulation from the following segment if there is one. However, the mora nasal in utterance-final position is often transcribed as velar, uvular, or even placeless. The present study examines the tongue shapes in Japanese using ultrasound imaging to investigate whether Japanese mora nasal /ɴ/ is placeless and to assess whether assimilation to following segments is gradient or categorical. Preliminary results from ultrasound imaging from one native speaker of Tokyo dialect showed three shapes for final /ɴ/, even though the researchers could not distinguish them perceptually. Results from assimilation contexts showed that the velar gesture for /ɴ/ was not deleted. All gestures remained and assimilation was not categorical, even though perceptually, it was. The velar gesture for /ɴ/ might be expected to be deleted before an alveolar /n/ because they are both lingual, but a blending of the two tongue gestures occurred instead. Variability in place of articulation in final position occurred even within one speaker. Categorical assimilation was not observed in any phonological environments studied. The mora nasal may vary across speakers, so further research is needed to determine whether it behaves similarly for more speakers.

**2:05**

**1pSCa4. A multi-modal imaging system for simultaneous measurement of speech articulator kinematics for bedside applications in clinical settings.** David F. Conant (Neurological Surgery, UCSF, 675 Nelson Rising Ln., Rm. 635, San Francisco, CA 94143, dfconant@gmail.com), Kristofer E. Bouchard (LBNL, San Francisco, CA), Anumanchipalli K. Gopala, Ben Dichter, and Edward F. Chang (Neurological Surgery, UCSF, San Francisco, CA)

A critical step toward a neurological understanding of speech generation is to relate neural activity to the movement of articulators. Here, we describe a noninvasive system for simultaneously tracking the movement of the lips, jaw, tongue, and larynx for human neuroscience research carried out at the bedside. We combined three methods previously used separately: videography to track the lips and jaw, electroglottography to monitor the larynx, and ultrasonography to track the tongue. To characterize this system, we recorded articulator positions and acoustics from six speakers during production of nine American English vowels. We describe processing methods for the extraction of kinematic parameters from the raw signals and methods to account for artifacts across recording conditions. To understand the relationship between kinematics and acoustics, we used regularized linear regression between the vocal tract kinematics and speech acoustics to identify which, and how many, kinematic features are required to explain both across vowel and within vowel acoustics. Furthermore, we used unsupervised matrix factorization to derive "prototypical" articulator shapes, and use them as a basis for articulator analysis. These results demonstrate a multi-modal system to non-invasively monitor speech articulators for clinical human neuroscience applications and introduce novel analytic methods for understanding articulator kinematics.

**2:25**

**1pSCa5. A study of tongue trajectories for English /æ/ using articulatory signals automatically extracted from lingual ultrasound video.** Jeff Mielke, Christopher Carignan, and Robin Dodsworth (English, North Carolina State Univ., 221 Tompkins Hall, Campus Box 8105, Raleigh, NC 27695-8105, ccarign@ncsu.edu)

While ultrasound imaging has made articulatory phonetics more accessible, quantitative analysis of ultrasound data often reduces speech sounds to tongue contours traced from single video frames, disregarding the temporal aspect of speech. We propose a tracing-free method for directly converting entire ultrasound videos to phonetically interpretable articulatory signals using Principal Component Analysis of image data (Hueber *et al.* 2007). Once a batch of ultrasound images (e.g., 36,000 frames from 10 min at 60 fps) has been reduced to 20 principal components, numerous techniques are available for deriving temporally changing articulatory signals that are both phonetically meaningful and comparable across speakers. Here we apply a regression model to find the linear combination of PCs that is the lingual articulatory analog of the front diagonal of the acoustic vowel space (Z2-Z1). We demonstrate this technique with a study of /æ/ tensing in 20 speakers of North American English varieties with different tensing environments (Labov 2005). Our results show that /m n/ condition a tongue raising gesture that is aligned to the vowel nucleus, while /g/ conditions anticipatory raising toward the velar target. /ŋ/ patterns consistently with the other velar rather than the other nasals.

**2:45–3:05 Break**

**3:05**

**1pSCa6. Combined analysis of real-time three-dimensional tongue ultrasound and digitized three-dimensional palate impressions: Methods and findings.** Steven M. Lulich (Speech and Hearing Sci., Indiana Univ., 4789 N White River Dr., Bloomington, IN 47404, slulich@indiana.edu)

Vocal tract and articulatory imaging has a long and rich history using a wide variety of techniques and equipment. This presentation focuses on combining real-time 3D ultrasound with high-resolution 3D digital scans of palate impressions. Methods for acquiring and analyzing these data will be presented, including efforts to accomplish 3D registration of the tongue and hard palate. Findings from an experiment investigating inter-speaker variability in palate shape and vowel articulation will also be presented.

**3:25**

**1pSCa7. AutoTrace: An automatic system for tracing tongue contours.** Gustave V. Hahn-Powell (Linguist, Univ. of Arizona, 2850 N Alvernon Way, Apt. 17, Tucson, AZ 85712, hahnpowell@email.arizona.edu) and Diana Archangeli (Linguist, Univ. of Hong Kong, Tucson, Arizona)

Ultrasound imaging of the tongue is used for analyzing the articulatory features of speech sounds. In order to be able to study the movements of the tongue, the tongue surface contour has to be traced for each recorded image. In order to capture the details of the tongue's movement during speech, the ultrasound video is generally recorded at the highest frame rate available. Detail comes at a price. The number of frames produced from even a single non-trivial experiment is often far too large to trace manually. The Arizona Phonological Imaging Lab (APIL) at the University of Arizona has developed a suite of tools to simplify the labeling and analysis of tongue contours. AutoTrace is a state-of-the-art automatic method for tracing tongue contours that is robust across speakers and languages and operates independently of frame order. The workshop will outline the software installation procedure, introduce the included tools for selecting and preparing training data, provide instructions for automated tracing, and overview a method for measuring the network's accuracy using the Mean Sum of Distances (MSD) metric described by Li *et al.* (2005).

## Contributed Papers



3:45

**1pSCa8. UATracker: A tool for ultrasound data management.** Mohsen Mahdavi Mazdeh and Diana B. Archangeli (Linguist, Univ. of Arizona, 3150 E Bellevue St., #16, Tucson, AZ 85716, mahdavi@email.arizona.edu)

This presentation introduces TraceTracker, a tool for efficiently managing language ultrasound data. Ultrasound imaging of the tongue is used for analyzing the articulatory features of speech sounds. Most analyses involve finding data points from individual images. The number of image frames and the volume of secondary data associated with them tend to grow quickly in speech analysis studies of this type, making it very hard to handle them manually. TraceTracker is a data management tool for organizing, modifying, and performing advanced searches over ultrasound tongue images and the data associated with those images. The setup operation of the program automatically iterates through file systems and generates a comprehensive database containing the image files and information such as the speaker, the video each frame is extracted from, an index, how they have been traced, etc. The program also automatically reads Praat format TextGrid files and associates specific image frames with the corresponding words and speech segments based on the annotations in the grids. Once the database is populated, TraceTracker can be used to tag images, generate copies, and perform advanced search operations over the images based on the aforementioned criteria including the specific sequence of segments in which it lies.

4:00

**1pSCa9. Optical flow analysis for measuring tongue-motion.** Adriano V. Barbosa (Electron. Eng., Federal Univ. of Minas Gerais, Belo Horizonte, Brazil) and Eric Vatikiotis-Bateson (Linguist, Univ. Br. Columbia, 2613 West Mall, Vancouver, BC V6N2W4, Canada, evb@mail.ubc.ca)

Most attempts to measure motion of the tongue have focused on locating the upper surface of the tongue or specific points on that surface. Recently, we have used our software implementation of optical flow analysis, FlowAnalyzer, to extract measures of tongue motion. The software allows identification of multiple regions of interest, consisting of rectangles whose dimensions and location are user-definable. For example, a large region encompassing the visible tongue body provides general information about the amount and direction (2D) of motion through time; while narrow vertical rectangles can measure the time-varying changes of tongue height at various locations. We will demonstrate the utility of the software, which is freely available upon request to the authors.

4:15

**1pSCa10. An acoustic profile of Spanish trill /r/.** Ahmed Rivera-Campos and Suzanne E. Boyce (Commun. Sci. and Disord., Univ. of Cincinnati, 3202, Eden Ave., Cincinnati, OH 45267, riveraam@mail.uc.edu)

Unlike English rhotic, there is limited data on the acoustic profile of Spanish trill /r/. It is well known that one key aspect of the English rhotic /ɹ/ is the lowering of the F3 formant but limited information can be found if Spanish trill shares the same characteristics. Although it has been described that a lowering of F3 is not something that characterizes /r/ production and that F3 values fall under certain ranges that are delimited by vowel contexts, analysis of F3 values has not been done using a large sample of native speakers of Spanish. The following study analyzed the F3 values of 20 participants after production of /r/ by different native speakers of Spanish from different regions of Latin America, and the Caribbean. Analysis of F3 values of /r/ provides information about articulatory requirements for adequate /r/ production. This information will benefit professionals that service individuals with articulatory difficulties or are learning Spanish as a second language.

4:30

**1pSCa11. Investigation of the role of the tongue root in Kazakh vowel production using ultrasound.** Jonathan N. Washington (Linguist, Indiana Univ., Bloomington, IN 47403-2608, jonwashi@indiana.edu)

It has been argued that Kazakh primarily distinguishes its anterior ("front") vowels from its posterior ("back") vowels through retraction of the tongue root. This analysis is at odds with the traditional assumption that the anteriority of Kazakh vowels is contrasted by tongue body position. The present study uses ultrasound imaging to investigate the extent to which the position of the tongue root and the tongue body are involved in the anteriority contrast in Kazakh. Native speakers of Kazakh were recorded reading words (in carrier sentences) containing target vowels, which were controlled for adjacent consonants and metrical position. An audio recording was also made of these sessions. Frames containing productions of the target vowels were extracted from the ultrasound video and the imaged surface of the tongue was manually traced. Analyses of tongue root and body position were analyzed for each vowel and will be presented together with formant measurements from the audio recordings.

4:45

**1pSCa12. Vowel production in sighted children and congenitally blind children.** Lucie Menard and Christine Turgeon (Linguist, Universite du PQ a Montreal, CP 8888, succ. Centre-Ville, Montreal, QC H3C 3P8, Canada, menard.lucie@uqam.ca)

It is well known that vision plays an important role in speech perception. At the production level, we have recently shown that speakers with congenital visual deprivation produce smaller displacements of the lips (visible articulator) compared to their sighted peers [L. Ménard, C. Toupin, S. Baum, S. Drouin, J. Aubin, and M. Tiede, J. Acoust. Soc. Am. 134, 2975–2987 (2013)]. To further investigate the impact of visual experience on the articulatory gestures used to produce intelligible speech, a speech production study was conducted with blind and sighted school-aged children. Eight congenitally blind children (mean age: 7 years old, from 5 years to 11 years) and eight sighted children (mean age: 7 years old, from 5 years to 11 years) were recorded using a synchronous ultrasound and Optotrak imaging system to record tongue and lip positions. Repetitions of the French vowels /i/, /a/, and /u/ were elicited in a /bVb/ sequence in two prosodic conditions: neutral and under contrastive focus. Tongue contours, lip positions, and formant values were extracted. Acoustic data show that focused syllables are less differentiated from their unfocused counterparts in blind children than in sighted children. Trade-offs between lip and tongue positions are examined.

# Session 1pSCb

## Speech Communication: Issues in Cross Language and Dialect Perception (Poster Session)

Tessa Bent, Chair
*Dept. of Speech and Hearing Sciences, Indiana Univ., Bloomington, IN 47405*

All posters will be on display from 1:00 p.m. to 5:00 p.m. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 1:00 p.m. to 3:00 p.m. and contributors of even-numbered papers will be at their posters from 3:00 p.m. to 5:00 p.m.

### *Contributed Papers*

**1pSCb1. Cross-language identification of non-native lexical tone.** Jennifer Alexander and Yue Wang (Dept. of Linguist, Simon Fraser Univ., 9201 Robert C Brown Hall Bldg., 8888 University Dr., Burnaby, BC V5A 1S6, Canada, jennifer_alexander@sfu.ca)

We extend to lexical-tone systems a model of second-language perception, the Perceptual Assimilation Model (PAM) (Best & Tyler, 2007), to examine whether native-language lexical-tone experience influences identification of novel tone. Native listeners of Cantonese, Thai, Mandarin, and Yoruba hear six CV syllables, each produced with the three phonemic Yoruba tones (High-level/H, Mid-level/M, Low-level/L), presented randomly three times. In a 3-AFC task, participants indicate a syllable's tone by selecting from a set of arrows the one that illustrates its pitch trajectory. Accuracy scores (proportion correct) were submitted to a two-way rANOVA with L1-Group (x4) as the between-subjects factor and Tone (x3) as the within-subjects factor. There was no main effect of Tone or Group. The Tone-by-Group interaction was significant ($p = 0.031$) but driven by one group: Thai listeners identified H and M more accurately than L (both $p < 0.05$), though L accuracy was above chance (59%; chance = 33.33%). Tone-error patterns indicate that Thai listeners primarily confused L with M (two-way L1-Group x Response-pattern rANOVA $p < 0.05$). Overall, despite their different tonal-L1 backgrounds, listeners performed comparably. As predicted by the PAM, listeners attended to gradient phonetic detail and acoustic cues relevant to L1 phoneme distinctions (F0 height/direction) in order to classify non-native contrasts. [NSF grant #0965227.]

**1pSCb2. Spectral and duration cues of English vowel identification for Chinese-native listeners.** Sha Tao, Lin Mi, Wenjing Wang, Qi Dong (Cognit. Neurosci. and Learning, Beijing Normal Univ., State Key Lab for Cognit. Neurosci. and Learning, Beijing Normal University, Beijing 100875, China, taosha@bnu.edu.cn), and Chang Liu (Commun. Sci. and Disord., The Univ. of Texas at Austin, Austin, TX)

This study was to investigate how Chinese-native listeners use spectral and duration cues for English vowel identification. The first experiment was to examine whether Chinese-native listeners' English vowel perception was related to their sensitivity to the change of vowel formant frequency that is a critical spectral cue to vowel identification. Identification of 12 isolated American English vowels was measured for 52 Chinese college students in Beijing. Thresholds of vowel formant discrimination were also examined for these students. Results showed that there was a significantly moderate correlation between Chinese college students' English vowel identification and their thresholds of vowel formant discrimination. That is, the lower vowel formant threshold of listeners, the better vowel identification. However, the moderate correlation between vowel identification and formant discrimination suggested some other factors accounting for the individual variability in English vowel identification for Chinese-native listeners. In Experiment 2, vowel identification was measured with and without duration cues, showing that vowel identification was reduced by 5.1% when duration cue was removed. Further analysis suggested that for the listeners who depended less on duration cue, the better thresholds of formant discrimination, the higher scores of vowel identification, but no such correlation for listeners who used duration cues remarkably.

**1pSCb3. The influence of lexical status in the perception of English allophones by Korean learners.** Kyung-Ho Kim and Jeong-Im Han (English, Konkuk Univ., 120 Neungdong-ro, Gwangjin-gu, Seoul 143-701, South Korea, gabrieltotti88@gmail.com)

This study investigated whether the allophonic contrast in the second language (L2) may require contact with the lexicon to influence the perception. Given that English medial voiceless stops occur with aspiration in stressed, but without aspiration in unstressed syllables, Korean learners of English were tested for aspirated and unaspirated allophones of /p/ for perceptual preference in appropriate and inappropriate stress contexts in the second syllable of disyllabic words. The stimuli included four types of non-words and eight pairs of real words (four pairs each for high-frequency and low-frequency words), and participants were asked to judge the perceptual preference of each token on a 7-scale (1 = a bad example, 7 = a good example). The results demonstrated that in tests with non-words, there was no significant difference in the ratings as a function of context appropriateness (e.g., [ípɚ] vs. [íphɚ]), with higher rankings for initially-stressed words. By contrast, in real words, participants preferred the correct allophones (e.g., [képɚ] vs. [képhɚ] "caper"). The frequency of real words further showed a significant effect. This finding suggests that allophony in L2 is driven by lexicality (Whalen *et al*., 1997). Exemplar theory (Pierrehumbert 2001, 2002) provides a more effective means of modeling this finding than do traditional approaches.

**1pSCb4. The perception of English coda obstruents by Mandarin and Korean second language learners.** Yen-Chen Hao (Modern Foreign Lang. and Literatures, Univ. of Tennessee, 510 14th St. #508, Knoxville, TN 37916, yenchenhao@gmail.com) and Kenneth de Jong (Linguist, Indiana Univ., Bloomington, IN)

This study investigates the perception of English obstruents by learners whose native language is either Mandarin, which does not permit coda obstruents, or Korean, which neutralizes laryngeal and manner contrasts into voiceless stop codas. The stimuli are native productions of eight English obstruents /p b t d f v $\theta$ ð/ combined with the vowel /ɑ/ in different prosodic contexts. Forty-one Mandarin and 40 Korean speakers identified the consonant from the auditorily presented stimuli. The results show that the two groups do not differ in their accuracy in the onset position, indicating that they are comparable in their proficiency. However, the Mandarin speakers are more accurate in the coda position than the Koreans. When the fricatives and stops are analyzed separately, it shows that the two groups do not differ with fricatives, yet the Mandarin speakers are more accurate than the Koreans with stops. These findings suggest that having stop codas in their L1 does not necessarily facilitate Koreans' acquisition of the L2 sounds. Despite their L1 differences, the two groups display very similar perceptual biases in their error patterns. However, not all of them can be explained by L1 transfer or universal markedness, suggesting other language-independent factors in L2 perception.

**1pSCb5. Effect of phonetic training on the perception of English consonants by Greek speakers in quiet and noise conditions.** Angelos Lengeris and Katerina Nicolaidis (Theor. and Appl. Linguist, Aristotle Univ. of Thessaloniki, School of English, Aristotle University, Thessaloniki 541 24, Greece, lengeris@enl.auth.gr)

The present study employed high-variability phonetic training (multiple words spoken by multiple talkers) to improve the identification of English consonants by native speakers of Greek. The trainees completed five sessions of identification training with feedback for seven English consonants (contrasting voiced vs. voiceless stops and alveolar vs. postalveolar fricatives) each consisting of 198 trials with a different English speaker in each session. Another group of Greek speakers served as controls, i.e., completed the pre/post test but received no training. Pre/post tests included English consonant identification in quiet and noise. In the noise condition, participants identified consonants in the presence of a competing English speaker at a signal-to-noise ratio of -2dB. The results showed that training significantly improved English consonant perception for the group that received training but not for the control group in both quiet and noise. The results add to the existing evidence that supports the effectiveness of the high-variability approach to second-language segmental training.

**1pSCb6. Perceptual warping of phonetic space applies beyond known phonetic categories: Evidence from the perceptual magnet effect.** Bozena Pajak (Brain & Cognit. Sci., Univ. of Rochester, 1735 N Paulina St. Apt. 509, Chicago, Illinois 60622, bpajak@bcs.rochester.edu), Page Piccinini, and Roger Levy (Linguist, Univ. of California, San Diego, San Diego, CA)

What is the mental representation of phonetic space? Perceptual reorganization in infancy yields a reconfigured space "warped" around native-language (L1) categories. Is this reconfiguration entirely specific to L1 category inventory? Or does it apply to a broader range of category distinctions that are non-native, yet discriminable due to being defined by phonetic dimensions informative in the listener's L1 (Bohn & Best, 2012; Pajak, 2012)? Here we address this question by studying perceptual magnets, which involve attrition of within-category distinctions and enhancement of distinctions across category boundaries (Kuhl, 1991). We focus on segmental length, known to yield L1-specific perceptual magnets: e.g., L1-Finnish listeners have one for [t]/[tt], but L1-Dutch listeners, who lack (exclusively) length-based contrasts, do not (Herren & Schouten, 2008). We tested 31 L1-Korean listeners in an AX discrimination task for [n]-[nn] and [f]-[ff] continua. Korean listeners have been shown to discriminate both (Pajak, 2012), despite only having the former set in the inventory. We found perceptual magnets for both continua, demonstrating that perceptual warping goes beyond the specific L1 categories: when a phonetic dimension is informative for contrasting some L1 categories, perceptual warping applies not only to the tokens from those categories, but also to that dimension more generally.

**1pSCb7. Language mode effects on second language categorical perception.** Beatriz Lopez Prego and Allard Jongman (Linguist, Univ. of Kansas, 1145 Pennsylvania St., Lawrence, KS 66044, lopezb@ku.edu)

This study investigates the perception of the /b/-/p/ voicing contrast in English and Spanish by native English listeners, native Spanish listeners, and highly proficient Spanish-speaking second-language (L2) learners of English with a late onset of acquisition (mean = 10.8) and at least three-year residence in an English-speaking environment. Participants completed a forced-choice identification task where they identified target syllables in a Voice Onset Time (VOT) continuum as "pi" or "bi." They listened to 10 blocks of 19 equidistant steps ranging from + 88 ms-VOT to -89 ms-VOT. Between blocks, subjects read and wrote responses to language background questions, thus actively processing the target language. Monolinguals completed the task in their native language (L1). L2 learners completed the task once in their L1 and once in their L2, thus providing a manipulation of "language mode" (Grosjean, 2001). The results showed that L2 learners' category boundary in English did not differ from that of monolingual English listeners, but their category boundary in Spanish differed from that of monolingual Spanish listeners and from their own category boundary in English. These results suggest that the language mode manipulation was successful and that L2 learners can develop new phonetic categories, but this may have an impact on their L1 categories.



**1pSCb8. Processing of English-accented Spanish voice onset time by Spanish speakers with low English experience.** Fernando Llanos (School of Lang. and Cultures, Purdue Univ., Stanley Coulter Hall, 640 Oval Dr., West Lafayette, IN 47907, fllanos@purdue.edu) and Alexander L. Francis (Speech, Lang. & Hearing Sci., Purdue Univ., West Lafayette, IN)

Previous research (Llanos & Francis, 2014) shows that the processing of foreign accented speech sounds can be affected by listeners' familiarity with the language that causes the accent. Highly familiar listeners treat foreign accented sounds as foreign sounds while less familiar listeners treat them natively. The present study tests the hypothesis that less familiar listeners may nevertheless be able to apply foreign categorization patterns to accented words by recalibrating phonetic expectations according to acoustic information provided by immediate phonetic context. Two groups of Spanish native speakers with little English experience will identify tokens drawn from a digitally edited VOT continuum ranging from baso "glass" (-60 ms VOT) to paso "step" (60 ms VOT). Tokens are embedded in a series of Spanish words beginning with /b/ and /p/ to provide phonetic context. In the English-accented condition, context words are digitally modified to exhibit English-like VOT values for /b/ (10 ms) and /p/ (60 ms). In the Spanish condition, these tokens are edited to exhibit prototypical Spanish /b/ (-90 ms) and /p/ (10 ms) VOT values. If listeners can accommodate foreign accented sounds according to expectations provided by immediate phonetic context, then listeners' VOT boundary in the English-accented condition should be significantly higher than in the Spanish condition.

**1pSCb9. Amount of exposure and its effect on perception of second language front vowels in English.** Andrew Jeske (Linguist, Univ. of Pittsburgh, 3211 Brereton St., Pittsburgh, PA 15219, arjeske@gmail.com)

Experience with a second language (L2) has been shown to positively affect learners' perception of L2 sounds. However, few studies have focused on how the amount of L2 exposure in foreign language classrooms impacts perception of L2 sounds during the incipient stages of language learning in school-age children. To determine what effect, if any, the amount of L2 exposure has on perception, 64 students from a Spanish-English bilingual elementary school and 60 students from two non-bilingual elementary schools participated in an AX Categorical Discrimination task, which contained tokens of five English front vowels: /i ɪ e ɛ æ/. Results show that students from the bilingual school earned perception scores significantly higher than those earned by the students from the non-bilingual school ($p = 0.002$). However, an ANOVA found there to be no significant simple main effect for grade or significant correlation between grade level and school type. The bilingual school students perceived all within-category word pairings (e.g., bat-bat) significantly more accurately than the non-bilingual school students suggesting that increased, early exposure to an L2 may heighten one's ability to disregard irrelevant, interpersonal phonetic differences and lead to a within-category perceptual advantage over those with less L2 exposure early on.

**1pSCb10. Does second language experience modulate perception of tones in a third language?** Zhen Qin and Allard Jongman (Linguist, Univ. of Kansas, 1541 Lilac Ln., Blake Hall, Rm. 427, Lawrence, KS 66045, qinzhenquentin2@ku.edu)

Previous studies have shown that English speakers pay attention to pitch height rather than direction, whereas Mandarin speakers are more sensitive to pitch direction than height in perception of lexical tones. The present study addresses if a second language (L2, i.e., Mandarin) overrides the influence of a native language (L1, i.e., English) in modulating listeners' use of pitch cues in the perception of tones in a third language (L3, i.e., Cantonese). English-speaking L2 learners (L2ers) of Mandarin constituted the

target group. Mandarin speakers and English speakers without knowledge of Mandarin were included as control groups. In Experiment 1, all groups, naïve to Cantonese tones, discriminated Cantonese tones by distinguishing either a contour tone from a level tone (pitch direction pair) or a level tone from another level tone (pitch height pair). The results showed that L2ers patterned differently from both control groups with regard to pitch cues under the influence of L2 experience. The acoustics of the tones also affected all listeners' discrimination. In Experiment 2, L2ers were instructed to identify Mandarin tones to measure their sensitivity to L2 tones. The results showed that L2ers' sensitivity to L2 tones is not necessarily correlated with their perception of L3 tones.

**1pSCb11. Does early foreign language learning in school affect phonemic discrimination in adulthood?** Tetsuo Harada (School of Education, Waseda Univ., 1-6-1 Nishi Waseda, Shinjuku, Tokyo 169-8050, Japan, tharada@waseda.jp)

Long-term effects of early foreign language learning with a few hours' classroom contact per week on speech perception are controversial: some studies show age effects of minimal English input in childhood on phonemic perception in adulthood, but others don't (e.g., Lin *et al.*, 2004). This study investigated effects of a younger starting age in a situation of minimal exposure on perception of English consonants under noise conditions. The listeners were two groups of Japanese university students: early learners ($n = 21$) who started studying English in kindergarten or elementary school, and late learners ($n = 24$) who began to study in junior high school. The selected target phonemes were word-medial approximants (/l, r/). Each nonword (i.e., *ala*, *ara*), produced by six native talkers, was combined with speech babble at the signal-to-noise ratios (SNRs) of 8 dB (medium noise) and 0 dB (quite high noise for L2 listeners). A discrimination test was given in the ABX format. Results showed that the late learners discriminated /l/ and /r/ better than the early learners regardless of the noise conditions and talker differences ($p < 0.05$). A multiple regression analysis revealed that length of learning and English use could contribute to their discrimination ability.

**1pSCb12. The identification of American English vowels by native speakers of Japanese before three nasal consonants.** Takeshi Nozawa (Lang. Education Ctr., Ritsumeikan Univ., 1-1-1 Nojihigashi, Kusatsu 525-8577, Japan, t-nozawa@ec.ritsumei.ac.jp)

Native speakers of Japanese identified American English vowels that are uttered before three nasal consonants /m, n, ŋ/ and three oral stop consonants /b, d, g/. Of the seven vowels /i, ɪ, eɪ, ɛ, æ, ɑ, ʌ/, /æ/ was generally less accurately identified before nasal consonants than before oral stop consonants, and this tendency was stronger when /ŋ/ follows. This tendency is probably attributed to the extended raising of /æ/ before /ŋ/ and the Japanese listeners' limited sensitivity to differentiate three nasal phonemes in coda position. /ɪ/, on the other hand, was identified more correctly before /ŋ/ than before the other two nasal consonants, also probably because the vowel is raised before /ŋ/. This vowel was more often misidentified as /ɛ/ before /m/ and /n/. /ɑ/ and /ʌ/ were less accurately identified before stop consonants, but after nasal consonants, /ʌ/ was more often misidentified as /ɑ/. /ɑ/ and /ʌ/ may sound alike to Japanese listeners in every context, but before nasal contexts, both of these vowels may sound closer to the Japanese vowel /o/. The results generally revealed that identification accuracy cannot be solely accounted for in terms of the place of articulation of the following consonant.

**1pSCb13. Effects of beliefs about first language orthography on second language vowel perception.** Mara Haslam (Dept. of Lang. Education, Stockholm Univ., S:t Ansgars väg 4, Solna 16951, Sweden, mara.haslam@gmail.com)

Recent research has identified that L1 orthography can affect perception of vowels in a second language (e.g., Escudero and Wanrooij, 2010). The present study investigates the effect that participants' beliefs about orthography have on their ability to perceive vowels in a second language. English- and Polish-speaking learners of Swedish have to encounter some new vowel sounds and also the characters that are used to represent them, e.g., å, ä, and ö. New survey data of native speakers of English, Polish, and Swedish confirm that L1 English speakers see these characters like these as familiar letters with diacritics, while L1 Swedish and L1 Polish speakers tend to see these types of characters as different characters of the alphabet. These differing beliefs about orthography may cause English speakers to confuse the vowels represented in Swedish by the characters å, ä and ö with vowels represented by the characters a, a, and o, respectively, while Polish speakers would not be similarly affected. Results of a Swedish vowel perception study conducted with native speakers of English and Polish after exposure to Swedish words containing these characters will be presented. These results contribute to increasing knowledge about the relationship between L1 orthography and L2 phonology.

**1pSCb14. A preliminary investigation of the effect of dialect on the perception of Korean sibilant fricatives.** Jeffrey J. Holliday (Second Lang. Studies, Indiana Univ., 1021 E. Third. St., Memorial Hall M03, Bloomington, IN 47405, jjhollid@indiana.edu) and Hyunjung Lee (English, Hankyong National Univ., Anseong, Gyeonggi-do, South Korea)

Korean has two sibilant fricatives, /$s^h$/ and /s*/, that are phonologically contrastive in the Seoul dialect but are widely believed to be phonetically neutralized in the Gyeongsang dialects spoken in southeastern South Korea, with both fricatives being acoustically realized as [$s^h$]. The current study investigated the degree to which the perception of these fricatives by Seoul listeners is affected by knowledge of the speaker's dialect. In the first task, the stimuli were two fricative-initial minimal pairs (i.e., four words) produced by 20 speakers each from Seoul and Gyeongsang. Half of the 18 listeners were told that the speakers were from Seoul, and the other half were told they were from Gyeongsang. Listeners identified the 160 word-initial fricatives and provided a goodness rating for each. It was found that neither the speaker's actual dialect nor the primed dialect had a significant effect on either identification accuracy or listeners' goodness ratings. In a second task, listeners identified tokens from a seven-step continuum from [sada] to [s*ada]. It was found that listeners who were primed for Gyeongsang dialect were more likely to perceive tokens as /s*/ than listeners primed for Seoul, which may reflect a dialect-based hypercorrective perceptual bias.

**1pSCb15. Language is not destiny: Task-specific factors, and not just native language perceptual biases, influence foreign sound categorization strategies.** Jessamyn L. Schertz and Andrew Lotto (Univ. of Arizona, Douglass 200, Tucson, AZ 85721, jschertz@email.arizona.edu)

Listeners were trained to distinguish two novel classes of speech sounds differing in both Voice Onset Time (VOT) and fundamental frequency at vowel onset (f0). One group was shown Korean orthography during the training period ("symbols" group) and the other English orthography ("letters" group). During a subsequent test phase, listeners classified sounds with mismatched VOT and f0. The two groups relied on different cues to categorize the contrast: those exposed to symbols used f0, while those exposed to letters used VOT. A second experiment employed the same paradigm, but the two dimensions defining the contrast were closure duration (instead of f0) and VOT. In this more difficult experiment, successful listeners in the "letters" group again classified the hybrid stimuli based on VOT, while the single listener in the "symbols" group who passed the learning criterion used closure duration. In both experiments, subjects showed different categorization patterns based on orthography used in the presentation, even though orthography was irrelevant for the experimental task. Listeners relied on VOT when the stimuli were presented with English, but not foreign, orthography, showing that task-related information (as opposed to native language biases alone) can direct attention to different acoustic cues in foreign contrast classification.

**1pSCb16. Generational difference in the perception of high-toned [il] in Seoul Korean.** Sunghye Cho (Univ. of Pennsylvania, 3514 Lancaster Ave., Apt. 106, Philadelphia, PA 19104, csunghye@sas.upenn.edu)

A word-initial [il] is most frequently H-toned in Seoul Korean (SK) when it means one, out of three homophones, one, day, and work (Jun & Cha, 2011). However, Cho (2014) finds that 25% of teenagers always produce [il] with a H tone, regardless of its meaning. This paper examines how young SK speakers perceive the phenomenon. Thirty-seven SK speakers (aged 14–29) participated in two identification tasks, hearing only [il] in the first task and four [il]-initial minimal pairs in the second task. All target words were manipulated into five pitch levels with 30 Hz intervals. In the first task, the 20s group identified [il] as one 70% of the time at higher pitch levels, while the teenagers identified [il] as one about 50% of the time at all pitch levels. In the second task, the 20s group showed a categorical perception, identifying [il]-initial words as one only at higher pitch levels, while the teenagers did not. The results suggest that the teenagers are aware that some peers always produce [il] with a H tone. It explains that the 20s group could identify the meanings of [il] depending on the pitch, while the teenagers could not.

**1pSCb17. The effect of perceived talker race on phonetic imitation of pin-pen words.** Qingyang Yan (Linguist, The Ohio State Univ., 591 Harley Dr., Apt. 10, Columbus, OH 43212, yan@ling.ohio-state.edu)

The current study investigated the phonetic imitation of the PIN-PEN merger by nonmerged participants. An auditory shadowing task was used to examine how participants changed their /ɪ/ and /ɛ/ productions after auditory exposure to merged and nonmerged voices. Black and white talker photos were used as visual cues to talker race. The pairing of voices (merged and nonmerged) with the talker photos (black and white) was counterbalanced across participants. A third group of participants completed the task without talker photos. Participants' explicit talker attitudes were assessed by a questionnaire, and their implicit racial attitudes were measured by an Implicit Association Task. Nonmerged participants imitated the PIN-PEN merger, and the degree of imitation varied depending on the experimental condition. The merged voice elicited more imitation when it was presented without a talker photo or with the black talker photo than with the white talker photo. No effect of explicit talker attitudes or implicit racial attitudes on the degree of imitation was observed. These results suggest that phonetic imitation of the PIN-PEN merger is more complex than an automatic response to the merged voice and that it is mediated by perceived talker race.

**1pSCb18. Foreign-accent discrimination with words and sentences.** Eriko Atagi (Volen National Ctr. for Complex Systems, Brandeis Univ., Volen National Ctr. for Complex Systems, MS 013, Brandeis University, 415 South St., Waltham, MA 02454-9110, eatagi@brandeis.edu) and Tessa Bent (Dept. of Speech & Hearing Sci., Indiana Univ., Bloomington, IN)

Native listeners can detect a foreign accent in very short stimuli; however, foreign-accent detection is more accurate with longer stimuli (Park, 2008; Flege, 1984). The current study investigated native listeners' sensitivity to the characteristics that differentiate between accents—both foreign versus native accents and one foreign accent versus another—in words and sentences. Listeners heard pairs of talkers reading the same word or sentence and indicated whether the talkers had the same or different native language backgrounds. Talkers included two native talkers (Midland dialect) and six nonnative talkers from three native language backgrounds (German, Mandarin, and Korean). Sensitivity varied significantly depending on the specific accent pairings and stimulus type. Listeners were most sensitive when the talker pair included a native talker, but could detect the difference between two nonnative accents. Furthermore, listeners were generally more sensitive with sentences than with words. However, for one nonnative pairing, listeners exhibited higher sensitivity with words; for another, listeners' sensitivity did not differ significantly across stimulus types. These results suggest that accent discrimination is not simply influenced by stimulus length. Sentences may provide listeners with opportunities to perceive similarities between nonnative talkers, which are not salient in single words. [Work supported by NIDCD T32 DC00012.]

**1pSCb19. Stimulus length and scale label effects on the acoustic correlates of foreign accent ratings.** Elizabeth A. McCullough (Linguist, Ohio State Univ., 222 Oxley Hall, 1712 Neil Ave., Columbus, OH 43210, eam@ling.ohio-state.edu)

Previous studies have investigated acoustic correlates of accentedness ratings, but methodological differences make it difficult to compare their results directly. The present experiment investigated how choices about stimulus length and rating scale labels influence the acoustic correlates of listeners' rating responses. Four conditions crossed two stimulus lengths (CV syllable vs. disyllabic word) with two sets of rating labels ("no foreign accent"/"strong foreign accent" vs. "native"/"not native"). Monolingual American English listeners heard samples of English from native speakers of American English, Hindi, Korean, Mandarin, and Spanish and indicated their responses on a continuous rating line. Regression models evaluated the correlations between listeners' ratings and a variety of acoustic properties. Patterns for accentedness and non-nativeness ratings were identical. VOT, F1, and F2 correlated with ratings on all stimuli, but vowel duration correlated with ratings on disyllabic word stimuli only. If vowel duration is interpreted as a reflection of global temporal properties, this result suggests that listeners may perceive such properties in utterances as short as two syllables. Thus, stimulus design is vital in identifying components of foreign accent perception that are related to differences between a talker's first and second languages as opposed to components that are related to general fluency.



**1pSCb20. Language proficiency, context influence foreign-accent adaptation.** Cynthia P. Blanco (Linguist, Univ. of Texas at Austin, 305 E. 23rd St., Austin, TX 78712, cindyblanco@utexas.edu), Hoyoung Yi (Commun. Sci. & Disord., Univ. of Texas at Austin, Austin, TX), Elisa Ferracane, and Rajka Smiljanic (Linguist, Univ. of Texas at Austin, Austin, TX)

Listeners adapt quickly to changes in accent (Bradlow & Bent, 2003; Clarke & Garrett, 2004; inter alia). The cause of this brief delay may be due to the cost of processing accented speech, or may reflect a surprise effect associated with task expectations (Floccia *et al.*, 2009). The present study examines a link between accent familiarity and processing delays with listeners who have varying degrees of familiarity with target languages: monolingual Texans with little or no formal exposure to Spanish, early Spanish-English bilinguals, and Korean learners of English. Participants heard four blocks of English sentences—Blocks 1 and 4 were produced by two native speakers of American English, and Blocks 2 and 3 were produced by native speakers of Spanish or Korean- and responded to written probe words. All listener groups responded more slowly after an accent change; however, the degree of delay varied with language proficiency. L1 Korean listeners were less delayed by Korean-accented speech than the other listeners, while changes to Spanish-accented speech were processed most slowly by Spanish-English bilinguals. The results suggest that adaptation to foreign-accented speech depends on language familiarity and task expectations. The processing delays are analyzed in light of intelligibility and accentedness measures.

**1pSCb21. When two become one—Orthography helps link two free variants to one lexical entry.** Chung-Lin Yang (Linguist, Indiana Univ.- Bloomington, Memorial Hall 322, 1021 E 3rd St., Bloomington, IN 47408, cy1@indiana.edu) and Isabelle Darcy (Second Lang. Studies, Indiana Univ.- Bloomington, Bloomington, IN)

L2 learners can become better at distinguishing an unfamiliar contrast by knowing the corresponding orthographic forms (e.g., Escudero *et al.*, 2008). We ask whether learners could associate two free variants with the same lexical entry when the orthographic form was provided during learning. American learners learned an artificial language where [p]-[b] were in free variation (both were spelled as <p>) (test condition) while [t]-[d] were contrastive (control condition), or vice-versa ([t]-[d] in test, counterbalanced across subjects). Using a word-learning paradigm modified from Hayes-Harb *et al.* (2010), in the learning phase, participants heard novel words paired with pictures. One subgroup of learners saw the spellings as well ("Orth+"), while another did not (i.e., auditory only, "Orth−"). Then in a picture-auditory word matching task, the new form of the word was paired with the original picture. Orth + learners were expected to be more accurate at accepting the variant as the correct label for the original test item than Orth−. The results showed that Orth + learners detected and learned the [p]-[b] free variation significantly better than Orth− ($p < 0.05$), but not the [t]-[d] free variation. Thus, the benefit of orthography in speech learning could vary depending on the specific contrasts at hand.

# Session 1pUW

## Underwater Acoustics: Understanding the Target/Waveguide System–Measurement and Modeling II

Aubrey L. Espana, Chair
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**Chair's Introduction—1:25**

### *Invited Paper*

**1:30**

**1pUW1. Mapping bistatic scattering from spherical and cylindrical targets using an autonomous underwater vehicle in BAYEX'14 experiment.** Erin M. Fischell, Stephanie Petillo, Thomas Howe, and Henrik Schmidt (Mech. Eng., MIT, 77 Massachusetts Ave., 5-204, Cambridge, MA 02139, emf43@mit.edu)

In May 2014, the MIT Laboratory for Autonomous Marine Sensing Systems (LAMSS) participated in the BAYEX'14 experiment with the goal of collecting full bistatic data sets around proud spherical and cylindrical targets for use in real-time autonomous target localization and classification. The BAYEX source was set to insonify both targets, and was triggered to ping at the start of each second using GPS PPS. The MIT Bluefin 21 in. AUV Unicorn, fitted with a 16-element nose array, was deployed in broadside sampling behaviors to collect the bistatic scattered data set. The AUV's Chip Scale Atomic Clock was synchronized to GPS on the surface, and the data was logged using a PPS triggered analog to digital conversion system to ensure synchronization with the source. The MIT LAMSS operational paradigm allowed the vehicle to be unpacked, tested and deployed over the brief three-day interval available for operations. MOOS-IvP and acoustic communication enabled the group to command AUV mission changes in situ based on data collection needs. During data collection, the vehicle demonstrated real-time signal processing and target localization, and the bistatic datasets were used to demonstrate real-time target classification in simulation. [Work supported by ONR Code 322OA.]

### *Contributed Papers*

**1:50**

**1pUW2. Elastic features visible on canonical targets with high frequency imaging during the 2014 St. Andrews Bay experiments.** Philip L. Marston (Phys. and Astronomy Dept., Washington State Univ., Pullman, WA 99164-2814, marston@wsu.edu), Timothy M. Marston, Steven G. Kargl (Appl. Phys. Lab., Univ. of Washington, Seattle, WA), Daniel S. Plotnick (Phys. and Astronomy, Washington State Univ., Pullman, WA), Aubrey Espana, and Kevin L. Williams (Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

During the 2014 St. Andrews Bay experiments some canonical metallic targets (a hollow sphere and some circular cylinders) were viewed with a synthetic aperture sonar (SAS) capable of acquiring data using a 110–190 kHz chirped source. The targets rested on mud-covered sand and were typically at a range of 20 m. Fast reversible SAS processing using an extension of line-scan quasi-holography [K. Baik, C. Dudley, and P. L. Marston, J. Acoust. Soc. Am. 130, 3838–3851 (2011)] was used to extract relevant signal content from images. The significance of target elastic responses in extracted signals was evident from the frequency response and/or the time-domain response. For example, the negative group velocity guided wave enhancement of the backscattering by the sphere was clearly visible near 180 kHz. [For a ray model of this type of enhancement see: G. Kaduchak, D. H. Hughes, and P. L. Marston, J. Acoust. Soc. Am. 96, 3704–3714 (1994).] In another example, the timing of a sequence of near broadside echoes from a solid aluminum cylinder was consistent with reflection and internal reverberation of elastic waves. These observations support the value of combining reversible imaging with models interpreted using rays. [Work supported by ONR and SERDP.]

**2:05**

**1pUW3. Boundary enhanced coupling processes for rotated horizontal solid aluminum cylinders: Helical rays, synthetic aperture sonar images, and coupling conditions.** Jon R. La Follett (Shell International Exploration and Production Inc., Houston, TX) and Philip L. Marston (Phys. and Astronomy Dept., Washington State Univ., Phys. and Astronomy Dept., Washington State Univ., Pullman, WA 99164-2814, marston@wsu.edu)

Experiments with solid aluminum cylinders placed near a flat free surface provide insight into scattering processes relevant to other flat reflecting boundaries [J. R. La Follett, K. L. Williams, and P. L. Marston, J. Acoust. Soc. Am. 130, 669–672 (2011); J. R. La Follett, Ph.D. thesis, WSU (2010)]. This presentation concerns the coupling to surface guided leaky Rayleigh waves that have been shown to contribute significantly to backscattering by solid metallic cylinders [K. Gipson and P. L. Marston, J. Acoust. Soc. Am. 106, 1673–1689 (1999)]. The emphasis here is on horizontal cylinders rotated about a vertical axis away from broadside viewed at grazing incidence. The range of rotation angles for which helical rays can contribute is limited in the free field by the cylinder's length [F. J. Blonigen and P. L. Marston, J. Acoust. Soc. Am. 112, 528–536 (2002)]. Some examples of surface enhanced backscattering may be summarized as follows. In agreement with geometrical considerations, the angular range for coupling to helical rays may be significantly extended when a short cylinder is adjacent to a flat surface. In addition, the presence of a flat surface splits synthetic aperture sonar (SAS) image features from various guided wave mechanisms on rotated cylinders. [Work supported by ONR.]

## *Invited Papers*

### 2:20

**1pUW4. Denoising structural echoes of elastic targets using spatial time–frequency distributions.** Karim G. Sabra (Mech. Eng., Georgia Inst. of Technol., 771 Ferst Dr., NW, Atlanta, GA 30332-0405, karim.sabra@me.gatech.edu)

Structural echoes of underwater elastic targets, used for detection and classification purposes, can be highly localized in the time–frequency domain and can be aspect-dependent. Hence, such structural echoes recorded along a distributed (synthetic) aperture, e.g., using a moving receiver platform, would not meet the stationarity and multiple snapshots requirements of common subspace array processing methods used for denoising array data based on their estimated covariance matrix. To handle these scenarios, a generalized space–time–frequency covariance matrix can be computed from the single-snapshot data using Cohen's class time-frequency distributions between all sensor data pairs. This space–time–frequency covariance matrix automatically accounts for the inherent coherence across the time-frequency plane of the received nonstationary echoes emanating from the same target. Hence, identifying the signal's subspace from the eigenstructure of this space–time–frequency covariance matrix provides a means for denoising these non-stationary structural echoes by spreading the clutter and noise power in the time–frequency domain. The performance of the proposed methodology will be demonstrated using numerical simulations and at-sea data.



### 2:40

**1pUW5. Measurements and modeling of acoustic scattering from targets in littoral environments.** Harry J. Simpson (Physical Acoust. Branch, Naval Res. Lab., 4555 Overlook Ave. SW, Washington, VA20375, harry.simpson@nrl.navy.mil), Zackary J. Waters, Timothy J. Yoder, Brian H. Houston (Physical Acoust. Branch, Naval Res. Lab., Washington, DC), Kyrie K. Jig, Roger R. Volk (Sotera Defense Solution, Crofton, MD), and Joseph A. Bucaro (Excet, Inc., Springfield, VA)

Broadband laboratory and at-sea measurements systems have been built by NRL to quantify the acoustic target strength of objects sitting on or in the bottom of littoral environments. Over the past decade, these measurements and the subsequent modeling of the target strength have helped to develop an understanding of how the environment, especially near the bottom interface, impacts the structural acoustic response of a variety of objects. In this talk we will present a set of laboratory, at-sea rail and AUV based back scatter, forward scatter, and propagation measurements with subsequent analysis to understand the impact of the littoral environment. Simple targets such as spheres, along with UXO targets will be discussed. The analysis will be focused on quantifying the changes to target strength as a result of being near the bottom interface. In addition to the traditional backscatter or monosatic target strength, we focus upon efforts to investigate the multi-static scattering from targets. [Work supported by ONR.]

### 3:00–3:15 Break

## *Contributed Papers*

### 3:15

**1pUW6. TREX13 target experiments and case study: Comparison of aluminum cylinder data to combined finite element/physical acoustics modeling.** Kevin Williams, Steven G. Kargl, and Aubrey L. Espana (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, williams@apl.washington.edu)

The apparatus and experimental procedure used during the target portion of TREX13 are described. A primary goal of the TREX13 target experiments was to test the high speed modeling methods developed and previously tested as part of efforts in more controlled environments where the sediment/water interface was flat. At issue is to what extent the simplified physics used in our models can predict the changes seen in acoustic templates (target strength versus angle and frequency) as a function of grazing angle, i.e., the Target-In-the-Environment-Response (TIER), for a target proud on an unprepared "natural" sand sediment interface. Data/model comparisons for a 3 ft. long, 1 ft. diameter cylinder are used as a case study. These comparisons indicate that much of the general TIER dependence is indeed captured and allows one to understand/predict geometries where the broadest band of TIER information can be obtained. This case study indicates the predictive utility of dissecting the target physics at the expense of making the model results "inexact" from a purely finite element, constitutive equation standpoint. [Work supported by ONR and SERDP.]

### 3:30

**1pUW7. Predicting the acoustic response of complicated targets in complicated environments using a hybrid finite element/propagation model.** Aubrey L. Espana, Kevin L. Williams, Steven G. Kargl (Acoust. Dept., Appl. Phys. Lab. - Univ. of Washington, 1013 NE 40th St., Box 355640, Seattle, WA 98105, aespana@apl.washington.edu), Marten J. Nijhof (Acoust. and Sonar, TNO, Den Haag, Netherlands), Daniel S. Plotnick, and Philip L. Marston (Phys. and Astronomy, Washington State Univ., Pullman, WA)

Previous work has shown that hybrid finite element (FE)/propagation models are a viable tool for estimating the Target-In-The-Environment-Response, or TIER, for simple shapes such as cylinders and pipes on a flat, undisturbed sand/water interface [K. L. Williams *et al.*, J. Acoust. Soc. Am 127, 3356–3371 (2010)]. Here we examine their use for more complicated targets located in complicated ocean environments. The targets examined include various munitions and ordnance-like targets, with intricate internal structure and filled with either air or water. A hybrid FE/propagation model is used to predict their TIER on flat, undisturbed sand. Data acquired during the target portion of TREX13 is used to validate the model results. Next, the target response is investigated in a more complicated environment, being partially buried with their axis tilted w.r.t. the flat sand interface. Again model results are validated using TREX13 data, as well as data acquired in a controlled tank experiment. These comparisons highlight the feasibility of using hybrid models for complex target/environment configurations, as well possible limitations due to the effects of multiple scattering.

**3:45**

**1pUW8. A correlation analysis of the Naval Surface Warfare Center Panama City Division's (NSWC PCD) database of simulated and collected target scattering responses focused on automated target recognition.** Raymond Lim, David E. Malphurs, James L. Prater, Kwang H. Lee, and Gary S. Sammelmann (Code X11, NSWC Panama City Div., 110 Vernon Ave, Code X11, Panama City, FL 32407-7001, raymond.lim@navy.mil)

Recently, NSWC PCD participated in a number of computational and experimental efforts aimed at assembling a database of sonar scattering responses encompassing a variety of objects including UXO, cylindrical shapes, and other clutter-type objects. The range of data available on these objects consists of a simulated component generated with 3D finite element calculations coupled to a fast Helmholtz-equation-based propagation scheme, a well-controlled experimental component collected in NSWC PCD's pond facilities, and a component of measurements in realistic underwater environments off Panama City, FL (TREX13 and BayEX14). The goal is to use the database to test schemes for automating reliable separation of these objects into desired classes. Here, we report on an initial correlation analysis of the database projected onto the target aspect vs frequency plane to assess the feasibility of the simulated component against the measured ones, to investigate some basic questions regarding environmental and range effects on class separation, and to try and identify phenomena in this plane useful for classification. [Work supported by ONR and SERDP.]

**4:05**

**1pUW9. Identifying buried unexploded ordnance with structural acoustics based numerically trained classifiers: Laboratory demonstrations.** Zachary J. Waters, Harry J. Simpson, Brian H. Houston (Physical Acoust. - Code 7130, Naval Res. Lab., 4555 Overlook Ave. SW, Bldg 2. Rm. 186, Washington, DC 20375, zachary.waters@nrl.navy.mil), Kyrie Jig, Roger Volk, Timothy J. Yoder (Sotera Defense Solutions Inc., Crofton, MD), and Joseph A. Bucaro (Excet Inc., Springfield, VA)

Strategies for the automated detection and classification of underwater unexploded ordnance (UXO), based upon structural acoustics derived features, are currently being transitioned to autonomous underwater vehicle based sonar systems. The foundation for this transition arose, in part, from extensive laboratory investigations conducted at the Naval Research Laboratory. We discuss the evolution of structural acoustic based methodologies, including research into understanding the free-field scattering response of UXO and the coupling of these objects, under varying stages of burial, to water-saturated sediments. In addition to providing a physics-based understanding of the mechanisms contributing to the scattering response of objects positioned near the sediment–water interface, this research supports the validation of three-dimensional finite-element-based models for large-scale structural–acoustics problems. These efforts have recently culminated with the successful classification of a variety of buried UXO targets using a numerically trained relevance vector machine (RVM) classifier and the discrimination of these targets, under various burial orientations, from several objects representing both natural and manmade clutter. We conclude that this demonstration supports the transition of structural acoustic processing methodologies to maritime sonar systems for the classification of challenging UXO targets. [Work supported by ONR and SERDP.]

**4:25**

**1pUW10. Detection and classification of marine targets buried in the sediment using structural acoustic features.** Joseph Bucaro (Excet, Inc. @ Naval Res. Lab., 4555 Overlook Ave SW, Naval Res. Lab., Washington, DC 20375, joseph.bucaro.ctr@nrl.navy.mil), Brian Houston, Angie Sarkissian, Harry Simpson, Zack Waters (Naval Res. Lab., Washington, DC), Timothy Yoder (Sotera Inc. @ Naval Res. Lab., Washington, DC), and Dan Amon (Naval Res. Lab., Washington, DC)

We present research on detection and classification of underwater targets buried in a saturated sediment using structural acoustic features. These efforts involve simulations using NRL's STARS3D structural acoustics code and measurements in the NRL free-field and sediment pool facilities, off the coast of Duck, NC, and off the Coast of Panama City, FL. The measurements in the sediment pool demonstrated RVM classifiers trained using numerical data on two features—target strength correlation and elastic highlight image symmetry. Measurements off the coast of Duck were inconclusive owing to tropical storms resulting in a damaged projector. Extensive measurements were then carried out in 60 ft. of water in the Gulf using BOSS, an autonomous underwater vehicle with 40 receivers on its wings. The target field consisted of nine simulant-filled UXO and two false targets buried in the sediment and twenty proud targets. The AUV collected scattering data during north/south, east/west, and diagonal flights. We discuss the data analyzed so far from which we have extracted 3-D images and acoustic color constructs for 18 of the targets and demonstrated UXO/false target separation using a high dimensional acoustic color feature. Finally, we present related work involving targets buried in non-saturated elastic sediments. [This work is supported by ONR and SERDP.]

## *Contributed Papers*

**4:45**

**1pUW11. Performance metrics for depth-based signal separation using deep vertical line arrays.** John K. Boyle, Gabriel P. Kniffin, and Lisa M. Zurk (Northwest Electromagnetics and Acoust. Res. Lab. (NEAR-Lab), Dept. of Elec. & Comput. Eng., Portland State Univ., 1900 SW 4th Ave., Ste. 160, Portland, OR 97201, jboyle@pdx.edu)

A publication [McCargar & Zurk, 2013] presented a method for passive depth-separation of signals received on vertical line arrays (VLAs) deployed below the critical depth in the deep ocean. This method, based on a modified Fourier transform of the received signals from submerged targets, makes use of the depth-dependent modulation inherent in the signals due to interference between the direct and surface-reflected acoustic arrivals. Examination of the transform is necessary to determine performance of the algorithm in terms of the minimum target depth and range, array aperture, and temporal sampling. However, traditional expressions for signal sampling requirements (Nyquist sampling theorem) do not directly apply to the measured signal along a target trace due to uneven sampling in vertical angle imposed by the spatiotemporal evolution of the target track as observed on the VLA. In this paper, the effects of this uneven sampling on the ambiguity in the estimated depth (i.e., aliasing) are discussed, and expressions for the maximum snapshot length are presented and validated using simulated data

produced with a normal-mode propagation model. Initial results are presented to show the requirements for snapshot lengths and target trajectories for successful depth separation of slow moving targets at low frequencies.

**5:00**

**1pUW12. Wideband imaging with the decomposition of time reversal operator.** Chunxiao Li, Mingfei Guo, and Huancai Lu (Zhejiang Univ. of Technol., 18# ChaoWang Rd., Hangzhou, Zhejiang, Hangzhou 310014, China, chunxiaoli@zju.edu.cn)

It has been shown that the decomposition of the time reversal operator (DORT) is effective to achieve detection and selectively focusing on pointlike scatterers. Moreover, the multiplicity of the invariant of the time reversal operator for a single extended (non-pointlike) scatterer has been also revealed. In this paper, we investigate the characterization and imaging of the scatterers when an extended scatterer and a pointlike scatterer are simultaneously present. The relationship between the quality of focusing and frequency is investigated by backpropagation of singular vectors using a model of the waveguide in each frequency bin. When the extended scatterer is present, it is shown that the second singular vector can also focus on the target. However, the task of focusing can only be achieved in frequency bins with relatively large singular values. When both scatterers are simultaneously present, the singular vectors are a linear combination of the transfer vector from each scatterer. The first singular vector can achieve focusing on the extended scatterer in frequency bins with relatively large singular values. The second singular vector can approximately focus on the pointlike scatterer in frequency bins that its scattering coefficients are relatively high and the first scattering coefficient of the extended scatterer are relatively low.



Note: Payment of separate fee required to attend

MONDAY AFTERNOON, 27 OCTOBER 2014 HILBERT CIRCLE THEATER, 7:00 P.M. TO 9:00 P.M.

## Session 1eID

## Interdisciplinary: Tutorial Lecture on Musical Acoustics: Science and Performance

Uwe J. Hansen, Chair

*Chemistry & Physics, Indiana State University, Terre Haute, IN 47803-2374*

***Invited Paper***

**7:00**

**1eID1. The physics of musical instruments with performance illustrations and a concert.** Uwe J. Hansen (Dept. of Chemistry and Phys., Indiana State Univ., Indiana State Univ., Terre Haute, IN 47809, uwe.hansen@indstate.edu) and Susan Kitterman (New World Youth Orchestras, Indianapolis, IN)

Musical Instruments generally rely on the following elements for tone production: a power supply, an oscillator, a resonator, an amplifier, and a pitch control mechanism. The physical basis of these elements will be discussed for each instrument family with performance illustrations by the orchestra. Wave shapes and spectra will be shown for representative instruments. A pamphlet illustrating important elements for each instrument group will be distributed to the audience. The Science presentation with orchestral performance illustrations will be followed by a concert of the New World Youth Symphony Orchestra. This orchestra is one of three performing groups of the New World Youth Orchestras, an organization founded by Susan Kitterman in 1982. Members of the Symphony are chosen from the greater Indianapolis and Central Indiana area by audition.