Session 2aAA

Architectural Acoustics and Physical Acoustics: Acoustic Parameters of Materials: Their Definition, Measurement, and Uses in Architectural Acoustics

Ronald Sauro, Chair
NWAA Labs, Inc., 90 Tower Blvd., Elma, WA 98541

Chair's Introduction—8:00

Invited Papers

8:05

2aAA1. A new way of determining the total absorption of gypsumboard wall structures. Ronald Sauro (NWAA Labs, Inc., 90 Tower Blvd., Elma, WA 98541, audio_ron@msn.com)

Gypsumboard wall construction is used throughout the United States and other parts of the world. Measuring the absorption of wall structures has been done in ways that always leave questions as to the efficacy of the measurement. Walls have been laid down on the floor and measured, and stood straight up and measured. Both methods leave questions as to the data because of differing results. There are questions about diaphragmatic movement that affects the absorption and its constrainment. We are proposing a different method that we think solves the problems inherent in each of the other methods, and we are presenting the results of measurements of indicated materials and comparisons with other methods

8:25

2aAA2. Acoustic diffusers, the effects of materials and finishes on diffusion efficiency and absorption coefficients. Richard L. Lenz (RealAcoustix LLC, 2637 N. Washington Blvd., #125, N. Ogden, UT 84414, RL@RealAcoustix.com)

Over the years, many different theories about the effects of material mass, finishes, porosity, and other physical attributes of diffuser design have been put forth. These theories have included excess absorption and/or reduced diffusion being caused by different woods or other materials being used as well as different types of paints and finishes. This paper is a study on the veracity of those theories and the effects of assorted materials in the construction of acoustic diffusers. The study will include the use of a specific diffuser design utilizing a periodic design with specific frequency ranges as well as more simplified geometric designs such as pyramids and barrels. Tests will be conducted to look at specific absorption results and the correlation to diffusion efficiency, if any. Standard testing procedures will be used to derive the information presented in the paper.

8:45

2aAA3. Determining sound transmission through damped partitions: Challenges in theoretical prediction and laboratory testing. Benjamin Shafer (Serious Energy, Inc., 1117 Tacoma Ave. South, Tacoma, Washington 98402, bshafer@craworld.com)

The sound transmission loss through traditional wall and ceiling building partitions can currently be predicted using software programs and/or laboratory test data. There is, however, a great divide that separates the theoretical prediction of sound transmission loss from the laboratory-measured values. Some of the most common prediction software packages do not account for dynamic (frequency- and temperature-dependent) material properties and are, therefore, incapable of predicting sound transmission accurately for some common solutions. The most accurate and precise software prediction tools available become impractical because the cross-correlation between laboratory testing facilities is so poor that it is not possible to replicate predicted performance dynamics in a laboratory setting. Using the structural properties of damping materials as an example, the divide between theoretical prediction and laboratory testing will be illustrated and possible solutions for closing such a divide will be presented.

9:05

2aAA4. An innovative acoustic muffler to reduce acoustic leakage from recessed lights, intake vents, exhaust vents, etc., while improving the acoustic environment within a room. Bonnie Schnitta (SoundSense, LLC, 46 Newtown Ln., Ste. One, East Hampton, NY 11937, bonnie@soundsense.com)

In 2011, Bonnie Schnitta of SoundSense received a patent for an Acoustic Muffler. This Muffler addresses the issue of significant sound transmission, or acoustic leakage, through openings in structures, such as a ceiling or wall, where airflow or heat dissipation is required. Examples of applications will be presented, such as openings created by recessed lights, speakers, or intake and exhaust vents. These openings in the structure significantly reduce the STC and/or IIC of the structure. The SoundSense Acoustic Muffler reduces acoustic leakage through an opening in a ceiling or wall that would potentially cause significant degradation in acoustic efficacy of the structure without resulting in any substantial pressure drop, while simultaneously allowing for the required air flow or heat dissipation.

The original application of the recessed light and additional similar applications as well as secondary purposes for this muffler will be detailed. One such secondary purpose is when the hole created by the recessed lights in the ceiling contributes to the room sounding better. The patented muffler not only allows this benefit to remain, but serves to inhibit the frequency(s) of concern from disturbing adjacent rooms. Cell tower equipment room noise reduction requiring airflow will also be detailed.

9:25

2aAA5. A comparison of predicted total absorption of different sized and shaped materials using traditional "absorption coefficient" vs a proposed absorption constant. Ronald Sauro (NWAA Labs, Inc., 90 Tower Blvd., Elma, WA 98541, audio_ron@msn.com)

After using traditional absorption coefficients and calculating total absorptions for a room and then underestimating room RTs, it was decided that a new constant was needed that could be used to determine actual absorption of a surface. A constant was developed that meets this need and this paper shows the differences in the predicted absorptions using both methods vs using the new constant and the advantages of using the new constant.

Contributed Papers

9:45

2aAA6. Laboratory measurement of the acoustic absorption coefficient based on the modal dispersion. Jevgenija Prisutova, Kirill Horoshenkov (Univ. of Bradford, Richmond Rd., Bradford BD7 1DP, United Kingdom, k.horoshenkov@bradford.ac.uk), Jean-Philippe Groby, and Bruno Brouard (Laboratoire d'Acoustique de l'Universite´ du Maine, Universite´ du Maine, Le Mans, France)

This work presents a novel method of measurement of the absorption coefficient of large material samples in an acoustic waveguide in a broad frequency range. The material sample is deployed at one end of an acoustic waveguide the other end of which is excited with a point source. The sound pressure data are obtained using a long horizontal microphone array deployed in this waveguide. The optimization analysis is then applied to the sound pressure data to calculate the modal reflection coefficients, which are then combined to determine the overall absorption coefficient of the material sample placed at the end of this waveguide. It believed that this method will be able to extend significantly the frequency range attained with the current ISO 10543-2 impedance tube method and be applied to those materials which have a corrugated surface or complex surface morphology such as acoustic diffusers or living plants. It is also believed that this method will provide the means to estimate efficiently the diffusivity of materials with complex surface morphology with a relatively simple laboratory setup.

10:00–10:15 Break

10:15

2aAA7. Uncertainty of normal-incidence absorption coefficient measurements using the two-microphone cross-spectral method. Matthew G. Blevins, Joshua Thede, and Lily M. Wang (Durham School of Architectural Eng. and Construction, Univ. of Nebraska - Lincoln, 1110 S. 67th St., Omaha, NE 68182, mgblevins@unomaha.edu)

Measured absorption coefficients have some uncertainty that can be expressed in terms of the uncertainty associated with each measured input quantity. The uncertainties of input quantities contribute to the combined uncertainty of the absorption coefficients in varying degrees dependent on the underlying relationship between each parameter. In this study, the propagation of uncertainty of the two-microphone cross-spectral method for measuring normal-incidence absorption coefficients is analyzed according to the ISO/IEC Guide 98-3:2008 "Guide to the expression of uncertainty in measurement." The results of an experimental investigation are explored to determine the chief sources of systematic error and the relationship between uncertainty of input quantities and uncertainty of intermediate calculations.

10:30

2aAA8. Optimization of sound absorption performance of a new ecological material. Seda Karabulut (MEZZO Studyo, Dept. of Architecture, Middle East Tech. Univ., ODTU Kosgeb Tekmer No112, Ankara 06800, Turkey, sedakarabulutt@gmail.com) and Mehmet Ĉalışkan (Dept. of Mech. Eng., Middle East Tech. Univ., Ankara, Turkey)

Material selection is foremost design parameter in providing acoustical comfort levels in both acoustically sensitive and noise generating spaces ranging from the small size meeting halls to large scale multipurpose auditoriums and even from foyers to shopping malls. Architects usually prefer seamless, unperforated materials in line with their interior design concepts. One of the objectives of this research is to contribute to the market portfolio of smooth faced, seamless acoustical materials with highest sound absorption performances. Another objective is to develop such a composition with ecological and sustainable ingredients, and binding techniques. Energy efficient and sustainable materials are frequently devised in construction industry for acoustically sensitive environments to get credits for international certification procedures such as LEED and BREEAM. Nevertheless, most of the acoustic materials in construction industry are perforated and/or supported with mineral wool based material backing, which have an adverse effect on indoor air quality. This article is on the improvement of an ecological unperforated sound absorptive material which is made of reed and pumice stone layers. The feasibility and effectiveness of a proposed configuration have already been studied. This paper seeks ways of optimizing number and thicknesses of different material layers in attaining maximum sound absorption performance.

10:45

2aAA9. Acoustic determination of impedance tube microphone locations. Cameron Fackler, Theodore S. Pitney, and Ning Xiang (Graduate Program in Architectural Acoust., Rensselaer Polytechnic Inst., 110 8th St., Greene Bldg., Troy, NY 12180, facklc@rpi.edu)

Impedance tube methods allow for convenient, rapid characterization of the normal-incidence acoustic properties of materials and constructions. Many such methods rely on the determination of precise phase relationships between microphones sensing the sound field inside the impedance tube; knowledge of the location of these microphones is crucial to the accuracy of the measurement. Due to the large physical size of typical microphone diaphragms, physical measurements of the microphone positions (such as with a ruler or caliper) are inadequate and have a large uncertainty. This paper presents a method to determine the impedance tube microphone acoustic center locations from a broadband acoustic reflection coefficient measurement of a rigid termination. Utilizing the Bayesian inference framework, the estimation procedure provides information about microphone locations and uncertainties in the position estimates.

11:00

2aAA10. Detection of specular and diffuse reflections in concert halls using continuous wavelet transforms. Jin Yong Jeon and Muhammad Imran (Dept. of Architectural Eng., Hanyang Univ., 17 Haengdang-dong, Seongdong-gu, Seoul, 133791, South Korea, jyjeon@hanyang.ac.kr)

Specular and diffuse reflections have been detected in fine spatio-temporal structures of room impulse responses (IRs) at different positions in a 1:10 scale model hall. Continuous wavelet transformation (CWT) coefficients were calculated using mother wavelet functions of the Daubechies wavelet families and specular reflections along with their degrees of diffuseness were investigated from the IRs. In CWT analysis, the early specular reflections were detected by fining their similarities with mother wavelet at different scales. While these reflections were treated as "defects" or "singularities" that embedded in the Schroeder decay curve. In addition, the time difference of arrival (TDOA) was applied to localize the reflections using a cross correlation function for time delay estimation and direction finding of binaural room IRs. The spectral characteristics in terms of delay time were calculated by auto-correlation and interaural cross-correlation functions (ACF/IACF).

11:15

2aAA11. Bayesian parameter estimation of single- and double-layer micro-perforated panel absorbers. Andrew Schmitt, Ning Xiang, and Cameron Fackler (Architectural Acoust., Rensselaer Polytechnic Inst., 110 8th St., Troy, NY 12180, aajschmitt@gmail.com)

Micro-perforated panel (MPP) absorbers have been widely accepted in applications as an efficient and clean sound-absorption solution providing extremely high, yet relatively narrow frequency band, acoustic absorption coefficients. Design for the custom acoustic performance of MPPs can be challenging for engineers and designers, however, due to the number of parameters making up the panel absorbers. This work examines the effects of the perforation rate, pore diameter, airspace depth, and panel thickness on the absorption profile of single- and double-layer MPP absorbers. A Bayesian inference framework is implemented in order to inversely determine parameter values and their uncertainties from experimentally measured acoustic absorption data of various MPP constructions of unknown parameters. This analysis also provides insight into the effects of each parameter on the acoustic performance as well as interdependence among the parameters.

11:30

2aAA12. Acoustics 101 for architects. Michael W. Fay (Contracting Div., Sound Image, 2415 Auto Park Way, Escondido, CA 92029, mfay@sound-image.com)

A presentation of acoustical terminology and concepts that relate directly to the design and construction of an architectural space, non-technical explanations, descriptions, and examples. Architectural acoustics can be defined as the study and application of acoustic principles as they are applied to the inside of a building or structure. Usually, these are buildings that will be used for a live performance or the presentation of other useful information. This paper is written with the architect in mind; to help define and explain some of the most basic concepts relating to architectural acoustics. In today's pro/commercial audio marketplace, architectural acoustics must play a significant role in the design process for any programmed meeting or entertainment space. This work puts essential terms and concepts into the hands of the architect, owner, or anyone else who would like to have a better understanding of this topic. Contents: What is Sound? Sound Propagation Three Acoustical Tools Where Does All the Unused Sound Go After it's Been Heard? Audio Volume Changes—What Do the Numbers Mean? The Inverse Square Law Room Geometry—The Good, Bad, and Ugly Reverberation and Echo Speech Intelligibility Noise Internal vs External Noise Room Modes Variable Acoustics Psychoacoustics Conclusion.

TUESDAY MORNING, 3 DECEMBER 2013 UNION SQUARE 23/24, 7:55 A.M. TO 11:50 A.M.

Session 2aAB

Animal Bioacoustics and Acoustical Oceanography: Broadening Applications of Tags to Study Animal Bioacoustics I

Marla M. Holt, Cochair
NOAA NMFS NWFSC, 2725 Montlake Blvd. East, Seattle, WA 98112

Alison K. Stimpert, Cochair
Dept. of Oceanogr., Naval Postgraduate School, Monterey, CA 93943

Chair's Introduction—7:55

Invited Papers

8:00

2aAB1. History of applications of tags to study animal bioacoustics. Peter L. Tyack (Biology, Univ. of St Andrews, Sea Mammal Res. Unit, Scottish Oceans Inst., St Andrews, Fife KY16 8LB, United Kingdom, plt@st-andrews.ac.uk)

I discuss the history of applications of tags to study animal bioacoustics, with an emphasis on toothed whales. My own interest in this topic was stimulated in the early 1980s by problems identifying which dolphin made a sound. Our inability to identify signaler and receiver hindered the study of communication among marine mammals. I discuss the evolution of devices and methods to solve this problem. The development of acoustic recording tags in the 1990s enabled the capacity to monitor what an animal hears at sea and how it responds. These tags form a critical enabler for field experiments on the relationship between acoustic exposure measured on the tag and behavioral responses, also often measured by the tag. Use of passive acoustic tags with echolocating animals has opened a new window on how toothed whales echolocate to find, approach, and capture prey, especially when the tags also include three-axis accelerometry and magnetometry to measure orientation and movement. The combination of these tags with passive acoustic monitoring provides a powerful method to improve localization, to estimate the three dimensional beampattern of sound production, and to estimate the absolute abundance of species that vocalize.

8:20

2aAB2. Acoustic time synchronization among tags on porpoises to observe their social relationships. Tomonari Akamatsu (Res. Ctr. for Fisheries System Eng., Fisheries Res. Agency & JST CREST, 7620-7, Hasaki, Kamisu, Ibaraki 314-0408, Japan, akamatsu@affrc.go.jp), Mai Sakai (Wildlife Res. Ctr., Univ. of Kyoto & JSPS, Kyoto, Japan), Ding Wang, Kexiong Wang, and Songhai Li (Key Lab. of Aquatic Biodiversity and Conservation of the Chinese Acad. of Sci., Inst. of Hydrobiology of the Chinese Acad. of Sci., Wuhan, China)

Observing and monitoring underwater social interactions of cetaceans is challenging. Because cetaceans spend most of their time underwater, it is important to monitor their underwater behavior individually. The finless porpoise is small and has no available natural identification marks that causes little knowledge of its sociality. Here we used acoustic datalogger to synchronize individual depth profile among individuals within a second. Acoustic and behavior tags were deployed on six free-ranging finless porpoises simultaneously and released in open water. Echolocation sounds were used as the trigger signal to synchronize the clock of all logging systems. Synchronous dives characterized by similar time-depth profile were used as an index of association. Two pairs tended to participate in long periods of synchronized diving more frequently than 13 other possible pairs, indicating that these four porpoises chose their social partners. Initiator and follower could be identified by precisely time synchronized data. The adult males tended to follow the immature female and juvenile male, respectively. However, the role of an initiator often changed within the pair during synchronized diving, and their body movements appeared to be non-agonistic. The time-synchronized bio-logging method was useful for observation of the social relationships of free-ranging aquatic animals.

8:40

2aAB3. Studying acoustic communication in pilot whale social groups. Frants H. Jensen (Biology, Woods Hole Oceanographic Inst., 266 Woods Hole Rd., M.S. # 50, Woods Hole, MA 02543, frants.jensen@gmail.com) and Peter L. Tyack (Scottish Oceans Inst., Univ. of St Andrews, St Andrews, United Kingdom)

Many cetaceans are gregarious animals with a complex group structure, and they depend on acoustic signals for mediating social interactions among individuals. However, the marine lifestyle and closed sound production system makes it difficult to study social signaling in groups of wild cetaceans. Acoustic and movement logging tags offer new possibilities for sampling the sounds and behavior of individuals, but themselves provide new challenges in determining the source of acoustic signals. Here, we draw on experiences from studies of short-finned and long-finned pilot whales to discuss how social signaling can be investigated in wild marine mammals. We discuss how specific social contexts, especially separations from the social group, can aid the interpretation of individual tag data to test whether calls of short-finned pilot whales are important in mediating social contact with group members, while emphasizing the pitfalls of using such methods. Specifically, we highlight the advantages of simultaneously instrumenting multiple closely associated pilot whales with acoustic and movement recording tags. This has improved our understanding of acoustic interactions through ready identification of the sender and simultaneously monitoring the reaction of other group members, and we use this dataset to discuss ongoing challenges of studying social dynamics using simultaneous tag deployments.

9:00

2aAB4. Insights into a complex communication system from tagged bottlenose dolphins. Laela Sayigh (Biology Dept., Woods Hole Oceanographic Inst., M.S. #50, Woods Hole, MA 02543, lsayigh@whoi.edu), Vincent Janik (Biology Dept., Univ. of St. Andrews, St. Andrews, United Kingdom), Frants Jensen (Biology Dept., Woods Hole Oceanographic Inst., Woods Hole, MA), Katherine McHugh, Randall Wells (Sarasota Dolphin Res. Program, Chicago Zoological Society, c/o Mote Marine Lab., Sarasota, FL), and Peter Tyack (Biology Dept., Univ. of St. Andrews, St. Andrews, United Kingdom)

Since 2011, we have deployed 30 acoustic and movement logging DTAGs on long-term, multi-generational resident bottlenose dolphins in Sarasota Bay, Florida, for a total of approximately 140 h. Twenty-two tags were deployed simultaneously on pairs of associated individuals, allowing for greater resolution of individual vocal activity. Virtually all dolphins in the Sarasota Bay community are identifiable both visually and by means of their individually distinctive signature whistles. Tags were attached during brief capture-release health assessments, and behavioral observations of tagged individuals post-release continued for as long as possible. Tag data reveal unique insights into foraging behavior, including distinctive acoustic and movement patterns associated with particular foraging modes (e.g., “pinwheel feeding”). In addition to echolocation clicks and buzzes, several distinctive pulsed sounds were recorded on the tags. Whistle copying was observed 18 times in a preliminary analysis of approximately two hours of data, and at least one instance involved more than two dolphins producing the same whistle. Finally, we obtained evidence for at least one shared, stereotyped non-signature whistle. Combining extensive longitudinal information on individual dolphins with fine scale behavioral and acoustic data provides tremendous opportunities for describing and quantifying the complexity of the bottlenose dolphin communication system.

9:20

2aAB5. Challenges in identifying (or not) focal animal sound production in baleen whale acoustic tag datasets. Alison K. Stimpert (Dept. of Oceanogr., Naval Postgrad. School, Monterey, CA 93943, alison.stimpert@gmail.com), Doug P. Nowacek (Nicholas School, Duke Univ., Beaufort, NC), Ari S. Friedlaender (Southall Environ. Associates, Inc., Aptos, CA), Jan Straley (Biology, Univ. of Alaska Southeast, Sitka, AK), David W. Johnston (Nicholas School, Duke Univ., Beaufort, NC), Jeremy A. Goldbogen (Cascadia Res. Collective, Olympia, WA), and Ching-Sang Chiu (Dept. of Oceanogr., Naval Postgrad. School, Monterey, CA)

Ascribing sounds on animal-borne tag recordings to individual sound producers is integral to understanding social behavior of animal groups. Previously, sounds recorded on tags have been assigned to the tagged individual (focal animal) based on proximity of other conspecifics, angle of arrival, low frequency artifacts in the sound, or a combination of signal-to-noise ratio (SNR) and received level (RL). However, most acoustic-based methods do not translate well to baleen whales producing low frequency sounds, as the tag often resides in the near field of the sound source. In addition, for social species that spend time in groups with conspecifics in close proximity,

sounds produced by nearby animals may have comparably high SNR and RL. Here we discuss the challenges of determining if a tagged whale is calling in baleen whale datasets, using acoustic records from two humpback whales, one fin whale, and one blue whale as examples. The datasets include intense song or feeding calls and are from several locations. We compare SNR, RL, harmonic content, and behavioral sensor data in these cases, and discuss the implications of confirming sound production by a tagged individual for measuring communication, behavior, and responses to external stimuli in baleen whales.

9:40

2aAB6. Tags, drifters, and Towfish: Using multiple recording platforms to characterize odontocete acoustic space. T. A. Mooney, Maxwell B. Kaplan (Biology Dept., Woods Hole Oceanographic Inst., 266 Woods Hole Rd., Woods Hole, MA 02543, amooney@whoi.edu), Robin W. Baird (Cascadia Res. Collective, Olympia, WA), and Jim Partan (Appl. Ocean Phys. and Eng. Dept., Woods Hole Oceanographic Inst., Woods Hole, MA)

Bioacoustic tags can reveal novel information about the behavior and ecology of animals on which they are deployed. Yet tags are often placed off the animals' acoustic axis, limiting some potential analyses. In order to broaden abilities to examine bioacoustic signals and behavior of several Hawaiian odontocetes we adapted recording methods to enhance data collection opportunities and free-field records. While bioacoustic DTAGs were deployed, we also used DMONs (digital acoustic recorders) in both a GPS-outfitted drifter buoy (Drifting Acoustic Wideband Gizmo = DAWG) and a Towfish around pantropical spotted dolphins (*Sa*), melon-headed whales (*Pe*), and short-finned pilot whales (*Gm*). Daytime tag recordings show *Pe* and *Sa* were limited to relatively shallow dives (< 50 m) but were relatively soniferous, whereas *Gm* made occasional deeper dives (to 700 m) and fewer individual calls. Group measures for *Pe* and *Sa* from the DAWG and Towfish revealed relatively high incidences of overlapping calls. Preliminary investigations of *Pe* whistles suggest some limited variation between *Pe* populations and considerable variability in individual call types. Such characterizations of call rates and variability support efforts to detect and classify odontocete calls. The different methods provided complementary means to collect substantial bioacoustic data on pelagic odontocetes for which few data exist.

10:00–10:15 Break

10:15

2aAB7. Classification of behavioral state using hidden Markov model analysis of animal-attached tag data: Applications and future prospects. Patrick J. Miller and Saana Isojunno (School of Biology, Univ. of St Andrews, Bute Bldg., St Andrews, Fife KY16 9QQ, United Kingdom, pm29@st-andrews.ac.uk)

Data from high-resolution animal-attached tags enable quantification of behavioral responses to anthropogenic noise. However, the duration of such detailed tag records on marine divers are typically too short to allow evaluation of the biological significance of such effects. To explore whether and how sperm whale behavior changed during exposure to sonar, we developed a discrete-time hidden activity state model that describes how observed parameters derived from measured Dtag data (depth, pitch, and clicking behavior) arise from five behavioral modes (surfacing, descent, bottom phase, ascent, resting, and silent active). Although the model assumed simple Markovian state-transitions, the state classification matched well with expert judgment of dive state. During experimental exposures to 1–2 kHz sonar, all four sperm whales tested reduced foraging time, increased silent active behavior, and buzz rates during foraging states decreased. None of those effects were found during 6–7 kHz experimental exposures of the same four whales, nor for three other whales exposed to distant sonar. Hidden classification of behavioral state using quantitative analysis of data collected by the animal attached tag is a procedure that has the potential to be processed autonomously on-board tags. This would enable collection and satellite telemetry of longer-term behavioral data sets with biologically significant interpretations.

10:35

2aAB8. Statistical analysis of data from acoustic tags: Methods for combining data streams and modeling animal behavior. Stacy L. DeRuiter, Catriona Harris, Dina Sadykova, and Len Thomas (School of Mathematics and Statistics, Ctr. for Res. into Ecological and Environ. Modelling, Univ. of St. Andrews, CREEM, St. Andrews KY169LZ, United Kingdom, sldr@st-andrews.ac.uk)

Statistical analysis of data from multi-sensor acoustic tags presents several characteristic challenges. Datasets generally include time-series of many measurements on a small number of individuals; different data streams often have distinct temporal resolutions and precisions. The MOCHA project (Multi-study Ocean acoustics Human effects Analysis) is a three-year effort focused on developing innovative statistical methods for such data. Here, we present several approaches for appropriate, effective statistical analysis of such datasets, with an emphasis on quantitative assessment of changes in marine mammal behavior in response to acoustic disturbance. Issues to be addressed will include: combining data streams from multi-sensor tags (and also concurrent visual observation data) for statistical analysis; statistical methods to characterize or summarize normal behavior and detect departures from normal; methods for analysis of call-production-rate data from acoustic tags; and methods for combining analysis of data from multiple tags, individuals, and species. Specific statistical methods to be presented will include Mahalanobis distance as a summary of multivariate data, state-switching models, random effects, and other extensions of generalized linear models appropriate to tag data.

10:55

2aAB9. Using acoustic tags to investigate sound exposure and effects on behavior in endangered killer whales (*Orcinus orca*). Marla M. Holt, M. Bradley Hanson, Candice K. Emmons (Conservation Biology Div., NOAA NMFS NWFSC, 2725 Montlake Blvd. East, Seattle, WA 98112, Marla.Holt@noaa.gov), Juliana Houghton (School of Aquatic and Fishery Sci., Univ. of Washington, Seattle, WA), Deborah Giles (Dept. of Wildlife, Fish and Conservation Biology, Univ. of California, Davis, Seattle, CA), Robin W. Baird, and Jeff Hogan (Cascadia Res. Collective, Olympia, WA)

In this investigation, acoustic tags (DTAGs) allow us to better understand noise exposure and potential behavioral effects in endangered Southern Resident killer whales (SRKWs). Designated critical habit of SRKWs includes summer foraging areas where vessel traffic from commercial shipping, whale-watching, and other boating activities is common. Risk factors of population recovery include

vessel and noise effects, and prey quality and availability. DTAGs, equipped with hydrophones and other sensors, are attached to individual whales to collect data on vocal and movement behavior, as well as their acoustic environment. Specific research goals include: (1) quantifying received noise levels in biologically relevant frequency ranges from tag data; (2) determining relationships between noise levels and detailed vessel variables obtained from precise geo-referenced vessel data collected concurrently; (3) investigating whale acoustic and movement behavior during different activities, including foraging, to understand sound use and behavior in specific biological and environmental contexts; and (4) determining potential effects of vessels and associated noise on behavior. This paper will describe the experimental approach taken, challenges faced, and results obtained from over 80 h of tag data. These data have been critical for addressing our research goals related to multiple population risk factors of endangered SRKWs.

11:15

2aAB10. Killers in the dark—Acoustic evidence for night-time predation by mammal-eating killer whales (*Orcinus orca*) in Alaska. Volker B. Deecke (Ctr. for Wildlife Conservation, Univ. of Cumbria, Penrith CA11 0AH, United Kingdom, volker.deecke@cumbria.ac.uk), Ari D. Shapiro (Biology Dept., Woods Hole Oceanographic Inst., Boston, MA), and Patrick J. Miller (Sea Mammal Res. Unit, Univ. of St. Andrews, St. Andrews, United Kingdom)

North Pacific killer whales that specialize on hunting marine mammals do not typically echolocate while searching for prey. This suggests that they detect prey by either relying on visual cues or listening for sounds generated by prey animals. If prey detection requires vision, hunting should be limited to the daylight hours. Documenting predation at night would therefore provide evidence supporting a passive listening hypothesis of prey detection. We used digital recording tags (DTAGs) to study the behavior of mammal-eating killer whales in Southeast Alaska. These tags recorded the underwater movements of the tagged individual and any sound emitted or received. Predation events were identified using distinctive sounds generated during prey capture and handling. We deployed 13 tags, of which 7 remained attached for at least part of the night. The majority of tags recorded night-time predation, even though nights were short (average of 4:18 h) during the study period. These findings show that mammal-eating killer whales can detect prey at night and thus suggest that passive listening is an important part of their hunting strategy. Acoustic data from digital recording tags can therefore provide valuable insights into the night-time activities and foraging behavior of killer whales and other marine mammals.

Contributed Paper

11:35

2aAB11. Acoustic and foraging behavior of tagged sperm whales under natural and depredation foraging conditions in the Gulf of Alaska. Delphine Mathias (GIPSA-Lab, 11 rue des Mathématiques, Saint Martin d'Hères 38402, France, delphine.mathias@gmail.com), Lauren Wild (Sitka Sound Sci. Ctr., Sitka, AK), Aaron Thode (Scripps Inst. of Oceanogr., La Jolla, CA), Jan Straley (Univ. of Alaska Southeast, Sitka, AK), John Calambokidis, and Greg S. Schorr (Cascadia Res. Collective, Olympia, WA)

Sperm whales have been depredating black cod (*Anoplopoma fimbria*) from demersal longlines in the Gulf of Alaska for decades, but the behavior has now become pervasive enough that it may be affecting government estimates of the sustainable catch, motivating further studies of this behavior. Over a three-year period, 11 B-Probe bioacoustic tags have been attached to seven adult sperm whales off Southeast Alaska, permitting observations of the animals' dive profiles and acoustic behavior during natural and depredation foraging conditions. Two rough categories of depredation were identified: "deep" and "shallow." "Deep depredating" whales consistently surface within 500 m of a hauling fishing vessel, have maximum dive depths greater than 200m, and display significantly different acoustic behavior than naturally foraging whales, with shorter inter-click intervals, occasional bouts of high "creak" rates, and fewer dives without creaks. "Shallow depredating" whales conduct dives that are much shorter, shallower, and more acoustically active than both the natural and deep depredating behaviors, with median creak rates three times that of natural levels. Occurrence of slow clicks and the behavioral context in which these vocalizations are produced were also investigated. These results provide insight into the energetic benefits of depredation behavior to sperm whales. [Work conducted under the SEAS-WAP program, supported by the North Pacific Research Board and the National Geographic Society.]

Session 2aBA

Biomedical Acoustics and Signal Processing in Acoustics: Application of Acoustic Radiation Force in Medical Imaging

Mostafa Fatemi, Chair
Physiol. & Biomedical Eng., Mayo Clinic College of Medicine, 200 First St. SW, Rochester, MN 55905

Chair's Introduction—7:55

Invited Papers

8:00

2aBA1. Acoustic radiation force on gas bubbles and soft elastic scatterers in tissue. Sangpil Yoon, Salavat R. Aglyamov, Andrei B. Karpiouk (Dept. of Biomedical Eng., The Univ. of Texas at Austin, Austin, TX), Stanislav Y. Emelianov (Dept. of Biomedical Eng., The Univ. of Texas at Austin, Austin, Tennessee), Yurii A. Ilinskii, Evgenia A. Zabolotskaya, and Mark F. Hamilton (Appl. Res. Labs., The Univ. of Texas at Austin, 204 E. Dean Keeton St., Stop C2200, Austin, TX 78712-1591, hamilton@mail.utexas.edu)

Acoustic radiation force on a scatterer in tissue depends on the compressibility and shear modulus of both the tissue and the scatterer. This force is related to the monopole and dipole scattering coefficients. The finite shear modulus of the tissue decreases the radiation force in comparison with the force exerted on the same scatterer surrounded by liquid. Shear moduli for soft tissue range from several kilopascals (breast, liver) to tens of kilopascals and higher for cornea, cartilage, and cancerous tissue. As reported previously, the radiation force on a bubble in tissue having 100 kPa shear modulus is 50% less than if the bubble is in water. This difference decreases for scatterers with finite shear moduli, examples of which are reported here. Additionally, displacement of a scatterer due to radiation force is inversely proportional to the shear modulus of the tissue, which permits measurement of the latter. Experiments demonstrating this technique are reviewed. In these experiments, the radiation force is applied to a gas microbubble produced by laser-induced optical breakdown, while displacement of the microbubble is measured by high-frequency ultrasound as a function of time. Results are reported for tissue-mimicking phantoms and animal crystalline lenses *in vitro*.

8:20

2aBA2. A review of the medical applications of shear wave elastography. Mickael Tanter, Mathieu Pernot, Jean Luc Gennisson, and Mathias Fink (Langevin Inst., ESPCI, 1 rue jussieu, Paris 75005, France, mickael.tanter@espci.fr)

Supersonic shear wave elastography (SWE) is a quantitative stiffness imaging technique based on the combination of a radiation force induced in tissue by an ultrasonic beam and ultrafast ultrasound imaging sequence (up to more than 10,000 frames per second) catching in real time the propagation of the resulting shear waves. Local shear wave speed is estimated and enables the two dimensional mapping of shear elasticity. This imaging modality is implemented on conventional probes driven by dedicated ultrafast echographic devices and can be performed during a standard ultrasound exam. The clinical potential of SSI is today extensively investigated for many potential applications such as breast cancer diagnosis, liver fibrosis staging, cardiovascular applications, and ophthalmology. This invited lecture will present an overview of the current investigated applications of SSI and the new trends of shear wave elastography research topics.

8:40

2aBA3. Application of shear wave imaging and shear wave dispersion ultrasound vibrometry in assessing viscoelastic properties of human thyroid: *In vivo* pilot study. Mohammad Mehrmohammadi, Pengfei Song, Carolina A. Carrascal, Matthew W. Urban (Physiol. and Biomedical Eng., Mayo Clinic, 200 First St. SW, Rochester, MN 55905, mehrmohammadi.mohammad@mayo.edu), Matthew R. Callstrom (Radiology-Diagnostic, Mayo Clinic, Rochester, MN), John C. Morris (Endocrinology, Mayo Clinic, Rochester, MN), Shigao Chen, James F. Greenleaf, Mostafa Fatemi, and Azra Alizad (Physiol. and Biomedical Eng., Mayo Clinic, Rochester, MN)

Thyroid cancer is the fastest growing age and gender adjusted cancer in 2011 according the American Cancer Society. The majority of the clinically diagnosed thyroid nodules are benign while less than 5% represent intrathyroidal cancers. Currently, the clinical gold-standard procedure for assessing the thyroid nodules is needle biopsy, a procedure that is associated with significant financial burden as well as pain and risk for patients. Therefore, a noninvasive, affordable, and potentially widely available method to differentiate between benign and malignant thyroid nodules can play an important role in reducing the number of unnecessary biopsies. In this study, we investigate the feasibility of two acoustic radiation force elastography techniques, shear wave dispersion ultrasound vibrometry (SDUV) and comb-push ultrasound shear wave elastography (CUSE imaging), in identifying thyroid nodules (imaging) and differentiating between benign and malignant pathologies based on their elasticity and viscosity (SDUV measurements). Our preliminary results show that the measured shear elasticity and shear viscosity parameters depend on tissue type; hence, these measurements may be utilized to differentiate between healthy normal thyroid tissue, benign nodules, and malignant nodules. Further studies on a large population of patients is required to better evaluate the role of the combination of elasticity and viscosity properties of tissue in differentiating various thyroid nodules.

9:00

2aBA4. Ultrasound-based shear wave evaluation in transverse isotropic tissue mimicking phantoms. Sara Aristizabal, Randall R. Kinnick, Carolina Amador, Ivan Z. Nenadic, James F. Greenleaf, and Matthew W. Urban (Physiol. and Biomedical Eng., Mayo Clinic College of Medicine, 1st St. SW, Rochester, MN 55905, aristizabaltaborda.sara@mayo.edu)

Introduction: Ultrasound radiation force-based methods can quantitatively evaluate tissue viscoelastic material properties. A limitation of current methods includes neglecting the inherent anisotropy nature of tissue. To explore this phenomenon, we created a phantom incorporating fibrous material that has preferential orientations. Methods: Two phantoms were made in a cube-shaped mold using a fibrous material arranged in multiple layers and embedded in porcine gelatin using two different concentrations of the gelatin (8%, 14%). Shear wave measurements were made in the phantoms at different angles by rotating the phantom, where 0° and 180° were defined along the fibers, and 90° and 270° across the fibers. Measurements were performed using a Verasonics ultrasound system equipped with a linear array transducer. Results/Discussion: The mean shear wave speeds and mean standard deviations for 8% and 14% gelatin along the fibers (0°) were (3.60 ± 0.03 and 4.10 ± 0.11 m/s) and across the fibers (90°) were (3.18 ± 0.12 and 3.90 ± 0.02 m/s), respectively. Conclusion: The fibrous gelatin-based phantoms exhibited anisotropy that could be measured using quantitative shear waves speed measurements. Increasing the gelatin percentage increases the shear wave speed and anisotropic moduli. [This study was supported by NIH grant DK092255.]

9:15

2aBA5. Characterizing dynamics of shear waves induced with acoustic radiation force impulse in histotripsy lesions for treatment feedback. Tzu-Yin Wang (Radiology, Stanford Univ., 1201 Welch Rd., Stanford, CA 94305, tzuyin@stanford.edu), Timothy L. Hall, Zhen Xu (Biomedical Eng., Univ. of Michigan, Ann Arbor, MI), J. Brian Fowlkes (Radiology, Univ. of Michigan, Ann Arbor, MI), and Charles A. Cain (Biomedical Eng., Univ. of Michigan, Ann Arbor, MI)

Histotripsy mechanically fractionates soft tissues into fluid-like homogenates that cannot support shear waves. We hypothesize that dynamics of shear waves excited from a histotripsy lesion using acoustic radiation force impulse (ARFI) change progressively during the fractionation process, and such change is related to the degree of tissue fractionation. To test this hypothesis, lesions with different degrees of fractionation were created in agar-graphite tissue mimicking phantoms and *ex vivo* kidneys with increasing numbers of histotripsy pulses (3-cycle 750- kHz ultrasound pulses at a peak negative/positive pressure of 17/108 MPa). The shear waves were excited by ARFI focused at the lesion center. The shear-induced temporal displacement profile was measured at a lateral location 10 mm offset to the lesion with M-mode imaging. Results showed significant changes in two characteristics: the peak-to-peak displacement decayed exponentially, and the relative time-to-peak displacement increased and saturated with increasing numbers of histotripsy pulses ($N=6$). Correspondingly, the degree of tissues fractionation, as indicated by the percentage of structurally intact cell nuclei, decreased exponentially. Strong linear correlations existed between the two characteristics and the degree of tissue fractionation ($R2 > 0.97$). These results suggest that characteristics of shear waves induced in a histotripsy lesion may provide useful feedback for treatment outcomes.

9:30

2aBA6. Robust shear wave motion tracking using ultrasound harmonic imaging. Pengfei Song, Heng Zhao, Matthew W. Urban, Armando Manduca (Physiol. and Biomedical Eng., Mayo Clinic College of Medicine, Med. Sci. 1-24, 200 First St. SW, Rochester, MN 55905, song.pengfei@mayo.edu), Sorin V. Pislaru (Cardiovascular Diseases, Mayo Clinic College of Medicine, Rochester, MN), Randall R. Kinnick, James F. Greenleaf, and Shigao Chen (Physiol. and Biomedical Eng., Mayo Clinic College of Medicine, Rochester, MN)

Robust shear-wave motion detection is essential for producing reliable shear elasticity measurements for ultrasound shear-wave elastography (SWE). However, because shear-wave motion is extracted from radiofrequency signals which can be heavily contaminated by noise, shear-wave motion detection can be very challenging, especially for *in vivo* applications. This study investigated the implementation of harmonic imaging (HI) to facilitate more robust shear-wave tracking based on HI's effectiveness in suppressing imaging noise associated with ultrasound reverberation, phase aberration, and clutter noise. A HI shear-wave tracking sequence was developed combining the pulse-inversion HI method with the plane wave imaging technique to transmit phase-inverted pulses at a high frame rate of several kilohertz. The backscattered ultrasound signals from phase-inverted pulses were added to suppress the fundamental and enhance the second harmonic component, from which the shear-wave motion was extracted. A pork belly phantom experiment showed that HI could significantly improve shear-wave motion detection by producing almost three-fold less underestimation of shear-wave motion and over ten-fold more precision for shear-wave speed measurements than fundamental imaging. An *in vivo* transthoracic case study of a human heart showed that HI substantially improved the success rate of shear-wave motion detection and could provide consistent estimates of the left ventricle myocardium stiffness.

9:45

2aBA7. Multipoint measurement of sound pressure and temperature in biological tissues by using optical fiber sensors. Takashi Kageyama (Faculty of Life and Medical Sci., Doshisha Univ., 1-3 Tataramiyakodani, Kyotanabe, Kyoto 610-0321, Japan, dmm1013@mail4.doshisha.ac.jp), Daisuke Koyama (Faculty of Sci. and Eng., Doshisha Univ., Kyotanabe, Kyoto, Japan), and Iwaki Akiyama (Faculty of Life and Medical Sci., Doshisha Univ., Kyotanabe, Japan)

Acoustic radiation force impulse (ARFI) has been actively studied in the field of medical ultrasonics. ARFI generates shear wave in the tissue to evaluate the tissue hardness using that velocity, and this method is applied to the diagnosis of liver tumor. Generally, long pulses are used for ARFI; therefore, we need to take account of thermal and physical effects on biological body. According to the regulation of Food and Drug Administration (FDA), the acoustic output of diagnostic ultrasound is approved as follows: $I_{spta.3} < 720$ mW/cm^2, MI < 1.9. However, there are some reports that we have possibilities to affect body thermally and physically even under these conditions. In this report, we propose the optical fiber system using fiber Bragg grating (FBG) to measure ultrasound pressure and temperature change separately. The purpose is that we measure the temperature range of 4 degrees at the resolution of 0.1 degree and the sound pressure range of 10 MPa. We utilized an optical fiber which had 10 mm FBG in the center; moreover a narrowband light source and a photo detector to modulate reflected light signals by FBG to electrical signals. We obtained 3.2 μV/Pa and 9.5 pm/degree as the sensitivities of 10 mm FBG.

10:00

2aBA8. Angle exchange symmetries for acoustic Bessel beam scattering and the radiation force on spheres. Philip L. Marston (Phys. and Astronomy Dept., Washington State Univ., Pullman, WA 99164-2814, marston@wsu.edu)

Bessel beams are characterized by a cone angle β and as previously noted there is an exchange symmetry in the scattering involving the cone and the scattering angle θ [Marston, J. Acoust. Soc. Am. **121**, 753–758 (2007)]. The sphere is taken to be on the beam's axis. In the present study a broader class of exchange symmetries in the scattering pattern in the (θ, β) domain are noted for vortex as well as zero-order beams. Reflection exchange symmetries are present in the scattering pattern about the lines: $\theta = \beta$ and $(180° - \theta) = \beta$. The radiation force is known to be affected by the angular asymmetry of the scattering pattern as a function of scattering angle [Zhang and Marston, Phys. Rev. E **84**, 035601 (2011)]. It follows from angle exchange that the radiation force is similarly related to cone-angle asymmetry of the scattering pattern for a fixed scattering angle. This applies to negative as well as positive forces and is easily illustrated. [Work supported by ONR.]

10:15–10:30 Break

10:30

2aBA9. Evaluation of frequency characteristics of shear waves produced by unfocused and focused beams. Matthew W. Urban, Carolina Amador, Heng Zhao (Dept. of Physiol. and Biomedical Eng., Mayo Clinic College of Medicine, 200 First St. SW, Rochester, MN 55905, urban.matthew@mayo.edu), Mingu Lee, Hwan Shim (Samsung Electronics Co., Suwon-Si, South Korea), Shigao Chen, and James F. Greenleaf (Dept. of Physiol. and Biomedical Eng., Mayo Clinic College of Medicine, Rochester, MN)

Measurements of viscoelasticity with shear wave velocity dispersion requires measurements over a large bandwidth. In this study, we explored the parameters that modulate the frequency characteristics of shear waves induced using radiation force push beams. We used a Verasonics ultrasound scanner equipped with a linear array transducer. We performed measurements of shear wave motion induced using both focused and unfocused ultrasound beams. Measurements were made in elastic phantoms with shear moduli of 1, 4, and 16 kPa. The number of elements used for the unfocused beams were varied from 8 to 24, and for the focused beams from 16 to 128. The shear wave motion was tracked using plane wave imaging, and a one--dimensional autocorrelation algorithm applied to the acquired in-phase/quadrature data. At each pixel we calculated the fast Fourier transform of the data and found the center frequency, center-of-gravity, and -3 dB bandwidth. We compared the frequency characteristics from the different push beams. The frequency characteristics were found to be spatially variant and dependent on the number of elements used as well as the shear modulus of the medium. The center frequency, center-of-gravity, and the bandwidth were found to be correlated to one another, and strongly associated with the stiffness of the medium.

10:45

2aBA10. Model-free quantification of shear wave velocity and attenuation in tissues and its *in vivo* application. Ivan Nenadic, Matthew W. Urban, Bo Qiang, Shigao Chen, and James Greenleaf (Mayo Clinic, 200 1st St. SW, Rochester, MN 55906, ivandulan@gmail.com)

We validate a technique for model-free measurement of shear wave velocity and attenuation. A mechanical shaker was used to excite harmonic plane and cylindrical waves in phantoms and excised tissue. Radiation force was used to excite impulsive cylindrical waves. 2D FFT of the displacement yielded the k-space whose coordinates are frequency and the wave number. The shear wave velocity at each frequency was obtained by finding the maximum at the given frequency in k-space and dividing the frequency coordinate by the wave number coordinate. The attenuation (α) at a given frequency was calculated using $\alpha = \text{FWHM} \times \pi/\sqrt{3}$, where FWHM is the full width at half maximum of the k-space peak along the given frequency. This method was applied to measure shear wave velocity and attenuation of transplanted kidneys and livers, and in the thyroid tumor, and compare it to the healthy tissues. The velocities and attenuations at each frequency for various excitation methods agree within one standard deviation. The k-space estimates of velocity and attenuation agreed with those obtained using the phase gradient (velocity) and amplitude decay (attenuation). The transplanted organs and the thyroid tumor had higher velocity and lower attenuation than healthy tissues.

11:00

2aBA11. Using equilibrium equation constraints to obtain precise lateral displacement estimates in ultrasound imaging. Olalekan A. Babaniyi, Paul E. Barbone (Mech. Eng., Boston Univ., 110 Cummington St., Boston, MA 02215, lekanb@bu.edu), and Assad A. Oberai (Mech., Aerosp., and Nuclear Eng., Rensselaer Polytechnic Inst., Troy, NY)

Ultrasound elasticity imaging, whether based on radiation force, quasistatic deformation, or other means, depends upon using ultrasound to measure tissue motion. Ultrasound image data with standard beamforming can provide very precise measurements of soft tissue displacement in the axial direction, i.e., in the direction of the ultrasound beam. Lateral (and elevational) displacement estimates are relatively noisy. The authors describe a new processing method designed to estimate a precise and accurate 2D full displacement vector field from accurate measurements of a single component, and a noisy measurement of a second component. The proposed variational approach finds the displacement field that best fits the data, but that satisfies Navier's equation locally. The equilibrium equation is automatically relaxed along internal interfaces in the material, which need not be identified beforehand. In this way, the method accommodates piecewise constant material property distributions without knowing in advance where the properties change. The iterative implementation for plane stress incompressible elasticity converges in 3–10 iterations. [Authors gratefully acknowledge funding from NSF and NIH (NSF Grant No. 50201109; NIH NCI-R01CA140271).]

11:15

2aBA12. Approximate analytical models for cylindrical shear wave propagation in viscoelastic media. Yiqun Yang (Elec. and Comput. Eng., Michigan State Univ., 2120 Eng. Bldg., East Lansing, MI 48864, mcgough@egr.msu.edu), Matthew W. Urban, Bob Qiang (Biomedical Eng., Mayo Clinic, Rochester, MN), and Robert J. McGough (Elec. and Comput. Eng., Michigan State Univ., East Lansing, MI)

Improved approximate frequency domain expressions are derived for cylindrical shear wave propagation in viscoelastic media. These expressions extend prior results that describe cylindrical wave propagation in lossless media. Previously, an analytical expression for a cylindrical wave was obtained in terms of a Hankel function and a large argument approximation was applied to the result. A leading frequency-dependent term was then treated as a constant with respect to frequency. In the improved expression, the frequency-dependence of the leading term is retained. For lossless media, the leading term is a fractional integrator, and for viscoelastic media, the leading term is either a fractional integrator or an integer-order integrator, depending on the frequency range. The lossless and the viscoelastic models are evaluated in the frequency domain for simplified source geometries and compared to numerical results. The comparison shows that the agreement between the analytical and the numerical models is excellent. Implications for time-domain calculations in viscous media will also be discussed. [This work was supported in part by NIH Grant Nos. R01 EB012079 and R01 DK092255.]

11:30

2aBA13. Ultrasound bladder vibrometry for evaluation of bladder compliance: Preliminary *in vivo* results. Mohammad Mehrmohammadi, Ivan Z. Nenadic, Matthew W. Urban, James F. Greenleaf, Azra Alizad (Physiol. and Biomedical Eng., Mayo Clinic, 200 First St. SW, Rochester, MN 55905, mehrmohammadi.mohammad@mayo.edu), Douglas A. Husmann, Lance A. Mynderse (Urology, Mayo Clinic, Rochester, MN), and Mostafa Fatemi (Physiol. and Biomedical Eng., Mayo Clinic, Rochester, MN)

Bladder compliance is defined as bladder's ability to expand in volume without a significant change in pressure. The bladder wall is a multi-layered structure including a muscular layer and fibrous connective tissue. As the percent of connective tissue in the bladder interstitium increases compared to smooth muscle, the bladder becomes more rigid and less capable of expanding during filling. Currently, urodynamic studies (UDS) are considered as the clinical gold standard for bladder compliance assessment. This procedure is invasive and is associated with patient discomfort and is expensive. Ultrasound bladder vibrometry (UBV) is a novel acoustic-radiation-forced-based method for noninvasive assessment of bladder compliance. In UBV, an impulsive acoustic radiation force is focused on the bladder wall (under B-mode ultrasound image guidance) to induce vibrations which may be modeled by as a Lamb wave. High frame rate ultrasound is then utilized to detect the induced waves. This wave motion is then used to estimate the viscoelastic properties and the compliance of the bladder wall. Our results reveal a remarkable agreement between UBV and UDS cystometry measures, suggesting the potential of UBV as a viable clinical tool for the assessment of bladder compliance.

11:45

2aBA14. Correlation of ultrasound speckle pattern and arrival time errors in shear wave elastography. Stephen A. McAleavey (Biomedical Eng., Univ. of Rochester, 309 Goergen BME/Optics Bldg., Rochester, NY 14627, stephen.mcaleavey@rochester.edu)

Speckle results from coherent illumination and phase-sensitive detection of echoes from many sub-resolvable scatterers. Speckle provides a texture that allows ultrasonic tracking of tissue motion, which in turn enables the many varieties of ultrasound elastography. A complication is that the presence of speckle can induce a shift in the apparent phase or arrival time of a propagating shear wave. The correlation between shear wave arrival time and the speckle pattern measured with a stationary transmit beam and swept receive beams is demonstrated. Simulated transient shear waves generated in response to acoustic radiation force are tracked with transmit f-numbers of 2 to 4 and receive f-numbers of 2 to 8. The shear wave arrival time error and the lateral first moment of the swept-receive speckle pattern scaled by the shear wave speed are shown to be strongly, though not perfectly, correlated (r ~0.6). These arrival time errors are comparatively insensitive to the amplitude (over a range of -20 to 20 μm) and direction of propagation of the shear wave; e.g., arrival times for -20 and $+20$ μm shear waves differed by $<10\%$ of the total arrival time error. Approaches for the suppression of speckle noise in shear wave elastography are discussed.

12:00

2aBA15. Concentration of blood components by acoustic radiation force. Daniel Kennedy (Pharmacology, Western New England Univ., Springfield, MA), Brianna Sporbert, Tyler Gerhardson (Biomedical Eng., Western New England Univ., Springfield, MA), Dane Mealy (FloDesign Sonics, Springfield, MA), Michael Rust (Biomedical Eng., Western New England Univ., Springfield, MA), and Bart Lipkens (Mech. Eng., Western New England Univ., 1215 Wilbraham Rd., Box S-5024, Springfield, MA 01119, blipkens@wne.edu)

We have recently reported on macro-scale separation of lipids from red blood cells (J. Acoust. Soc. Am. **133**(5, Pt. 2), 3279). In this system, the acoustic radiation force is used to trap red blood cells and lipid particles in a standing wave. Subsequent clumping of red blood cells results in gravitational settling of the red blood cells. Similarly, coalescing of lipids results in rising of the lipids out of solution. We now report on more detailed measurements of the concentration and separation of various blood components, such as red blood cells, white blood cells, and platelets. Porcine blood was used in the experiments and diluted by a factor of ten in phosphate buffered saline. A VetScan HM5 hematology analyzer was used to do the blood count. Inlet flow rate through the device was 16 ml/min, the concentrate flow rate of the blood components was typically about 1 ml/min. The transducer was a 2 MHz PZT-8 operating at 10 W. No lysing was observed in any of the experiments. Results indicate successful capture of red blood cells and white blood cells with separation efficiencies in excess of 90% in a single pass. [Work supported by NSF PFI:BIC 1237723.]

TUESDAY MORNING, 3 DECEMBER 2013 MASON, 8:30 A.M. TO 10:35 A.M.

Session 2aEA

Engineering Acoustics and Structural Acoustics and Vibration: Composite Plates

Andrew J. Hull, Chair
Naval Undersea Warfare Ctr., 1176 Howell St., Newport, RI 02841

Invited Papers

8:30

2aEA1. Far-field approximation for a point-excited anisotropic plate. Elizabeth A. Magliula (NAVSEA Newport, 1176 Howell St., Bldg. 1302, Newport, RI 02841, elizabeth.magliula@navy.mil), J. Gregory McDaniel, and Allan D. Pierce (Dept. of Mech. Eng., Boston Univ., Boston, MA)

An analytic approximation is derived for the far-field response of a generally anisotropic plate subject to a time-harmonic point force acting normal to the plate. This approximation quantifies the directivity of the flexural wave field that propagates away from the force, which is expected to be useful in the design and testing of anisotropic plates. Derivation of the approximation begins with a two-dimensional Fourier transform of the flexural equation of motion. Inversion to the spatial domain is accomplished by contour integration over the radial component of wave number followed by an application of the method of stationary phase to integration over the circumferential component of wave number. The resulting approximation resembles that of an isotropic plate but involves wave numbers, wave amplitudes, and phases that depend on propagation angle. Numerical results for a plate comprised of bonded layers of a graphite-epoxy material illustrate the accuracy of the method compared to a numerical simulation based on discrete Fourier analysis. Three configurations are analyzed in which the relative angles of the layers are varied. In all cases, the agreement is quite good when the distance between force and observation point is greater than a few wavelengths.

8:50

2aEA2. Optimal design of composite plates based on wave propagation characteristics. J. Gregory McDaniel (Mech. Eng. Dept., Boston Univ., 110 Cummington Mall, Boston, MA, jgm@bu.edu) and Elizabeth A. Magliula (NAVSEA Newport, Newport, RI)

The present work is concerned with the design of composite plates that are optimized with respect to their wave propagation characteristics. For example, one may wish to design a composite plate such that the attenuation of a dominant wave is maximized. The optimization proposed here considers a composite plate with a specified number of layers. The material properties and thicknesses of the layers are considered as optimization parameters and these parameters are typically constrained in some way. For each choice of the

optimization parameters, complex wave numbers and their associated wave shapes are calculated by using the semi-analytical finite element method developed by others. This method uses a finite element discretization in the thickness coordinate and a propagating wave solution in the lateral coordinates, resulting in a quadratic eigenvalue problem for the complex-valued wave number of each wave that the plate supports. One then computes a scalar cost function based on the complex wave numbers and their associated wave shapes, and proceeds by adjusting the optimization parameters to minimize the cost function. This presentation will describe the computational aspects of the optimization approach and will illustrate its potential by example. [Work supported by ONR under grant N000141210428.]

9:10

2aEA3. Establishing the admissible waves and the steady state response of a rib-stiffened, layered plate structure subjected to high frequency acoustic loading. Kirubel Teferra and Jeffrey Cipolla (Appl. Sci. Div., Weidlinger Assoc., 375 Hudson St., New York, NY 10014, kirubel.teferra@wai.com)

An existing pseudo-analytical, frequency domain solution for wave propagation in coated, ribbed, three-dimensional elastic layered plates excited by acoustic plane waves provides fast solutions for high frequency excitations. The solution methodology, which is found to be numerically unstable under certain conditions, contains a fundamental ansatz regarding the set of excited wave forms expressed through a particular wave number expansion in the direction of periodicity. We propose to identify the set of admissible propagating (and attenuating) waves via an eigenvalue analysis as a preprocessing step when executing the pseudo-analytical solution. A significant challenge lies in determining the admissible waves of structures with periodicity in two-dimensions. The Wave Guide Finite Element (WFE) method leads to a two parameter, nonlinear eigenvalue problem, which is extremely difficult to solve. By formulating the problem with the Scale Independent Element (SIE) formulation, the admissible waves can be expressed through a two-parameter quadratic eigenvalue problem. This formulation overcomes the numerical conditioning issues associated with WFE method. This study compares the aforementioned ansatz with that computed by the SIE formulation as well as the improvements in the numerical stability associated with this additional step. The forced response results are compared to a finite element analysis.

9:30

2aEA4. Acoustic scattering from finite bilaminar composite cylindrical shells-three-dimensional solution. Sabih I. Hayek (Eng Sci., Penn State, 953 McCormick Ave., State College, PA 16801-6530, sihesm@engr.psu.edu) and Jeffrey E. Boisvert (NAVSEA Div Newport, Newport, RI)

The acoustic scattering from a finite bilaminar cylindrical shell is analyzed using the exact theory of three-dimensional elasticity. The two lamina are perfectly bonded having the same lateral dimension but have different radii and material properties. The finite shell is submerged in an infinite fluid medium terminated by two semi-infinite rigid cylindrical baffles. The shell has shear-diaphragm supports at the ends $z = 0$, L and is internally filled with an acoustic medium. It is insonified by an incident plane wave at an oblique incidence angle. The scattered acoustic farfield is evaluated for various incident wave wavenumbers, shell thicknesses, shell dimensions, radii, and material properties. A uniform steel shell in water was initially analyzed to study the influence of shell geometries on the scattered acoustic farfield. A second shell made up of an outer elastomer shell bonded to an inner steel shell was also analyzed to study the influence of elastomeric properties on acoustic scattering. [Work supported by NAVSEA Division Newport under ONR Summer Faculty Program.]

Contributed Papers

9:50

2aEA5. Ultrasonic phased array techniques for composite material evaluation. Hossein Taheri, Fereidoon Delfanian, and Jikai Du (Mech. Eng., South Dakota State Univ., Box 2219 SCEH 216, Brookings, SD 57006, hossein.taheri@sdstate.edu)

The increasing rate of composite materials usage in the industries and researches implies finding the suitable method for testing and evaluation of composite materials. The composites are susceptible to flaws during production and the inspection costs suggest the use of NDT methods. Non-destructive testing is an appropriate method to detect the flaws and anomalies in the materials, however, due to an-isotropic structure of the composite materials, it is required to modify the techniques to find the most appropriate and accurate process for detecting flaws and anomalies in composite materials. The objective of this paper is to evaluate the feasibility and accuracy of nondestructive testing methods with emphasis on ultrasonic phased array technique for the integrity and structural evaluation of composite materials. In this approach three different composite samples were used for testing including one carbon fiber and two glass fiber samples. ultrasound phased array technique evaluated to perform the testing on the composite materials to find the appropriate method and procedure to apply ultrasound phased array techniques on composite materials.

10:05

2aEA6. Elastic response of an orthogonally reinforced plate. Andrew J. Hull (Autonomous Systems and Technol. Dept., Naval Undersea Warfare Ctr., 1176 Howell St., Newport, RI 02841, andrew.hull@navy.mil) and Jason M. Maguire (Sensors and Sonar Systems Dept., Naval Undersea Warfare Ctr., Newport, RI)

This paper develops a three-dimensional fully elastic analytical model of a solid plate that has two sets of embedded, equally spaced stiffeners that are orthogonal to each other. The dynamics of the solid plate are based on the Navier-Cauchy equations of motion of an elastic body. This equation is solved with unknown wave propagation coefficients at two locations, one solution for the volume above the stiffeners and the second solution for the volume below the stiffeners. The forces that the stiffeners exert on the solid body are derived using beam and bar equations of motion. Stress and continuity equations are then written at the boundaries, and these include the stiffener forces acting on the solid. A two-dimensional orthogonalization procedure is developed, and this produces an infinite number of double indexed algebraic equations. These are all written together as a global system matrix. This matrix can be truncated and solved resulting in a solution to the wave propagation coefficients, which allows the systems displacements to be determined. The model is verified by comparison to thin plate theory and finite element analysis. An example problem is formulated. Convergence of the series solution is discussed.

10:20

2aEA7. On fluid-structure interactions of a cloaked submerged spherical shell. Clyde Scandrett (Appl. Mathematics, Naval Postgrad. School, Spanagel Hall Bldg. 232, Monterey, CA 93943, clscandr@nps.edu) and Ana M. Vieira (Naval Res. Ctr., Lisbon, Portugal)

Backscattering from a cloaked submerged spherical shell is analyzed in the low, mid, and high frequency regimes. Complex poles of the scattered pressure amplitudes using Cauchy residue theory are evaluated in an effort to explain dominant features of the scattered pressure and how they are affected by the introduction of a cloak. The methodology used is similar to that performed by Sammelmann and Hackman in a series of papers written on scattering from an uncloaked spherical shell in the early 1990s. In general, it is found that cloaking has the effect of diminishing the amplitude and shifting tonal backscatter responses. Extreme changes of normal and tangential fluid phase velocities at the fluid-solid interface when cloaking is employed leads to elimination of the "mid-frequency enhancement" near the coincidence frequency for even modestly effective cloaks, while reduction of the "high-frequency enhancement" resulting from the "thickness quasi-resonance" near the cutoff frequency of the symmetric (S_B^2) mode require more effective cloaking, but can be practically eliminated by employing a cloak that creates tangential acoustic velocities in excess of the S_B^2 mode phase speed near cutoff.

TUESDAY MORNING, 3 DECEMBER 2013 CONTINENTAL 5, 8:00 A.M. TO 11:25 A.M.

Session 2aED

Education in Acoustics: Engaging and Effective Teaching Methods in Acoustics

Michelle C. Vigeant, Cochair
Graduate Prog. in Acoustics, Penn State Univ., State College, PA 16801

Preston S. Wilson, Cochair
Mech. Eng., Univ. of Texas at Austin, 1 University Station, C2200, Austin, TX 78712-0292

Chair's Introduction—8:00

Invited Papers

8:05

2aED1. Community-based inquiry: An example involving wind turbine noise. Andrew A. Piacsek and Bruce Palmquist (Phys., Central Washington Univ., 400 E. University Way, Ellensburg, WA 98926, piacsek@cwu.edu)

Community-based inquiry (CBI) is an approach to providing an active and collaborative learning environment for students in a science classroom. Inquiry-based learning has been shown to improve critical thinking skills, as well as retention of new concepts [Quitadamo *et al*., Life Sci. Educ. **7**, 327–337 (2008)]. For a typical CBI project, all students in a class work simultaneously on single problem that has significance to the local community. The problem is generally structured as an open-ended investigation that incorporates concepts covered in class. Specific components of the problem can be assigned to groups of students; collaboration among groups helps the class arrive at "big picture" conclusions. At Central Washington University, the introductory calculus-based physics course was recently revised to incorporate a CBI project involving the acoustics of local wind turbines. Wind speed and sound level measurements were taken by faculty at two different sites under different weather conditions. Each student group chose a subset of the data to analyze, collaborating via an Internet-based shared document platform. The design and implementation of this project, as well some assessment results, will be discussed.

8:25

2aED2. Modifying the just in time teaching strategy for introductory acoustics. Andrew C. Morrison (Natural Sci. Dept., Joliet Junior College, 1215 Houbolt Rd.,Joliet, IL 60431, amorrison@jjc.edu)

Just-in-time-teaching (JiTT) is a teaching and learning strategy where a feedback loop is created between instructor and students in order to focus classroom activities toward concepts the class is having the most difficulty in understanding. The two parts of a JiTT strategy are short exercises done outside of class and the response to the exercises in class. In previous iterations of my introductory course on acoustics, students have been assigned readings to complete and questions to answer based on the assigned readings. Students are required to answer all questions online before the start of class. The instructor is responsible for reading student responses before the start of class and has the opportunity to adjust class activities to address concerns raised by students in their answers. This year, we are modifying the JiTT method to encourage student reflection on the activities completed in class and emphasize critical reading skills. The benefits and challenges of the JiTT method and present modifications are discussed.

8:45

2aED3. Using acoustics to enhance physics education. Tracianne B. Neilsen and Kent L. Gee (Brigham Young Univ., N311 ESC, Provo, UT 84602, tbn@byu.edu)

The authors were recently invited to write a Resource Letter for the *American Journal of Physics* on the role of acoustics in enriching physics education. The Letter contains citations related to textbooks, physical and virtual acoustics demonstrations, software, and instances where acoustical examples can lead to a deeper understanding of, e.g., general wave phenomena. This talk provides a summary of the diverse nature of resources that exist as well as specific tools and examples.

9:05

2aED4. Techniques used to promote active learning in an undergraduate architectural acoustics course. Michelle C. Vigeant (Graduate Program in Acoust. & Architectural Eng. Dept., The Penn State Univ., Penn State Univ., State College, PA 16801, michelle.vigeant@gmail.com)

The five-year undergraduate architectural engineering (AE) program at Penn State requires all students to take an introductory course in architectural acoustics (AA) and for most students in the program, no further AA courses are required. As a result, it is very important to capture the students' interest early on in the course and use strategies that will improve long-term retention of the material. A number of techniques were used to improve student engagement with the introductory AA course that had an enrollment of 94 students. The first homework assignment was to summarize a recent magazine article in any area of acoustics, which allowed the students to immediately see a real-world application of the subject. During lecture periods, students were encouraged to participate through the use of personal response devices, also known as "clickers." Several different types of clicker questions were used, including review questions from the preceding lecture, questions to prompt participation during the presentation of example problems, and conceptual questions. The results from the multiple choice clicker questions are immediately displayed, which provides useful feedback to both the students and the instructor. Some hands-on activities during the practicum periods were also incorporated to reinforce the content from the lectures.

9:25

2aED5. Experimental learning in acoustics: A project oriented graduate course on architectural acoustics. Michael R. Haberman (Appl. Res. Labs. and Dept. of Mech. Eng., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, haberman@arlut.utexas.edu)

The idea of learning-by-doing, known in educational theory as experimental learning, has wide acceptance as an efficient and effective teaching method. The subject of architectural acoustics is particularly well suited for this type of instruction. This talk reports on the current graduate course on architectural acoustics in the graduate acoustics program at The University of Texas at Austin, which attempts to implement an experimental learning approach. The course focuses on in-class demonstrations, site visits, and an extended course project to supplement traditional lectures. The talk will demonstrate how this approach satisfies an experimental learning model which emphasizes a cyclic process of abstract conceptualization (lectures on acoustical theory), experimentation and experience (demonstrations, visits, and projects), and reflective observation (reporting and testing). Specific example course projects are provided that illustrate the breadth and depth of course projects resulting from this approach and perspectives are provided for avenues for improvement using this teaching method.

9:45

2aED6. Development of educational stations for the Acoustical Society of America outreach activities. Cameron T. Vongsawad, Tracianne B. Neilsen, and Kent L. Gee (Phys. & Astronomy, Brigham Young Univ., 1041 E. Briar Ave., Provo, UT 84604, cvongsawad@byu.net)

One of ASA's outreach activities is hands-on exploration workshops for local school groups and girl scouts troops held during our semiannual conferences. In general, outreach programs have three main purposes: (1) public service, (2) generate enthusiasm and interest, and (3) supplement learning. Despite the good work that has been done in the past, it is apparent that the goals of the current ASA outreach activities could be better achieved through increased efficiency. This project continues the development of the ASA outreach programs held in conjunction with our meetings to better meet the above goals. Specifically, we have developed a structure in which demonstrations are grouped into five-minute stations each pertaining to a different physical system. To increase the ease with which volunteers can assist with the stations, one-page summaries and reference posters outlining the basic principals of each station have been prepared. Volunteers are encouraged to be guided by the interests of the students in their interactive discussions. In addition, we have created a brief introductory presentation for the workshops to explain what can be done in the field of acoustics. These improvements not only provide opportunities to excite student interest but also increase the efficacy of the outreach efforts.

10:05–10:20 Break

10:20

2aED7. Computer modeling as teaching tool in underwater acoustics. Marcia J. Isakson (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78713, misakson@arlut.utexas.edu)

One of the challenges of teaching acoustics is to bridge the gap between theory and application. However, laboratory experiments are often too difficult or costly particularly for large-scale phenomena such as ray bending in ocean acoustics. Computer modeling is a low cost, easily accessible way for students to visualize acoustic phenomena without the overhead of laboratory experiments. In the underwater acoustics course at the University of Texas at Austin, students develop an acoustic modeling suite throughout the course culminating in a code to solve the sonar equation for a target in a shallow water environment. Beginning with simple attenuation, students model rays bending due to sound speed profiles, modes, both propagating and leaky, reverberation, target scattering, array steering, and signal processing. Each component builds on previous code and, at the end of the course, the student has a coherent collection of underwater acoustic modeling tools.

Contributed Papers

10:40

2aED8. Using game-based learning with integrated computer simulation to teach core concepts in underwater acoustics. Jason E. Summers (Appl. Res. in Acoust. LLC, 1222 4th St. SW, Washington, DC 20024-2302, jason.e.summers@ariacoustics.com)

Game-based learning promises enhanced student engagement coupled with effective pedagogy, which motivated development of WaveQuest, a computer game that employs simulation to assist secondary students in learning core concepts from underwater acoustics. Prior work in physics education found simulations can aid learning by providing environments for inquiry that, through constraints on action imposed by concrete connection to the real world, implicitly scaffold the learning process, yet also support clarifying representations of phenomena unavailable to observers in the real world. The interactive 3D narrative game developed in this work provides motivational context for computer simulations and clarifying representations, which enables and assists students in pursing inquiry-based exploration of the simulated physical environment. At the same time, the level progression and game play naturally scaffold an adaptive learning process. Beyond facilitating this inquiry-based learning process, the game mechanics have been designed to incentivize players toward internalizing the novel concepts and models that are presented, in order to facilitate more rapid acquisition of new cognitive schema. This presentation will describe and demonstrate key elements of the game, with particular attention given to illustrating their relation to this pedagogical strategy. [Work supported by ONR Code 321 US.]

10:55

2aED9. Development of an online undergraduate course in acoustics at the Berklee College of Music. Eric L. Reuter (Liberal Arts, Berklee College of Music, 1140 Boylston St., Boston, MA 02215, ereuter@berklee.edu)

Between 125 and 150 students take an introductory course in acoustics each semester at the Berklee College of Music in Boston, MA. As the popularity of acoustics has grown (owing in part to a new minor in acoustics and electronics) limited classroom facilities have made it difficult to meet the demand for sections of the introductory class. Development is nearly complete on a fully online version of the course that will be offered beginning in 2014. This paper will examine the curriculum, development of media assets, and various hurdles in the development process.

11:10

2aED10. A water-level controlled wind instrument developed by a team-based project for an undergraduate acoustic class. Seung-Cheol Kang and Dong-Guk Paeng (Ocean System Eng., Jeju National Univ., 102 Jejudaehakno, Jeju 690-756, South Korea, stealyour@nate.com)

During an introductory undergraduate class of acoustics, a team-based project was given to design and implement their own musical instrument, after studying basic concepts of resonance, standing wave, and modes of string, pipe, bar, membrane, and plate. One of the teams designed and implemented "W.L.C WINDY (Water-Level Controlled Wind Instrument)," which the length of the pipe can be adjusted by water level inside the pipe. The lengths satisfies the modes of a stopped pipe (open-closed end) with end correction. There are several advantages of this instrument for education of acoustics; cheap, easy to make, and easy to understand the principles of modes. It costs less than one dollar (a transparency hose, a syringe, an O-ring) to make "W.L.C WINDY". This is how to make the instrument; 1. Connect a transparent hose with a syringe without needle using an O-ring. 2. Fill the hose with colored water for easy visualization. 3. Press the thumb rest of the syringe to make various sounds by controlling the water level. Even though W.L.C WINDY is not optimized to play music, it may contribute to easy demonstration of the principles of the modes for education of acoustics. [Supporting program: NIPA-2013-H0401-13-1007.]

TUESDAY MORNING, 3 DECEMBER 2013 — PLAZA A, 8:00 A.M. TO 12:00 NOON

Session 2aID

Interdisciplinary: Academic Genealogy (Poster Session)

Steven L. Garrett, Chair
Grad. Prog. in Acoustics, Penn State, Appl. Res. Lab., P. O. Box 30, State College, PA 16804-0030

Contributed Papers

All posters will be on display from 8:00 a.m. to 12:00 noon. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 8:00 a.m. to 10:00 a.m. and contributors of even-numbered papers will be at their posters from 10:00 a.m. to 12:00 noon.

2aID1. The Ear Club: Ervin R. Hafter's academic family. Frederick J. Gallun (National Ctr. for Rehabilitative Auditory Res., Portland VA Medical Ctr., 3710 SW US Veterans Hospital Rd., Portland, OR 97239, Frederick.Gallun@va.gov), G. Christopher Stecker (Dept. of Hearing and Speech Sci., Vanderbilt Univ. Medical Ctr., Nashville, TN), Psyche Loui (Dept. of Psych. and Program in Neurosci. and Behavior, Wesleyan Univ., Middletown, CT), and Anastasios Sarampalis (Experimental and Work Psych., Univ. of Groningen, Groningen, Netherlands)

Ervin R. Hafter is a direct academic descendent of Wilhelm Wundt, William James, James Cattell, Robert Woodworth, and Warner Brown. Erv's Ph.D. advisor, Lloyd Jeffress, is one of the most renowned names in binaural hearing theory. For more than 40 years, auditory scientists from around the globe have been traveling to Berkeley, where Erv has been teaching and researching at the University of California, Berkeley, mentoring undergraduate students and Ph.D. students and supervising postdocs. Many of those who worked in Erv's lab have gone on to run labs of their own and have graduated and supervised many of their own students and postdocs, who in turn now run their own labs. This presentation will document the extensive web of connections that all started with the Hafter Lab and the weekly seminar series, the Ear Club, where for decades auditory scientists of all ranges of background and experience have been coming together to talk, listen, and make friends for a lifetime.

2aID2. Bob Apfel: Acoustics at Yale University and beyond. Jeffrey A. Ketterling (Lizzi Ctr. for Biomedical Eng., Riverside Res., 156 William St., New York, NY 10038, jketterling@riversideresearch.org), Christy K. Holland (Cardiovascular Inst., Internal Med., Univ. of Cincinnati, Cincinnati, OH), Ronald A. Roy (Dept. of Mech. Eng., Boston Univ., Boston, MA), and E. Carr Everbach (Eng., Swarthmore College, Swarthmore, PA)

Bob Apfel passed away in 2002 after having taught at Yale University for over thirty years. During his tenure at Yale, Bob graduated 28 Ph.D. students, trained 12 post-doctoral students and influenced countless masters and undergraduate students. Bob devoted considerable time and effort to the Acoustical Society of America (his honors include the Gold Medal for Lifetime Achievement, Silver Medal in Physical Acoustics, and the Biennial Award) and his influence is still felt by the continued involvement of many of his students. Prior to his time at Yale University, Bob trained at Harvard University with Frederic V. Hunt and the chain of advisors can be loosely traced back, all at Harvard, to 1802. Although Bob's direct impact on the field of acoustics was prematurely cut short in 2002, his legacy and influence are still growing through the contributions of his students, and the students of his students.

2aID3. The acoustic genealogy of the Nebraska Acoustics Group at the University of Nebraska—Lincoln. Lily M. Wang (Durham School of Architectural Eng. and Construction, Univ. of Nebraska - Lincoln, PKI 101A, 1110 S. 67th St., Omaha, NE 68182-0816, lwang4@unl.edu)

The Nebraska Acoustics Group at the University of Nebraska—Lincoln was founded in the year 2000 with the hiring of Dr. Lily Wang into the newly formed Architectural Engineering Program. This poster will review the acoustic genealogy extending to and from this acoustics group, connecting to a number of other acousticians at academic institutions including Princeton University, Georgia Institute of Technology, Pennsylvania State University, Technical University of Denmark, Vassar College, University of Hartford, and Colombia College in Chicago. Connections to architectural acoustic consulting firms will also be highlighted.

2aID4. Greetings from the West: The people of Veneklasen Associates. Jerry P. Christoff, John J. LoVerde, and Joe Ortega (Veneklasen Assoc., 1711 16th St., Santa Monica, CA 90404, jchristoff@veneklasen.com)

Western Electro-Acoustic Laboratory (WEAL) was founded in 1947 by Paul S. Veneklasen, FASA, for the manufacture of precision acoustical instruments utilizing the Western Electric 640AA Precision Condenser Microphone. When asked in 1951 to provide acoustical consulting services on a jet engine test cell, the architect insisted on dealing with an individual, not a laboratory, and Paul S. Veneklasen & Associates (PSVA) was born. Renamed Veneklasen Associates (VA) after Paul Veneklasen's death in 1996, VA and WEAL have had the privilege of employing excellent associates over 65 years of business. Many within this group have contributed to the science of acoustics, in several cases founding firms to continue the practice of acoustical consulting. Others like the principal author, employed since 1956, have remained at VA continuing to contribute and mentor others in the science of acoustics. All of us have benefited from Paul Veneklasen's approach to consulting projects, his use of corroborating measurements, reporting style and the quest for solutions to acoustical problems. In addition, the firm carries on Paul Veneklasen's tradition of in-house acoustical research to advance the field. The Poster traces this history in terms of its current and former employees and their careers with PSVA and VA.

2aID5. Katherine Safford Harris' academic family tree. Fredericka Bell-Berti (Commun. Sci. and Disord., St. John's Univ., 8000 Utopia Parkway, Queens, NY 11439, bellf@stjohns.edu), Lawrence J. Raphael (Commun. Sci. and Disord., Adelphi Univ., Garden City, NY), and Diane Kewley-Port (Speech and Hearing Sci., Indiana Univ., Bloomington, IN)

At Haskins Laboratories, Katherine Harris lead a research team of engineers, programmers, technicians, ENT physicians, and fellow scientists to study speech production. As the reputation of this innovative research program grew, students, post-doctoral fellows, and visiting scientists from many universities and international laboratories came to join Kathy in this effort. Her contributions to training doctoral students grew when she became a faculty member at the CUNY Graduate Center. She was a pioneer role model especially—but not exclusively—for women scientists. Her enthusiasm and dedication to science attracted a host of students and young scientists who have gone on to academic research careers in Speech Communication. Kathy nurtured her students, always ready to listen and encourage, guiding but never pushing them through their degrees, and always keeping sight of the person.

2aID6. One view of the Rudnick-Putterman UCLA-Acoustics dynasties. Steven L. Garrett (Grad. Prog. in Acoust., Penn State, Appl. Res. Lab., P. O. Box 30, State College, PA 16804, sxg185@psu.edu)

During the second half of the 20th century, the research group headed by Isadore Rudnick, augmented, then extended to this day by Seth Putterman, brought the formalism and techniques of classical acoustics to problems in the quantum mechanics of condensed matter systems. In doing so, Izzy advised 32 Ph.D. students who graduated between 1951 and 1986, half of which went on to academic careers of their own (excluding two of his sons who are also academic scientists), making the UCLA-Acoustics "family tree" extraordinarily dense and complex. Although Putterman's academic roots can be traced all the way back to Stefan, Boltzmann, Ehrenfest, and Uhlenbeck, it appears that Rudnick's inspiration was divine in its origin. Traveling forward from my time as a Rudnick-Putterman graduate student in the mid-1970's, several clusters of my "academic siblings" (fellow graduate students and post-docs) can be identified that exerted a strong influence on acoustics research and education at the Naval Postgraduate School and Penn State and were influenced by the Swift-Wheatley-Migliori (Thermoacoustics) Group at Los Alamos National Laboratory.

2aID7. Acoustic lineage of the electroacoustic research laboratory at University of Massachusetts Dartmouth. David A. Brown (ATMC/ECE, Univ. of Massachusetts Dartmouth, 151 Martine St., Fall River, MA 02723, dbAcoustics@cox.net)

This poster summaries the academic and industrial lineage of the electroacoustics research laboratory at the ATMC-UMass focusing on the influence and training of mentor Boris Aronov, his use of the energy method for solving transducer problems, and a summary of some of the new generation of students and their research projects. Aronov adapted variational methods and generalized coordinates to solve transducer problems using an energy-balance approach in the 1950's in Russia under the tutorage of Lev Gutin at the Morphyspribor (Marine Physics Research Center), the soviet navy's main Sonar Research Institute located in Leningrad (now Saint Petersburg). Shortly after immigrating to the United States in 1998, Aronov teamed up with David Brown at UMass Dartmouth and BTech Acoustics LLC, building a fruitful partnership involving undergraduate and graduate students focusing on underwater electroacoustic transducer and electroacoustic subsystem problems—ranging from basic research, materials characterization, transducer and subsystem modeling, prototype development, calibration, and manufacturing.

2aID8. Academic genealogy of Lawrence A. Crum. Lawrence A. Crum (Appl. Phys. Lab., Univ. of Washington, Ctr. for Industrial and Medical Ultrasound, Seattle, WA 98105, lac@apl.washington.edu)

Lawrence Crum received his Ph.D. in physics from Ohio University in 1967; his advisor was Prof. F. Burt Stumpf. He studied as a postdoc under Prof. F. V. Hunt at Harvard from 1967 to 1968, and held faculty positions in physics at the U.S. Naval Academy from 1968 to 1978, where he supervised Trident Scholars—selected undergraduates who were able to perform the equivalent of a MS thesis. From 1978 to 1992, he held positions in the Department of Physics at the University of Mississippi, where he supervised a number of graduate students toward advanced degrees. Since 1992, he has held positions in the Departments of Electrical Engineering and Bioengineering at the University of Washington, where he has continued to supervise students toward advanced degrees. This poster will attempt to document the many students who have studied under Dr. Crum's tenure during his academic career.

2aID9. Historical perspective on R. Bruce Lindsay, A. O. Williams, R. T. Beyer, and P. J. Westervelt: Connections and branches surrounding the Brown era. Murray S. Korman (Phys. Dept., U.S. Naval Acad., 572 C Holloway Rd., Chauvenet Hall Rm. 295, Annapolis, MD 21402, korman@usna.edu)

Starting from a timeline when R(obert) Bruce Lindsay graduated from Brown (1920), there is a lineage of so-called great-grandfather, grandfather, and father advisors leading up to his own research, and branches connecting new Ph.D.'s. His Ph.D. on the atomic models of the alkali metals began in Copenhagen, Denmark, under Niels Bohr and H.A. Kramers in 1922, and was completed at M.I.T. in 1924. Bruce Lindsay returned to Brown University in 1930 (from Yale), became chair in 1934 and Hazard professor in 1936. His interests in atomic physics switched to acoustics during and after WWII, obtained strong support from ONR and kindled a legendary relationship with the ASA, including 20 years as Editor-in Chief, and a gold medal. He brought on board Art Williams (1940), Robert Beyer (1945), and Peter Westervelt (1951). They also received medals. R.T. Beyer's advisor at Cornell, Harry Sack, was a former student of Peter Debye. A.O. William's and P. J. Westervelt's advisors were Bruce Lindsay and Richard Bolt (M.I.T.), respectively. This poster presentation will also include threading together acoustics at Brown in an era including Jordan Markham, Bob Morse, Rohn Truell, Bruce Chick, Wes Nyborg, Charles Elbaum, and Humphrey Maris.

2aID10. Percy Wilson, acoustical consultant. Geoffrey L. Wilson (Acoust. (retired), Penn State Univ., 441 West Nittany Ave., State College, PA 16801-4057, glw4@psu.edu)

Percy Wilson was Technical Adviser to The Gramophone magazine in England from 1925 to 1938. He wrote a monthly column called Technical Talk and also wrote articles for the Wireless Magazine. He was very well known in the UK Audio industry. He was a senior civil servant at the Board of Education and in 1937 was appointed Assistant Secretary of the Ministry of Transport, in administrative charge of the trunk roads division. Because of pressures of work due to the probable imminence of war with Germany, he was forced to give up most of his outside activities, but was able to resume them after the end of WW2 and his subsequent retirement from the Civil Service. He was Technical Editor of The Gramophone for many more years, and was active in the Audio Engineering Society, being one of the founders of the British Section. As the self-appointed family historian I will endeavor to present highlights of his life from the time of his birth in Halifax, Yorkshire on March 3, 1893, to his demise in Oxford on April 30, 1977, focusing primarily on his technical work.

2aID11. The academic family tree of Ingo Titze. Brad Story (Speech, Lang., and Hearing Sci., Univ. of Arizona, 1131 E. 2nd St., Rm. 214, Tucson, AZ 85721, bstory@u.arizona.edu), Eric Hunter (National Ctr. for Voice and Speech, Salt Lake City, UT), and Ronald Scherer (Dept. of Commun. Sci. and Disord., Bowling Green State Univ., Bowling Green, OH)

Pursuing a deep interest in understanding the complexities of human sound production, particularly singing, Ingo Titze received his Ph.D. in Physics under the tutelage of Bill Strong at Brigham Young University in 1972. His dissertation on computational modeling of vocal fold vibration launched a career in research and teaching that has had a profound influence on the field of voice and speech science. Although his research alone is prodigious, Ingo Titze created an academic family that stretches far and wide. This presentation will trace the careers and research of the many students and scientists that were trained by Ingo Titze or influenced by his training.

2aID12. Genealogy of the National Center for Physical Acoustics. Josh R. Gladden, James Sabatier, and Henry E. Bass (Phys. & NCPA, Univ. of MS, 108 Lewis Hall, University, MS 38677, jgladden@olemiss.edu)

NCPA traces its roots to the Physical Acoustics Research Group which grew from the Department of Physics and Astronomy. The current Department of Physics and Astronomy acoustics effort was initiated with Prof. F. Douglas Shields who received his Ph.D. at Vanderbilt University under the direction of Prof. Robert Lageman. Professor Shields came to the University of Mississippi in 1959. He directed the dissertation of 10 Ph.D. students and seven Masters students. In 1964, Prof. Shields was joined in the department by Prof. Roy Arnold who studied the ultrasonic properties of solids. Professor Arnold directed the dissertation of two Ph.D. students and six Masters students. In 1970, Prof. Randall Peters and Prof. Henry Bass joined the faculty; Bass doing acoustics in gases with a Ph.D. from Oklahoma State under Tom Winter and Peters doing solid state acoustics. The following year, Ron Carter joined the group doing solid state acoustics. Before his passing in 2008, Professor Bass directed the dissertation of nineteen Ph.D. students and eight Masters students. Professor Larry Crum, a student of Prof. Stump at Ohio State, joined the faculty in 1978. In 1986, an act of Congress established the National Center for Physical Acoustics and over a period of two years, PARGUM was transferred to NCPA. Professor Ralph Goodman joined NCPA as Director in 1989. A listing of all additional faculty and scientists to join NCPA and the students directed by all will be updated to 2013 and presented.

2aID13. Genealogy of Kenneth N. Stevens and the Speech Communication Group at the Massachusetts Institute of Technology. Helen Hanson (ECE Dept., Union College, 807 Union St., Schenectady, NY 12308, helen.hanson@alum.mit.edu) and Stefanie Shattuck-Hufnagel (Speech Commun. Group, MIT Res. Lab. of Electronics, Cambridge, MA)

Ken often told a story to his advisees, illustrating how a seemingly insignificant event could change the course of a life. When Ken began his graduate studies at MIT, Leo Beranek learned that he had taken an acoustics class as an undergraduate and recruited him as a teaching assistant. Beranek was doing research on speech acoustics, and soon Ken was one of his research assistants. Thus, Ken's decision to take an undergraduate acoustics course led him to his lifetime of work in speech communication, in which he found boundless pleasure. After completing his Sc.D. in 1952, Ken joined the MIT Faculty. His first doctoral student was James Flanagan. He supervised at least 31 more doctoral students and scores of post-doctoral, masters, undergraduate, and visiting students over 53 years. Ken collaborated with MIT colleagues (e.g., Morris Halle and Jay Keyser, Linguistics Dept.) and many researchers well beyond 77 Massachusetts Ave (e.g., Gunnar Fant, KTH, Sweden). His interest in how the physics of the vocal tract interacts with contrastive phonological categories unequivocally changed the field of Speech Communication, as well as the lives of his many students and collaborators who have gone on to establish their own laboratories and research programs.

2aID14. Roots and branches of the acoustics program at Brigham Young University. Derek C. Thomas, Kent L. Gee, Tracianne B. Neilsen, Timothy W. Leishman, Scott D. Sommerfeldt (Dept. of Phys. and Astronomy, Brigham Young Univ., N223 ESC, Provo, UT 84602, derekcthomas@gmail.com), Jonathan D. Blotter, Scott L. Thomson (Dept. of Mech. Eng., Brigham Young Univ., Provo, UT), and William J. Strong (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT)

Acoustics research at Brigham Young University has a rich history extending back over half a century. The tradition of acoustics continues to this day as BYU is home to one of the largest student chapters of the ASA. The academic roots of current and former faculty will be presented. The locations of the branches (students) will also be presented to illustrate the growth and development of the program and the impact on the broader acoustics community.

2aID15. One descendant of David Blackstock of the University of Texas at Austin. Bart Lipkens (Mech. Eng., Western New England Univ., 1215 Wilbraham Rd., Box S-5024, Springfield, MA 01119, blipkens@wne.edu)

In the first academic genealogy session, held at the 153rd meeting of the Acoustical Society of America in Salt Lake City in 2007, David T. Blackstock displayed a poster detailing the Harvard academic tree leading back to Helmholtz [J. Acoust. Soc. Am. **121**(5, Pt. 2), (2007)]. David also included the tree going up and discussed his Ph.D. and M.S. students. The author of this poster, Bart Lipkens, was one of David's Ph.D. students. I was a faculty at Virginia Commonwealth University for two years, where I had one Ph.D. student, Shaozeng Dong. I then became a faculty at a predominantly undergraduate teaching institution, Western New England University in Springfield, Massachusetts. Western New England offers M.S. programs but limited Ph.D. opportunities. Even though my efforts are primarily focused on undergraduate teaching, I continue an acoustics research program, built mostly around M.S. and undergraduate students through a very active program of REUs (research experiences for undergraduates). This program has led several students to pursue further graduate education and careers in acoustics. A review of these students will be given.

Session 2aMU

Musical Acoustics and Education in Acoustics: Experiments and Laboratory Curricula for Undergraduate Courses in Musical Acoustics

Jack Dostal, Chair
Phys., Wake Forest Univ., P.O. Box 7507, Winston-Salem, NC 27109

Invited Papers

8:00

2aMU1. Making music: A serious attempt at designing new instruments by undergraduate students. Thomas R. Moore (Dept. of Phys., Rollins College, 1000 Holt Ave., Winter Park, FL 32789, tmoore@rollins.edu), Joshua Almond (Dept. of Art and Art History, Rollins College, Winter Park, FL), and Daniel Crozier (Dept. of Music, Rollins College, Winter Park, FL)

We describe a course in musical acoustics that required undergraduate students to design and build unique musical instruments, compose music for ensembles of them, and then perform the compositions in a public concert. Unlike most courses in musical acoustics which require the students to build home-made instruments as a final project, the construction of the instrument and composing original music were the primary goals of the course. The instruments were required to be artistic, visually interesting and play a pitch collection from the Western scale. We will describe the challenges and successes, show examples of the instruments, and review the lessons learned.

8:20

2aMU2. Flute measurements in a physics of music lab. Randy Worland (Phys., Univ. of Puget Sound, 1500 N. Warner, Tacoma, WA 98416, worland@pugetsound.edu)

Physics of Music students often benefit from laboratory exercises that make use of real musical instruments, in addition to the more traditional labs that are designed to illustrate physical principles as simply as possible. These "real instrument" labs help bridge the gap between idealized cases and the musical instruments the students are familiar with. Modern woodwinds are particularly challenging in this regard due to the complex set of keys, levers, and mechanical linkages that tend to obscure the underlying acoustics of these instruments. Among the woodwinds, the flute is relatively simple, both mechanically and acoustically, and thus provides an excellent subject for a real world woodwind study. Laboratory exercises are described in which the flute's tone hole locations and diameters are measured. The data are analyzed in terms of the acoustics of open cylindrical tubes, revealing the logical order behind the spacing and use of the holes to play the lowest chromatic octave, as well as the higher registers of the flute. The measurement techniques and analysis are presented along with the pedagogical role of the experiment.

8:40

2aMU3. An interactive audio synthesizer for investigating formants and timbre. Andrew A. Piacsek (Phys., Central Washington Univ., 400 E. University Way, Ellensburg, WA 98926, piacsek@cwu.edu)

Computer-based investigations of the nature of formants and their role in characterizing vowel sounds and the timbre of musical instruments will be described. One version of this activity requires students to obtain detailed spectra of different vowel sounds, measure relative amplitudes of harmonics in the recorded sounds, and perform additive synthesis using Mathematica to recreate the different vowels. Students can investigate the minimum number of harmonics and formants needed to distinguish vowels. A new version of this activity relies on a recently developed interactive audio synthesis program that reduces the emphasis on tedious measurements and the need to edit Mathematica notebooks, allowing students to devote intellectual effort on investigation. The new program and the associated lab activity will be tested on students during Fall quarter of 2013 as part of the "Physics of Musical Sound" course at Central Washington University.

Contributed Paper

9:00

2aMU4. A carillon bell laboratory in an introductory physics of music class. Jack Dostal (Phys., Wake Forest Univ., P.O. Box 7507, Winston-Salem, NC 27109, dostalja@wfu.edu)

The Physics of Music is a lecture and laboratory class which satisfies a quantitative physical science requirement. It is open to students of all majors. Students taking the class may or may not have musical backgrounds, and generally are not majors in the physical sciences. In this laboratory exercise, students record sounds of the bells of the Harris Carillon in the tower of WFU's Wait Chapel. Using Audacity and other software, students generate frequency spectra of different bells. The prominent presence of minor thirds in the spectrum makes for a useful comparison between spectra of bells and typical concert instruments. In this lab, students identify spectral features of individual bells, find similarities and differences between different bell spectra, and observe the decay of spectral features over time. In this talk, I will describe some of the activities performed by the students in the lab and comment on some of the challenges of performing this laboratory exercise with students of varied musical and scientific backgrounds.

Invited Paper

9:15

2aMU5. A laboratory investigation of the effect of trumpet mouthpieces in an introductory musical acoustics course. Andrew C. Morrison (Natural Sci. Dept., Joliet Junior College, 1215 Houbolt Rd.,Joliet, IL 60431, amorrison@jjc.edu)

In our introductory musical acoustics course for non-science majors, a major goal of the course is to introduce students to the principles of scientific thinking and investigation. Many students have preconceived ideas or beliefs about factors influencing the sounds produced by musical instruments. It is the role of the instructor to help students frame their ideas into scientifically testable experiments, which can be conducted in the laboratory. In our course, we have designed a laboratory activity to challenge students to explore the role of the mouthpiece of brass musical instruments. A parametric model of a trumpet mouthpiece is customized by students in order to investigate the effects of changing mouthpiece geometry on the trumpet acoustics. The models are manufactured on a 3D printer and used in the laboratory to make input impedance measurements. Students compare their quantitative results to the qualitative conceptions they held before completing the laboratory investigation.

TUESDAY MORNING, 3 DECEMBER 2013 CONTINENTAL 9, 8:10 A.M. TO 12:00 NOON

Session 2aNS

Noise and ASA Committee on Standards: Standardization in Soundscape

Brigitte Schulte-Fortkamp, Cochair
Technische Universität Berlin, Einsteinufer 25, Berlin 10178, Germany

Bennett M. Brooks, Cochair
Brooks Acoustics Corporation, 30 Lafayette Square - Ste. 103, Vernon, CT 06066

Chair's Introduction—8:10

Invited Papers

8:15

2aNS1. The urgent need for standardization of soundscape. Brigitte Schulte-Fortkamp (Inst. of Fluid Mech. and Eng. Acoust., TU Berlin, Einsteinufer 25, Berlin 101789, Germany, b.schulte-fortkamp@tu-berlin.de), Klaus Genuit, and André Fiebig (Tech. Management, HEAD Acoust. GmbH, Herzogenrath, Germany)

In recent years, the soundscape approach gains increasingly in importance. It has shown its potential dealing with environmental noise issues, which conventional noise control approaches frequently fail to cope with. The soundscape concept is applied to comprehend the way humans experience and understand their acoustic environments. It does not simply reduce noise to an averaged acoustic quantity evoking unpleasantness estimated by statistical probabilities, but it considers noise as a valuable resource, which can be purposefully utilized. The standardization of "soundscape" in ISO 12913 is currently in progress. The standard will include definitions, terms and a conceptual framework illustrating that soundscape goes beyond physics and psychoacoustics, since it reflects human perception including cognitive aspects, context and interaction. The paper describes the progress in standardizing soundscape and the implications of the standard with its potential relevance to noise policy. Moreover, it discusses how to meet the challenge to standardize and harmonize the soundscape approach without losing the openness and multidimensionality of the soundscape idea.

8:35

2aNS2. Types of soundwalks and their applications in soundscape design. Gary W. Siebein (Architecture, Univ. of Florida, 625 NW 60th St. Ste. C, Gainesville, FL 32607, gsiebein@siebeinacoustic.com), Keely M. Siebein, and Gary Siebein (Siebein Associates, Inc., Gainesville, FL)

Seven types of soundwalks are described with examples of the data obtained and the application of each to a case study project. The first type is one where an observer walks through an environment and just listens to get an introduction to the environment or an inspiration for later work. The second type is one where the observer walks with a group of stakeholders to listen to or elicit observations about the environment. The third type of soundwalk is one in which qualitative data in the form of questionnaire responses or quantitative data in the form of sound level measurements or recordings are made. The fourth type is an undercover soundwalk where observations or

data are recorded in a way that the observer is not identified as one is not a normal participant in the soundscape. The fifth type is one where stakeholders experience an existing environment that has elements being considered for use in their project. A sixth type of soundwalk is one that occurs in a computer simulation where one experiences a virtual environment. The seventh type is one that combines two or more of the individual types of soundwalks in a composite soundwalk.

8:55

2aNS3. Evaluation of indoor and outdoor soundscapes—The benefit of combining soundwalks and laboratory tests. Kay S. Voigt (Inst. of Fluid Mech. and Eng. Acoust., Technische Universität Berlin, Einsteinufer 25, Berlin 10587, Germany, kay.s.voigt@gmx.de)

Experiences and expectations of listeners have a significant influence to ratings of soundscapes. Sounds do not only occur on a single level of perception to peoples' ears, they appear with a setting of information of the source(s) to the "listening background" of the individual at a certain location. Although these possible combinations seem to be infinite, common judgments are unfolded on appropriateness of sounds on places. For the evaluation of two public indoor and two public outdoor locations in the city of Berlin, the soundwalk has been used with students in a series of five consecutive years to investigate predominant aspects of rating soundscapes. This tool provides a substantial data acquisition of the environment with technical measurements and documentations as well as ratings and appraisals by the participants. The data-triangulation enables to detect contextual moderators of the participants' evaluation by adjusting situational focus and importance. In subsequent laboratory tests—with the recorded stimuli—this connotation of non-acoustical features is discovered, especially by their absence, also the underlying attitude towards sound sources is identified. Finally, a reflecting discourse with the attendees on their ratings shows the need of elucidation of the meaning of individual use of words and phrases.

9:15

2aNS4. Combining elements of the soundscape—Lessons from the recording studio. Alex Case (Sound Recording Technol., Univ. of Massachusetts Lowell, 35 Wilder St., Ste. 3, Lowell, MA 01854, alex@fermata.biz)

So many soundscapes—found, archived, preserved, or composed—consist of multiple sound elements. The interaction among these individual sounds can conflict and obscure, or complement and harmonize. The creation of sound recordings through the multitrack production process offers a point of view into how an overall soundscape can be successfully created, and its multiple sounds effectively orchestrated.

9:35–9:45 Break

9:45

2aNS5. Structural equation modeling of soundscape perception based on urban contexts. Jin Yong Jeon and Joo Young Hong (Dept. of Architectural Eng., Hanyang Univ., 17 Haengdang-dong, Seongdong-gu, Seoul 133791, South Korea, jyjeon@hanyang.ac.kr)

The aim of this study is to explore influential factors on urban soundscape perception. Objective and subjective assessments of the soundscapes in various urban spaces were conducted to characterize soundscape contexts using soundwalks. Evaluations were conducted at 21 locations, including residential, office, park, and commercial spaces in Seoul: the participants assessed their perceptions of both sound and landscape environments. In addition, laboratory experiments using audio-visual stimuli were carried out to validate the dominant soundscape factors derived from the soundwalks. From the results, a perception model of urban soundscapes consisting of physical and subjective indicators was suggested using structural equation modeling.

10:05

2aNS6. Standards vs off-standards in the soundscape design process. Juergen Bauer (Dept. of Architecture, Waterford Inst. of Technol., Granary, Hanover St., Waterford, Co Waterford 00000, Ireland, jbauer@wit.ie)

Standards are widely acknowledged agreements that make issues and procedures comparable, compatible, and safe. Standards, therefore, help speak a common language and are considered a means of quality assurance, particularly in the field of design. However, in respect to Soundscape research being regarded as a subject of acoustic ecology, there is a limitation: How can standardization assess and evaluate Soundscape entities in an appropriate way which takes ecologic diversity into account? Can descriptors be found which indicate Soundscape qualities other than by measured sound pressure levels, or by surveyed human perception of pleasantness and annoyance? While it is fully acknowledged that these indicators will continue to play a key role in the discussion on Soundscape quality, they fail to address the distinction and the specific "atmosphere" of a specific Soundscape entity. "Atmosphere," as suggested by Gernot Boehme (1993), is "the inbetween" of environmental qualities and human states. By this definition, a specific "atmosphere" is bound to a specific space. This paper explores the potential of using descriptors of "atmosphere," such as in the architectural and urban design process, to help further expand the specification of Soundscape standards, but also to exploit the concept of off-standard Soundscape solutions.

10:25

2aNS7. Psychoacoustics triggering the soundscape standardization. Klaus Genuit (Tech. Management, HEAD Acoust. GmbH, Ebertstr. 30a, Herzogenrath 52134, Germany, klaus.genuit@head-acoustics.de), Brigitte Schulte-Fortkamp (Inst. Tech. Acoust., Tech. Univ. Berlin, Berlin, Germany), and André Fiebig (Tech. Management, HEAD Acoust. GmbH, Herzogenrath, Germany)

Before the start of COST Action TD0804 (network on soundscape of European cities and landscapes) in 2009, the Working Group 54 of ISO/TC 43/SC 1 was established to start consideration of a standardized protocol for assessment of soundscape. The progress is slow since many different protocols are available, which make it difficult to define a harmonized protocol reflecting diverse requirements. No protocol covers all needs, which is characteristic in research dealing with quality of life, where space, social and built environment and culture are often very different. Unfortunately, there is still a lack of open-mindedness to use new soundscape techniques, e.g., in noise action plans and for the protection of quiet areas. The situation differs, however, broadly from country to country. The

Working Group 54 of ISO/TC 43/SC 1 works on reaching consensus to provide the necessary stimulus for further worldwide progress. Currently, the ISO WG 54 is working on measurement procedures and techniques that will trigger the procedures in soundscape research. Psychoacoustics and explorative interviews play a major role, and the triangulation of the respective data is offering consolidated findings. This paper will present related techniques and will discuss their advantages.

Contributed Papers

10:45

2aNS8. Educating for soundscape design. Fernando J. Elizondo - Garza (Acoust. Lab., FIME-UANL, Calle 8 #422, Col. Villazul, San Nicolas, N.L. 66420, Mexico, fjelizon@hotmail.com)

The development of standardization on soundscapes leads to produce the need of experts and designers of acoustic spaces that be educated in a wide cultural and technological base. Although there is nothing new for well educated acousticians, we cannot forget that people involved in architectural or noise control problems are not always acousticians, thus they do not have a full scope of the possible solutions neither of the listeners expectations about sound environment. Engineering, technology, economics, society, diversity, education, psychology, communication, interdisciplinarity, workgroups, ethics, and other educational issues necessary for soundscapes design are discussed in this paper. Many of these aspects are difficult to be included in the curricula directly, so they must be handled as homework and extracurricular activities, under the direction of an acoustics professor interested in the generation of better sound spaces for the welfare of society.

11:00

2aNS9. Recent developments in an old problem: Helping communities deal with noise. Alice H. Suter (Suter & Associates, 1106 NE Tillamook St., Portland, OR 97212, ahsuter@comcast.net) and Monica S. Hammer (Network for Public Health Law, Univ. of Michigan School of Public Health, Ann Arbor, MI)

This presentation will suggest areas where professionals in acoustics can collaborate with policy makers to remediate their noise problems and promote an environment more conducive to the public health and welfare. Research over recent years has confirmed and elucidated the adverse effects of noise on health in ways that earlier cross-sectional studies had indicated, but with the added power of a prospective approach and extensive controls. The implications of these findings should be considered widespread and serious. Lacking a coordinated federal program, U.S. Cities and counties are attempting to grapple with their noise issues, without federal support and often with limited technical expertise. Despite passage of the Noise Control Act of 1972 (subsequently de-funded), millions of Americans remain exposed to levels of environmental noise that are harmful to their health and welfare.

11:15

2aNS10. Delivering noise information to communities in presentations and with asynchronous media. David Dubbink (Noise Management Inst., 864 Osos St., San Luis Obispo, CA 93401, dubbink@noisemanagement.com)

Working with communities to resolve noise problems demands not just accurate data about noise impacts but also effective means of exchanging information. The Interactive Sound Information System (ISIS), developed by the author, represents one means of presenting noise impacts data. Central to the ISIS concept is the idea that actual noise examples, shaped to reflect local situations, are the very best way to increase community understanding of noise management issues. Such a strategy works well in a public hearing or community workshop setting. Presentations can be structured to address local concerns and recorded sounds can be accurately calibrated. However, alternate modes of communicating information have been developed that are asynchronous and individual. The question is how to adapt presentations to such delivery systems and retain the reproducibility and accuracy of written reports or professionally moderated presentations. The paper describes current mechanisms for transmitting noise management data and explores the issues involved in adapting delivery to hand-held devices or other modes of individual access.

11:30–12:00 Panel Discussion

Session 2aPAa

Physical Acoustics and Structural Acoustics and Vibration: Acoustics of Pile Driving: Measurements, Models, and Mitigation I

Kevin M. Lee, Cochair
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Karl-Heinz Elmer, Cochair
OffNoise-Solutions GmbH, Leinstr. 36, Neustadt a. Rbge. 31535, Germany

Chair's Introduction—8:00

Invited Papers

8:05

2aPAa1. Offshore pile driving noise—Prediction through comprehensive model development. Marcel Ruhnau, Tristan Lippert, Kristof Heitmann, Stephan Lippert, and Otto von Estorff (Inst. of Modelling and Computation, Hamburg Univ. of Technol., Denickestraße 17, Hamburg, Hamburg 21073, Germany, marcel.ruhnau@tuhh.de)

Offshore wind energy is one of the most potent among renewables and thus the worldwide number of offshore wind turbines increases rapidly. The foundations of the wind turbines are typically fastened to the seabed by impact pile driving, which comes along with a significant amount of waterborne noise. To protect the marine biosphere, the use of noise mitigation systems, like bubble curtains or cofferdams, may become necessary. In this context, the model-based prediction of underwater sound pressure levels as well as the design and optimization of effective sound mitigation measures by using numerical models is one of today's challenges. The current work presents a modeling approach that consists of a near field finite element model and a far field propagation model. Furthermore, it has been found necessary to generate a benchmark to allow for a qualitative and quantitative comparison between the manifold modeling approaches that are currently developed at various institutes and companies.

8:25

2aPAa2. Model for underwater noise radiated by submerged piles. Todd Hay, Yurii A. Ilinskii, Kevin M. Lee, Evgenia A. Zabolotskaya, and Mark F. Hamilton (ARL:UT, Appl. Res. Labs., P.O. Box 8029, Austin, TX 78713-8029, hayta@arlut.utexas.edu)

There is concern that underwater noise generated by marine construction activities and radiated by towers supporting offshore wind turbines may disturb marine mammals, or interfere with passive sensors and communication equipment. In order to understand these effects a semi-analytic frequency-domain model was developed previously for the sound radiated in the water column by a pulsating cylindrical structure embedded in horizontally stratified layers of viscoelastic sediment. This model was in turn coupled to a parabolic equation code for long-range propagation over range-dependent environments [Hay *et al*., J. Acoust. Soc. Am. **133**, 3396 (2013)]. A time-domain version of this model is now presented which enables simulation of impulsive sound sources such as those due to underwater pile driving, and pulsed tonal sources appropriate for use in a finite-sized laboratory tank. In order to validate the model a scaled physical model, consisting of a laboratory tank and metallic cylindrical tube driven in the high kilohertz frequency range, was constructed. Simulations will be presented for a variety of sound sources, and preliminary comparisons with measurements from the scaled model experiments will be made.

8:45

2aPAa3. Effect of the transient fluid-structure interaction on sound radiation from a partially submerged cylindrical pile. Mardi C. Hastings (Georgia Inst. of Technol., George W. Woodruff School of Mech. Eng., 801 Ferst Dr., Atlanta, GA 30332-0405, mardi.hastings@gatech.edu)

A time-domain structural acoustics model of a partially submerged cylindrical pile excited by an impact force has been developed to predict underwater sound radiation for arbitrary field conditions. Radial boundary conditions are formulated using the velocity potential so pressure waveforms near the pile wall can be estimated; however, the rapid decay observed in measured waveforms is not easily predicted because radiation loading on the structure is also transient. The sudden acceleration of the wall initially results in a net flow of energy from the structure into surrounding fluids. Subsequently part of this energy radiates to the far field as a compressional wave; however, as the forced wall motion decreases and the spatial pressure gradient at the wall reverses direction, fluid accelerates back to the pile and re-excites the wall. A model for this transient fluid-structure interaction was formulated and integrated into the time-domain structural acoustics numerical model. By accounting for energy flow back into the structure, predicted pressure waveforms are in good agreement with field data. Results indicate that a significant portion of the kinetic energy remains in the near field and does not radiate sound, and that temporal characteristics of the impact force may influence far field sound radiation.

9:05

2aPAa4. Numerical simulations of hydrosound and ground vibrations during offshore pile driving. Christian Kuhn (Technische Universität Braunschweig, Inst. for Soil Mech. and Foundation Eng., Beethovenstr. 51 b, Braunschweig 38106, Germany, c.kuhn@tu-braunschweig.de)

Most foundations for offshore wind turbines are currently constructed with monopiles. During the installation, the monopile is driven into the ground using a hydraulic hammer. A pressure wave is induced into the pile and passes through the pile shaft which as a result expands radially and induces underwater noise in the surrounding water. Similarly, a shock wave is introduced into the ground which propagates there spherical. In a numerical simulation the driving process is simulated. The focus is on the energy radiated from the pile into the surrounding water-/ground continuum. As part of a research project, an offshore test was performed. During this, the underwater noise in the water as well as the soil vibrations on the seabed were measured. The numerical results are compared with these measurement results.

9:25

2aPAa5. A model for underwater sound levels generated by marine impact pile driving. Alexander O. MacGillivray (JASCO Appl. Sci., 2305–4464 Markham St., Victoria, BC V8Z7X8, Canada, alex@jasco.com)

Marine impact pile driving generates very high underwater sound pressures that can be harmful to aquatic life. Environmental assessments for pile driving projects typically require advance estimates of acoustic impact zones for marine mammals and fish. A computer model has been developed to predict the radiated acoustic field from impact driving of cylindrical piles. The stress wave in the pile is predicted using a force-generator model of the hammer-pile system. The force-generator model is coupled to a 1-D finite-difference model of longitudinal stress waves in a cylindrical pile. The radiated pressure is computed by matching the velocity boundary condition at the pile wall using a superposition of monopole sources distributed over the length of the pile in a layered 2-D fluid medium. The transfer function for the monopoles is computed using the near-field Hankel transform for radial particle velocity at the pile wall. Standard ocean acoustic modeling techniques are used to compute the Mach wave propagating away from the pile. As an example, predictions of the model are compared to field measurements obtained in a riverine environment.

9:45

2aPAa6. Prediction of offshore impact pile driving noise using numerical and analytical approach. Huikwan Kim, James H. Miller, and Gopu R. Potty (Ocean Eng., Univ. of Rhode Island, 215 South Ferry Rd., Narragansett, RI 02882, hkkim524@my.uri.edu)

Noise generated by offshore impact pile driving can radiate into and propagate through the air, water, and sediment. Predicting noise levels around the support structures at sea is required to estimate the effects of the noise on marine life. We have been investigated acoustic impact using our previously verified coupled Finite Element (Commercial FE code ABAQUS) and Monterey Miami Parabolic Equation (2D MMPE) models [J. Acoust. Soc. Am. **131**(4), 3392 (2012) and J. Acoust. Soc. Am. **133**, 3419 (2013)]. A simple analytical model for the impact pile driving noise is developed to reduce computation cost and modeling effort when compared to the numerical model. In the present study, we are developing a MATLAB based analytical model to calculate the Receive Levels (RL) in water. This analytical model solves simplified Donnell's equation, i.e., governing equation for the thin cylindrical shell using the modal superposition approach assuming azimuthal symmetry. It solves the Helmholtz or Kirchhoff integral equation to evaluate the acoustic pressure field. We will compare the results of the coupled FE-MMPE numerical model and the MATLAB based analytical model and discuss their advantages and disadvantages. [Work sponsored by the Link Foundation Ocean Engineering and Instrumentation Ph.D. Fellowship program.]

10:05–10:20 Break

10:20

2aPAa7. Toward a standard for measurement of underwater noise from marine impact piling. Stephen Robinson (National Physical Lab., Hampton Rd., Teddington TW110LW, United Kingdom, pdt@npl.co.uk), Christ de Jong (Acoust. and Sonar, TNO, The Hague, Netherlands), Andreas Mueller (Müller-BBM GmbH, Hamburg, Germany), and Pete Theobald (National Physical Lab., Teddington, United Kingdom)

Marine impact piling is a significant source of low-frequency impulsive noise in the ocean and is typically used to position piled foundations in relatively shallow water for offshore construction, for example, offshore wind farms, bridge supports, etc. However, there are currently no international standards to define appropriate measurement methodologies, and there is a need to underpin incipient regulation with appropriate measurement standards. In the scientific literature, attempts to report the measured noise levels can be difficult to compare because different metrics are often used (peak-to-peak pressure, rms pressure, sound exposure level, etc). Furthermore, simple assumptions about equivalent point sources are often used in measurements and modeling without sufficient validation. Agreed acoustic metrics and a common way of deriving a meaningful measure of source output are urgently required. This paper describes work to address this need within the International Standards Organisation, specifically Working Group 3 of Technical Committee 43, Sub-Committee 3 (ISO TC43 SC3 WG3). Work to develop a new standard builds upon the expertise already gained by researchers in a number of countries (for example, United Kingdom, Netherlands, and Germany). A description is given of the methodology currently proposed, and the process for engagement with experts within different countries.

10:40

2aPAa8. Results from background noise measurements in the North Sea. Max Schuster (DW-ShipConsult, Lise-Meitner-Str. 1-7, Schwentinental 24223, Germany, schuster@dw-sc.de)

In Germany, 12.8 GW offshore wind power shall be installed until 2023, which involves that at minimum 200 wind turbines need to be erected each year. As the majority of the turbines' foundations are based pile driving vast amounts of high level impulsive noise are introduced into the North- and Baltic Sea. Until today, research on pile driving noise largely focuses on sound propagation and noise mitigation in close vicinity of the pile. Further effects of impulsive noise in larger distances over 10 km are rather unknown due to a lack of knowledge on the propagation and on the effects of noise at levels below injury. The German Agency for Nature Conservation has founded a campaign to measure underwater noise in the marine protected areas of the German EEZ. In this course, 10 noise recorders are simultaneously deployed in the large protected area Sylt Outer Reef. At least three wind farms are to be installed in the area's vicinity; therefore, recordings of background noise during the installation phase allow a systematic investigation of impulsive noise propagation over large distances. Results in time domain and in frequency domain are compared with results from commonly available calculation codes.

11:00

2aPAa9. Properties influencing the transmission loss and mitigation of the underwater sound from marine pile driving. Peter H. Dahl (Appl. Phys. Lab. and Mech. Eng. Dept., Univ. of Washington, Seattle, WA 98105, dahl@apl.washington.edu)

The transmission loss of underwater sound generated by impact or vibratory pile driving, combined with the potential mitigation of this sound, ultimately determine the regulatory zone-of-impact on marine life. For impact pile driving the pile source assumes properties of a "phased array" as shown in Reinhall and Dahl [J. Acoust. Soc. Am. **130**, 1209–1216 (2011)], where sound radiation arises from the state of radial expansion which travels down the pile at a speed which is supersonic relative to the speed of sound in water. The acoustic field that develops is shown to be the dominant contributor to peak underwater sound pressure. An important range scale can be identified that is useful for both guiding measurements, and for transmission loss modeling. For vibratory pile driving, measurements made at close range (within about a one water depth) show considerable low-frequency content that can be below the cut-off frequency of the waveguide. Some key properties of impact and vibratory pile driving will be discussed that govern the nature of transmission loss from, and influence mitigation strategies for, these sound sources. These properties are based on measurements made at marine construction sites in Puget Sound, Washington.

Contributed Papers

11:20

2aPAa10. Transmission loss for vibratory pile driving in shallow water: Modeling and field measurements for a Puget Sound location. Dara M. Farrell and Peter H. Dahl (Appl. Phys. Lab., Dept. of Mech. Eng., Univ. of Washington, Henderson Hall, Seattle, WA 98105, daraf@uw.edu)

Measurements of the underwater noise from vibratory pile driving were collected at a marine construction site in January 2013, during which 0.76 m steel piles were driven in shallow water (less than 10 m). The sound field was simultaneously measured at three locations at distances 200 m (1 hydrophone) and 400 m (2 hydrophones) from the pile driving location and roughly 150 m to 300 m from the shore. In the Practical Spreading Model (PSM), transmission loss is 15 log10(R/R0): R is the range to which TL is calculated, and R0 is the range for a close range measurement, typically O(10 m) from the source. Models such as the PSM that do not account for bathymetric or sediment properties could not be expected to predict the anomalously high transmission loss (TL) that was observed between the 200 m location and one of the 400 m locations. Important bathymetric features (of order 10 m in range, 1 m in depth) and sedimentary properties for the area were incorporated into a Parabolic Equation model; modeled TL was compared to observed TL for third octave bands. Using a Geographic Information System tool, modeled TL was visualized for the area. [Work supported by Washington Sea Grant.]

11:35

2aPAa11. Depth dependence of the intensity vector from impact pile driving. David R. Dall'Osto and Peter H. Dahl (Mech. Eng., UW-Seattle, 914 N 38th St., Seattle, WA 98103, dallosto@u.washington.edu)

The direction of the time-integrated acoustic intensity vector defines active intensity streamlines which are perpendicular to this vector. For impact pile driving, these intensity streamlines depend on both depth and time. The time-integrated intensity vector (active intensity) has been shown to change direction over the initial downward traveling Mach cone, the upward traveling reflection of this from the bottom of the pile, and subsequent downward traveling reflection from the top of the pile, referred to as phases in Dahl and Reinhall [J. Acoust. Soc. Am. **134**, EL1 (2013)]. In this study, vertical line array data are used to demonstrate the depth dependence of the active intensity vector during each of the three phases. The vertical component of the active intensity is approximated between each pair of neighboring hydrophones by finite difference methods. A numerical approach based on parabolic wave equation simulations [Reinhall and Dahl, J. Acoust. Soc. Am. **130**, 1209 (2011)] is modified here and used to construct theoretical intensity streamlines, motivated by those depicted in Zampolli *et al.* [Acoust. Soc. Am. **133**, 72 (2013)]. These intensity streamlines are validated by the experimental measurements. Energy propagation through the water column and the sediment-water interface is also discussed.

Session 2aPAb

Physical Acoustics and Structural Acoustics and Vibration: Phononic Crystals and Metamaterials

Joel Mobley, Cochair
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Cecille Labuda, Cochair
National Ctr. for Physical Acoust., Univ. of Mississippi, 1 Coliseum Dr., University, MS 38677

Chair's Introduction—8:00

Invited Papers

8:05

2aPAb1. Coupled membranes with doubly negative mass density and bulk modulus. Ping Sheng (Dept. of Phys., Hong Kong Univ. of Sci. & Technol., Clear Water Bay, Kowloon, Hong Kong 000, China, sheng@ust.hk)

We present a structurally and conceptually simple acoustic double negative metamaterial comprising two coupled membranes [Phys. Rev. Lett. **110**, 134301 (2013)]. Owing to its symmetry, the system can generate both monopolar and dipolar resonances that are separately tunable, thereby making broadband double negativity possible. A homogenization scheme is implemented that enables the exact characterization of our metamaterial by the effective mass density and bulk modulus even beyond the usual long-wavelength regime, with the measured displacement fields on the sample's surfaces as inputs. Double negativity is achieved in the frequency range of 520–830 Hz. Transmission and reflection predictions using effective parameters are shown to agree remarkably well with the experiment. Work done in collaboration with M. Yang, G. C. Ma, and Z. Y. Yang.

8:25

2aPAb2. Acoustic cloaking for airborne sound based on inclusions of rigid scatterers. Victor Manuel Garcia-Chocano, Francisco Cervera, Ana Díaz-Rubio, Alfonso Climente, Daniel Torrent, and José Sánchez-Dehesa (Universitat Politècnica de València, C/Camino de Vera S/N, Departamento de Ingeniería Electrónica, Valencia 46022, Spain, vicgarch@upvnet.upv.es)

Acoustic cloaking is a phenomenon whose physical realization depends on the ability of designing metamaterials with the appropriate parameters. When dealing with airborne sound almost any solid behaves as an acoustically rigid material, so cloaks based on rigid scatterers are here studied. Since these structures are not able to increase the effective sound speed with respect to the background, additional mechanisms should be introduced to allow this effect. Here we will report an acoustic cloak based on temperature gradients. Another possibility of hiding objects from an external sound source consists of using a set of external layers that cancels the scattered field by such objects at selected frequencies. We present the practical realization of this approach for 2D and 3D structures. [Work supported by MINECO from Spain and ONR from United States.]

8:45

2aPAb3. Anomalous ultrasonic transport in phononic crystals with overlapping Bragg and hybridization gaps. John H. Page, Eric JS Lee, and Charles Croënne (Phys. and Astronomy, Univ. of MB, 301 Allen Bldg., 30A Sifton Rd., Winnipeg, MB R3T 2N2, Canada, john.page@umanitoba.ca)

Many of the interesting properties of phononic crystals are due to the existence of band gaps, which may arise from a number of different mechanisms. These include Bragg scattering as well as hybridization effects, the latter occurring when there are strong scattering resonances that hybridize with a propagating mode of the embedding medium. In this talk, we investigate the interaction between Bragg and hybridization effects on the band gap properties of 2D phononic crystals consisting of nylon rods arranged in a triangular lattice and immersed in water. The lattice constant and rod diameter were chosen to ensure that both mechanisms occur in the same frequency range. The scattering resonances of the nylon rods can be tuned in frequency by varying the temperature, enabling fine control of the overlap between hybridization and Bragg effects. The dispersion relations and transmission coefficient were measured experimentally from the phase and amplitude of transmitted ultrasonic pulses and calculated theoretically by finite element simulations. Strikingly unusual dispersion effects are observed, revealing a novel way of tuning band gap properties. These remarkable characteristics are explained by the competition between two co-existing propagation modes, reflecting strong coupling between scattering resonances of the rods.

9:05

2aPAb4. An effective Cosserat continuum model for waves in a material with microstructure. Paul E. Barbone (Mech. Eng., Boston Univ., 110 Cummington St., Boston, MA 02215, barbone@bu.edu)

The effective acoustic properties of a medium can be tailored by appropriately designing its microstructural components. The effective properties of a material may be generally defined as those that describe the limiting case where the wavelength of propagation is infinite compared to the characteristic scale of the microstructure. Generally, the limit of vanishingly small microstructural scale in an elastic medium results in an effective medium that is again elastic. In this contribution, we give an example for which the above limit results not in an effective elastic medium, but rather, an effective Cosserat medium. We briefly review the properties of Cosserat continua, provide a basic derivation of their fundamental equations of motion, and interpret the continuum properties in terms of the microstructural elements.

9:20

2aPAb5. Nonlinearity parameters B/A and C/A and wave equation for heterogeneous media containing negative stiffness 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)

This work considers nonlinear propagation in a medium consisting of a low volume fraction of metamaterial inclusions dispersed in a fluid-like material. The metamaterial inclusions of interest are assumed to possess non-monotonic stress-strain constitutive relations, which results in regimes of negative stiffness. For modeling purposes, the constitutive relation for these inclusions is approximated with an expansion to third order in volume strain with coefficients that can be tuned with the geometry of the metamaterial structure and ambient pressure. A far-reaching goal of this research is to model the hysteretic response of the heterogeneous medium resulting from metamaterial inclusion snapping events and the associated effect on acoustic disturbances that cycle through regimes of both positive and negative stiffness. As an initial step, results are presented here for small but finite-amplitude disturbances limited to local regions of the constitutive relation. For this case, the quadratic and cubic nonlinearity parameters B/A and C/A, respectively, as traditionally defined for fluids are obtained. An evolution equation with both quadratic and cubic nonlinearity is also obtained. Numerical solutions of the evolution equation illustrate nonlinear waveform distortion as a function of the volume fraction and constitutive behavior of the inclusions. [Work supported by ARL:UT McKinney Fellowship in Acoustics.]

9:35

2aPAb6. Two-dimensional broadband acoustic black hole for underwater applications. Christina J. Naify (National Res. Council PostDC Naval Res. Lab Code 7160, 4555 Overlook Ave SW, Washington, DC 20375, christina.naify.ctr@nrl.navy.mil), Theodore P. Martin (Acoust., Naval Res. Lab Code 7160, Washington, DC), Christopher N. Layman (National Res. Council PostDC Naval Res. Lab Code 7160, Washington, DC), David C. Calvo, and Gregory J. Orris (Acoust., Naval Res. Lab Code 7160, Washington, DC)

Transformation acoustics using sub-wavelength elements to obtain homogenized properties has been utilized increasingly in recent years to manipulate the propagation of acoustic waves. Gradient index (GRIN) lenses are designed by varying local properties of the elements through a given geometry for a variety of devices including focusing lenses and black holes. This study presents a cylindrical, two-dimensional acoustic black hole for underwater applications. The black hole designed focuses acoustic energy to the center of the cylinder using a radially decreasing sound speed profile. An absorbing core then prevents scattering of the acoustic waves into the surrounding fluid. Transformation acoustics was used to design the index gradient through the black hole structure and multiple scattering theory (MST) was used to predict the scattering profile over a broadband range of frequencies. The black hole was constructed of concentric rings of silicone rubber cylinders, which behave as effective fluids in a multiple scattering configuration. Experimental realization of the black hole structure was fabricated and analyzed with measured pressure intensity agreeing with predicted results. [Work sponsored by the Office of Naval Research.]

9:50–10:15 Break

10:15

2aPAb7. A negative stiffness metamaterial inclusion via small-scale curved structural elements. Timothy Klatt and Michael R. Haberman (Appl. Res. Labs. and Dept. of Mech. Eng., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, haberman@arlut.utexas.edu)

This work presents a means to produce negative stiffness metamaterial units without employing inherent nonlinear material behavior or dynamic effects, but rather through large deformations of tailored microscale geometry. Specifically, the work of Qiu *et al*. [J. Microelectromech. Syst. **13**, 137–146 (2004)] is generalized to create metamaterial units displaying non-monotonic stress-strain relationships with regions of negative stiffness behavior. These cells are designed to be embedded into a continuous matrix material and negative stiffness is elicited through pre-strain of the metamaterial unit cells brought about by changes in ambient pressure. This work presents a nonlinear hierarchical multiscale material model to estimate the macroscopic stiffness and loss of a composite material containing these pre-strained microscale structured inclusions. The multiscale model consists of two scale transition models: (i) an energy-based nonlinear finite element (FE) method to determine the anisotropic tangent moduli of the inclusion, and (ii) an analytical micromechanical model to determine the effective stiffness and loss tensors due to small perturbations about the local strain state of the metamaterial inclusions. Models and results for various volume fractions are presented and discussed.

10:30

2aPAb8. Negative refraction and focusing of acoustic waves using a foam-like metallic structure. Andrew Norris (Mech. and Aerosp. Eng., Rutgers Univ., 98 Brett Rd., Piscataway, NJ 08854, norris@rutgers.edu), Jérôme O. Vasseur, Gérard Haw (Institut d'Electronique, de Microelectronique et de Nanotechnologie, UMR 8520 CNRS, Lille, France), Charles Croënne (Phys. and Mater. Sci., City Univ. of Hong Kong, Hong Kong, Hong Kong), Lionel Haumesser (Groupe de Recherche en Materiaux, Microelectronique, Acoustique et Nanotechnologie, Univ. François Rabelais, Tours, France), and Anne-Christine Hladky-Hennion (Institut d'Electronique, de Microelectronique et de Nanotechnologie, UMR 8520 CNRS, Lille, France)

A phononic crystal (PC) slab made of a single metallic phase is shown, theoretically and experimentally, to display perfect negative index matching and focusing capability in water. The PC slab is a centimeter scale air-filled foam-like metal structure arranged in a regular hexagonal lattice in which the acoustic energy is mediated via the metal lattice. The structure has the density and longitudinal velocity of water at low frequency. The negative index property arises from an isolated higher frequency branch of the dispersion curves corresponding to a mode that couples to incident acoustic waves in water. This band intercepts the sonic line at a frequency in the ultrasonic range, which leads to perfect index matching. The metallic structure is consequently a candidate for the negative refraction of incident longitudinal waves, and the realization of flat superlenses which can focus acoustic waves with a resolution lower than the diffraction limit.

10:45

2aPAb9. Effective acoustic response and spectral representation. Raul Esquivel-Sirvent (Instituto de Fisica, UNAM, Apdo Postal 20-364, Mexico DF 01000, Mexico, raul@fisica.unam.mx)

The problem of finding an effective or average acoustic properties of inhomogeneous materials has been treated extensively in connexion with problems of porous media, rock physics and composite materials. Recently, there has been a renewed interest due to the potential applications of acoustic metamaterials. There are different techniques for finding effective acoustic properties. These range from simple averaging techniques, to variational methods, to coherent phase approximations. There is not a unique method for finding effective properties. In the case of dielectric materials, there are at least thirteen procedures reported. In this work, we present the acoustic version of the spectral representation of effective media, first developed to find dielectric effective properties. This method has the advantage that it separates the geometric contribution from the physical property to be calculated, in our case the acoustic impedance. The method is based on a Green's function solution of the acoustic wave equation and finding the effective properties is done by calculating the poles of the so called spectral function. Furthermore, we show that any effective medium model that can be described in the spectral representation satisfies the Kramers-Kronig relations. Numerical examples and comparisons between the spectral representation and other existing procedures will be discussed.

11:00

2aPAb10. Sound attenuation in ducts using locally resonant periodic flush mounted flexible silicon aerogel patches. Maaz Farooqui, Wael Akl, and Tamer Elnady (Mech. Eng., Ain Shams Univ., ASU GARDS, Abdou basha, Abbasseya, Cairo, Cairo 11517, Egypt, moaz.farooqui@eng.asu. edu.eg)

In recent years, low frequency noise has become an important factor especially in the Aircraft, HVAC, and Automotive industries. In order to reduce this low frequency noise, noise attenuation by the classical Helmholtz resonators has size limitations due to the large wavelengths. Promising noise reductions, with flush mounted Silicon Aerogel patches, can be obtained implementing attenuation due to local resonance and that too without any size constraints. The objective of the current paper is to introduce locally resonant Silicon Aerogel patches flush mounted to an acoustic duct walls aiming at creating frequency stop bands at the low frequency zone (below 500 Hz). Green's Function is used under the framework of interface response theory to predict the degree of attenuation of the local resonant patches. Realistic techniques for expanding the stop bandwidth have been introduced and difference between the Bragg scattering and the locally resonant mechanism was demonstrated using mathematical models. The effect of the arrays of patches on the effective dynamic density and bulk modulus has also been investigated. It is also shown that the numbers and periodicity of these local resonators also plays role in determining the depth and width of the acoustic band gap.

TUESDAY MORNING, 3 DECEMBER 2013 — POWELL, 9:00 A.M. TO 11:45 A.M.

Session 2aSA

Structural Acoustics and Vibration: Vibration from Aeroacoustic Loads

Micah R. Shepherd, Cochair
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Matthew D. Shaw, Cochair
Acoust., Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802

Chair's Introduction—9:00

Invited Papers

9:05

2aSA1. A history of random vibration. Thomas L. Paez (Thomas Paez Consulting, 185 Valley View Dr., Sedona, AZ 86336, tlpaez4444@gmail.com)

Natural, random, dynamic environments are ubiquitous, and humans have observed them for millennia. Some random dynamic environments are earthquake ground motions, winds, ocean waves, rough roads, and acoustic pressure environments arising from thunder. Before people could easily conceptualize harmonic motions, they observed random, oscillatory environments. Today, random vibration can be mathematically modeled as the motion of a mechanical system excited by a random input. The mathematical theory of random vibration is essential to the realistic modeling of structural dynamic systems. This paper summarizes the work of some key contributors to the theory of random vibration from the time of Rayleigh and Einstein to the present. Among many other things, we describe the works of (1) Rayleigh, who, in the late nineteenth century introduced ideas about the representation of random signals as the superposition of random harmonics; (2) Schuster, who, in the late nineteenth and early twentieth centuries described, without taking its limit, the spectral density; (3) Einstein who, in 1905, wrote the first actual paper on the theory of random vibration (Brownian movement); and (4) Wiener who, in 1930, formally defined the spectral density. Several graphic examples are included.

9:25

2aSA2. Uncertainty bounds estimation when simulating vibration response under aeroacoustic loading. Paul Bremner (AeroHydroPLUS, 2311 Via Aprilia, Del Mar, CA 92014, pbremner@aerohydroplus.com)

The simulation of linear vibration response under aeroacoustic loading, involves the aeroacoustic loading spectrum and its spatial distribution, and the modal dynamics of the loaded structure. In most practical applications, this problem can only be addressed statistically because the aeroacoustic loading is temporally random and only partially space-correlated, requiring a time-averaged and space-averaged cross-spectrum description of both the loading and the structure vibration response. However, it is equally common that the spectrum and spatial correlation of the loading is uncertain and—particularly at higher frequencies or higher modal densities—even the modal dynamics of the structure are uncertain (e.g., due to uncertainties in the as-built boundary conditions). This paper will explore what is required to predict the statistical variance and the maximum expected vibration response—in addition to the space-time average cross spectrum—given uncertainty in the modeling parameters.

9:45

2aSA3. Measurement of high amplitude relief valve noise during a full scale blowdown. Neal Evans (Southwest Res. Inst., Div 18, B77, 6220 Culebra Rd., San Antonio, TX 78238, neal.evans@swri.org)

Dynamic pressure fluctuations inside a pipe were measured downstream of a pressure relief valve during a full scale blowdown test. Nitrogen gas flowed with a maximum rate of 33.5 kg/s through a 3 in.x4 in. relief valve generating a peak dynamic pressure level greater than 650 kPa and sustained levels of over 450 kPa (peak). An accurate estimate of valve-generated noise is necessary when predicting radiated noise and acoustic induced vibration, which has been shown to cause fatigue failures at welded discontinuities in piping systems downstream of high pressure drop devices. These failures can be hazardous and costly, particularly when the process involves hydrocarbons such as natural gas. The measured level is compared to existing noise calculation techniques which appear to under-predict the generated noise.

10:05–10:20 Break

10:20

2aSA4. Sonic fatigue coherence models. Robert D. Blevins (United Technologies, 3818 Pringle St., San Diego, CA 92103, rdblevins@AOL.com)

Computational and experimental based models for coherence in acoustic and turbulent pressure fields are developed for application to sonic fatigue of large structures—structures with dimensions greater than a wavelength or of the turbulence length scale. Conservative analytical methods for sonic fatigue of structures exposed to coherent pressure fields are first developed. Then models are introduced to account for finite coherence of turbulent and non-coherent pressure fields that arise in turbojet engines. The computational results are compared with experimental data.

10:40

2aSA5. Reduced order modeling for the skin panels of hypersonic vehicles and nonlinear normal modes. Matthew S. Allen (Eng. Phys./Eng. Mech., Univ. of Wisconsin-Madison, 535 Eng. Res. Bldg., 1500 Eng. Dr., Madison, WI 53706-1609, msallen@engr.wisc.edu)

The skin panels on concept hypersonic (Mach > 5) vehicles are subjected to intense acoustic and thermal loading. As a result, nonlinear structural dynamic models are needed to predict their response with the required level of accuracy. Vehicles such as this carry a large amount of fuel and so the structure must be very light if the overall vehicle is to meet its performance requirements. This talk will discuss the reduced order modeling frameworks that are being used to create models for these types of structures and discuss how nonlinear normal modes are being used to understand how the structure's response changes with the loading amplitude. Nonlinear modes are also being used to evaluate the reduced order models in order to predict the frequency bandwidth and the range of forcing amplitude over which they will be valid. This approach allows the analyst to develop a considerable level of confidence in the reduced order model without having to compute time responses of the full nonlinear finite element model; time response simulations are far too expensive to be used in practice on the structures of interest.

Contributed Papers

11:00

2aSA6. Low noise blower fan for heating and cooling applications. Yi Liu (Ingersoll Rand, 618 Davis CT, Indianapolis, IN 46234, yiliu1975@gmail.com) and Percy Wang (Ingersoll Rand, Tyler, Texas)

A new low noise fan called foam blower fan is presented here for low noise heating, ventilation, and air conditioning (HVAC) applications. Traditional HVAC blower fans are made of discrete metal or plastic fan blades, which generate blade tonal noise when each blade interacts with blower cutoff or other housing structures. It is proposed here that foam wedge (porous media) is inserted between blades at strategic locations, to reduce the tonal noise generation. It is analytically and experimentally demonstrated in this paper that the proposed new foam fan can reduce discrete tonal noise at various operating speeds.

11:15

2aSA7. Industry tubing characterization method for fixed speed applications. Yi Liu (Ingersoll Rand, Indianapolis, IN) and Yufeng Shen (Ingersoll Rand, 9/11F Tower B City Ctr. of Shanghai, No. 100 Zun Yi Rd., Shanghai, China, MichaelYuFeng.Shen@trane.com)

Efforts for structural tubing characterization are presented here for HVAC industry fixed speed applications. Structural tubing is a type of metal connector typically with hollow circular or rectangular cross-sections. Copper tubing is widely used in HVAC industry to transfer refrigerant between compressors and condensers. However, when the natural frequencies of tubing system are at the running speeds of the compressors; design life of the product is reduced. Therefore, tubing configuration structural change is needed to shift these problem frequencies. Traditionally, industry has been using finite element method and vibration tests to identify the critical geometric parameter for frequency shift. New efforts are been proposed here to present designer a sensitivity study tool to identify these critical parameters quickly, therefore, shorten the design cycle time.

11:30

2aSA8. Modification of the spectral response of a pipe resonator using a subordinate array of coupled Helmholtz resonators. Aldo A. Glean, Joseph F. Vignola, John A. Judge, and Teresa J. Ryan (Mech. Eng., Catholic Univ. of America, 620 Michigan Ave. NE, Washington, DC 20064, 10glean@cardinalmail.cua.edu)

The dynamic response of a resonant system can be manipulated by attaching a set of substantially smaller resonators with a prescribed distribution of properties. These smaller resonators are collectively referred to as a subordinate array. This work describes an experimental demonstration of such a manipulation, in which we alter one resonant peak of the primary system to create a flat bandpass response, while leaving the other resonances unaffected. In this instance, the primary resonant system is a pipe with one end closed and the other open, and the subordinate array consists of a set of small Helmholtz resonators. We demonstrate the desired modification of the system response at the third resonant frequency without significant change in adjacent resonances. Specifically, a single system resonance of the primary resonator shows a peak response when no subordinates are attached and exhibits a band pass response when the subordinates are attached. Sensitivity of system response to errors in design parameters is discussed and experimental results are compared to theoretical predictions.

TUESDAY MORNING, 3 DECEMBER 2013 — PLAZA B, 8:30 A.M. TO 12:00 NOON

Session 2aSC

Speech Communication: Speech Perception I

Grant L. McGuire, Chair
Linguistics, UC Santa Cruz, 1156 High St., Stevenson Academic Services, Santa Cruz, CA 95064

Contributed Papers

8:30

2aSC1. Perception of acoustically similar vowels from English and Hebrew. Richard J. Morris (Commun. Sci. and Disord., Florida State Univ., 201 West Bloxham Rd., 612 Warren Bldg., Tallahassee, FL 32306-1200, richard.morris@cci.fsu.edu) and Yonit A. Shames (Audiol., Helix Hearing Care, Orlando, FL)

The formant theory suggests that vowels are differentiated perceptually based on the presence of concentrated bands of harmonic energy in the vowel's acoustical signal. Vowels from different languages often contain formants with similar configurations, but it is unclear what effect cross-language variations have on the perception of the vowels. This study was completed to find when English-speaking listeners are able to differentiate between vowels spoken in two languages, General American English and Modern Hebrew. Natural vowels in kVp syllables were recorded and acoustically normalized using the Bark scale. The English and Hebrew vowels were paired in test groups based on the normalized formant values and the fundamental frequencies of the speakers. Listeners designated each pair as "same" or "different." It was hypothesized that listeners were more likely to differentiate between the vowels in English-Hebrew pairs than in same-language pairs, and that they were more likely to differentiate between vowels with higher Bark scale differences. However, Bark scale differences did not always match perceived differences. Listeners demonstrated the least difficulty discriminating between different-language pairs containing /e/ and /o/ (60–90% accuracy), more difficulty for English /ɑ̃/ and Hebrew /a/ (40–80% accuracy), and the most difficulty for /i/ and /u/ (10–60% accuracy).

8:45

2aSC2. Acoustic cue weighting across modalities in a non-native sound contrast. Jessamyn L. Schertz (Dept. of Linguist, Univ. of Arizona, Douglass 200, Tucson, AZ 85721, jschertz@email.arizona.edu), Andrew J. Lotto (Dept. of Speech, Lang., and Hearing Sci., Univ. of Arizona, Tucson, AZ), Natasha Warner (Dept. of Linguist, Univ. of Arizona, Tucson, AZ), and Taehong Cho (Hanyang Phonet. and PsychoLinguist Lab, Dept. of English Lang. and Lit., Hanyang Univ., Seoul, South Korea)

This work investigates the production and perception of the English stop voicing contrast on multiple acoustic dimensions by native speakers of Seoul Korean. Subjects completed a production task as well as a forced-choice identification task on stimuli varying on three acoustic dimensions (aspiration duration, pitch, and closure duration) in both English and Korean. On average, native Korean listeners relied more on pitch than on aspiration duration to categorize the English stop voicing contrast (which native English listeners distinguish primarily by aspiration). However, individual categorization patterns differed considerably, with some listeners using only pitch, some using only aspiration, and most using both. In contrast, in production, aspiration duration was a better predictor of voicing category than pitch, although pitch was still a stronger predictor than previously found in native English productions. The heavier reliance on pitch by Korean listeners may be attributable to the greater importance of pitch in their native stop contrast; however, there does not appear to be a consistent, straightforward mapping of English sounds onto the Korean categories. Results will be discussed in the terms of the perception-production interface on the level of individual acoustic cues, as well as the influence of native language cue weights on non-native sound contrasts.

9:00

2aSC3. Perception of stressed vs unstressed vowels: Language-specific and general patterns. Priscilla Shin, Natasha L. Warner, Maureen Hoffmann (Linguist, Univ. of Arizona, Box 210028, Dept. Ling, Univ. AZ, Tucson, AZ 85721-0028, nwarner@u.arizona.edu), James McQueen (Donders Inst., Radboud Univ. of Nijmegen, Nijmegen, Netherlands), and Anne Cutler (MARCS Inst., Univ. of Western Sydney, Penrith, NSW, Australia)

Unstressed vowels are somewhat centralized (even full vowels such as the second in "city, taco"), reducing their acoustic distinctiveness. The current work compares listeners' perception of stressed and unstressed vowels in English and Dutch. The data come from two large projects on native listeners' perception of all possible diphones (CV, VC, CC, and VV sequences, all vowels stressed and unstressed) in English and Dutch. These datasets provide information about listeners' uptake of perceptual cues over time that is comparable across the two languages. Both groups perceived unstressed vowels less accurately than stressed, but this effect was far larger for English. English listeners showed a very large stress effect for lax vowels and a moderate effect for other vowels, while the Dutch listeners showed effects that were small and largely restricted to /a/. Dutch listeners may be able to identify unstressed vowels better than English listeners because the stressed-unstressed distinction has more informational value in Dutch than in English. However, both languages showed a larger stress effect just before a following consonant. This suggests that consonantal coarticulation obscures the quality of unstressed vowels in both languages. Thus, perception of stressed vs unstressed vowels demonstrates both language specificity and cross-language commonality.

9:15

2aSC4. Phonetic and orthographic cues are weighted in speech sound perception by second language speakers: Evidence from Greek speakers of English. Anastasia Giannakopoulou (Psych., Univ. of Bedfordshire, University Square, Luton LU1 3JU, United Kingdom, anastasia.giannakopoulou@beds.ac.uk), Maria Uther (Psych., Winchester Univ., Winchester, United Kingdom), and Sari Ylinen (Psych., Univ. of Helsinki, Helsinki, Finland)

Speech-sound contrasts that have multiple phonetic cues can be particularly difficult for foreign-language learners especially if the cues are weighted differently in the foreign and native languages (e.g., Giannakopoulou *et al.*, 2013). The orthographic representation of words is suggested to also interfere with speech sound perception in way of presenting additional cues for the second language learner. In order to examine the possibility that orthographic representation of the word stimuli provides additional cues, this study explores perceptual identification with the use of pictures as visual stimuli. Greek child and adult speakers of English were studied to determine on what basis they are making perceptual identification between English vowels. One task involved the use of minimal pairs in their orthographic form (word stimuli), another task used relevant pictures that resembled the meaning of the respective words. The auditory stimuli used in both task types were identical. Performance was impaired for Greek speakers across all tasks but worst for Greek speakers for the picture stimuli task. Interestingly, child Greek speakers performed significantly worse in the picture stimuli task, even though a picture translation control task revealed high performance. These results are discussed in terms of the strategies used to acquire new languages.

9:30

2aSC5. Evidence for cognitive restoration of time-reversed speech by a language-trained chimpanzee (*Pan troglodytes*). Lisa A. Heimbauer (Psych., Penn State Univ., 442 Moore Bldg., University Park, PA 16802, lisa.heimbauer@gmail.com), Michael J. Beran (Lang. Res. Ctr., Georgia State Univ., Atlanta, GA), and Michael J. Owren (Psych., Emory Univ., Atlanta, GA)

Previously, we reported on the ability of Panzee, a language-trained chimpanzee, to identify sine-wave and noise-vocoded speech by attending to the amplitude and frequency modulations in the altered signals. Here, we report on her ability to perceive phoneme-length information in words reproduced in time-reversed form. While this manipulation preserves the amplitude of frequency components, it reverses the pattern of energy changes within each reversal window. Listeners easily recognize speech at reversal windows up to 100-ms length, but at longer reversal lengths unintelligibility begins to occur (Saberi and Perrott, 1999). The theoretical interpretation is that individual phonetic segments range from 50 to 100 ms (Crystal and House, 1988), and reversal-windows less than 100 ms provide for restoration of phoneme perception. Hypothesizing that Panzee also perceives speech based on phonemic segments, we tested her and humans with words in eight reversal forms ranging from 25 to 200 ms. Results revealed time-reversal window length significantly predicted percentage-correct word identification for Panzee and the humans. Additionally, window lengths exceeding 100 ms produced partial word intelligibility, with 50% intelligibility occurring at approximately 130 ms for both species. We therefore conclude that Panzee attends to phoneme-related cues in time-reversed speech, and hence in natural speech.

09:45–10:00 General Discussion

10:00–10:30 Break

10:30

2aSC6. Simple auditory elements induce perception of a phonetic feature. Gregory Finley (Linguist, Univ. of California, 5820 Occidental St., Oakland, CA 94608, finley@berkeley.edu)

In this presentation, I demonstrate that certain nonspeech sounds can have perceptual phonetic value. I focus on a single phonetic/articulatory feature, lip rounding, and its detection in simple auditory stimuli. Behavioral experiments show that a rounding percept is possible for two types of nonspeech. One stimulus type, which yields the more robust response, is a complex periodic source filtered by a single narrow band reminiscent of a speech formant. The resulting nonspeech varies in perceived roundedness depending on the filter's frequency, corresponding roughly with F2. The other stimulus type is a pure tone modulated upward in frequency. Preliminary results suggest that rounding can indeed be perceived on these sounds, but only with specific modulation rates within a certain frequency range. These findings indicate that minimally simple auditory objects, including pure tones and filtered bands, can be sufficient to encode phonetic information. Additionally, these two types of cues diverge in their ability to trigger this percept: a filtered band works as a static spectral cue, whereas a pure tone requires spectrotemporal modulation. This observation is consistent with findings that there are auditory STRFs specifically sensitive to modulation and the theoretical perspective that auditory organization directly predicts the processing of speech.

10:45

2aSC7. The effect of speaking rate, vowel context, and speaker intelligibility on the perception of consonant vowel consonants in noise. Anirudh Raju and Abeer Alwan (Elec. Eng., Univ. of California, Los Angeles, 550 Veteran Ave., Apt. #102, Los Angeles, CA 90024, anirudh90@ucla.edu)

In this paper, we perform pilot experiments to evaluate the feasibility of a model to predict human recognition of speech sounds in the presence of noise at different speaking rates. CVC stimuli comprising a phonetically balanced set of 13 consonants and 3 vowels (/i/, /a/, /u/) were recorded in a sound proof booth by two talkers at two different speaking rates (fast and slow). Noisy stimuli were generated by adding babble noise at different levels to the quiet recordings. These stimuli were used to conduct perceptual experiments in which listeners were asked to listen and repeat back the CVC phrases presented in babble noise under 3 SNR conditions and both speaking rates. The data were transcribed by two trained linguists. Consonant confusion matrices were generated from these data and were analyzed by noise level, speaker, center vowel, and speaking rate. With the exception of /CuC/ stimuli, speaking rate had the most pronounced effect on perception with slow speech being more intelligible than fast speech in noise. /CaC/ stimuli were, on average, more robust than other stimuli in all conditions and one talker was significantly more intelligible than the other. A detailed analysis of the results will be presented. [Work supported in part by the NSF.]

11:00

2aSC8. Perceptual compensation with familiar and unfamiliar rounding coarticulation. Keith Johnson, Shinae Kang, and Emily Cibelli (Dept. of Linguist, UC Berkeley, 1203 Dwinelle Hall, Berkeley, CA 94720-2650, keithjohnson@berkeley.edu)

We compared the integration of three kinds of contextual information in the perception of the fricatives [s] and [ʃ]. We asked American English listeners to identify sounds on an [s] to [ʃ] continuum and manipulated (1) the vowel context of the fricative ([Ce], [Co], [Cœ]), (2) the original fricative of the CV ([s] vs [ʃ]), and (3) the modality of the stimulus (audio-only, AV). There was a large compensation for coarticulation effect on perception—subjects responded with "s" more often when the following vowel was round. Interestingly, and perhaps significantly, perceptual compensation was not as great with the less familiar vowel [œ] even when listeners saw the face. Measurements of lip rounding in these stimuli show that [o] and [œ] have about the same degree and type of rounding over the CV. In a second experiment, we measured reaction time to audio-visual mismatches in these stimuli (again in a fricative identification task). We found that mismatches of audio and video consonant information slowed reaction time, and that vowel mismatches did as well. However, mismatch between [o] and [œ] did not slow reaction time. These data suggest that linguistic experience and stimulus properties affect perception.

11:15

2aSC9. Visual cue salience depends on place of articulation and syllabic context. Shinae Kang and Keith Johnson (Linguist, UC Berkeley, 1203 Dwinelle Hall, UC Berkeley, Berkeley, CA 94720-2650, sakang2@berkeley.edu)

This study is on audio-visual perceptual intelligibility of consonants in intervocalic clusters (VC1C2V). Previous studies have yielded inconsistent findings on perceptual salience of different stop consonants and very few have tested salience in clusters. Consequently, it has been unclear as to whether greater or less perceptual salience leads to greater degree of place assimilation. In Korean, labials are often produced with more gestural overlap than velars in C1. I tested whether labials are perceptually more or less salient in both audio and audio-visual conditions. VC and CV syllables spoken by both English and Korean speakers were first embedded in noise and spliced together for non-overlapping VCCV sequences. Korean listeners identified the two consonants in either audio or AV presentations. A confusion matrix analysis for each stop consonant shows that in C1 there is asymmetric improvement with the addition of videos for labial consonants only, while in C2 this asymmetry was not found. The result suggests that listeners make differential use of visual cues depending on place of articulation and syllabic context. Also, the result supports the talker enhancement view of sound change, which assumes that talkers are aware of perceptual salience and enhance (with less gestural overlap) the weak contrast.

11:30

2aSC10. Psychophysiological indices of effortful listening in younger and older adults. Alexander L. Francis (Speech, Lang. and Hearing Sci., Purdue Univ., SLHS, Heavilon Hall, 500 Oval Dr., West Lafayette, IN 47907, francisa@purdue.edu), Megan K. MacPherson (School of Commun. Sci. and Disord., Florida State Univ., Talahassee, FL), Bharath Chandrasekaran (Dept. of Commun. Sci. and Disord., The Univ. of Texas at Austin, Austin, TX), Ann M. Alvar (Speech, Lang. and Hearing Sci., Purdue Univ., West Lafayette, IN), and Fernando Llanos (School of Lang. and Cultures, Purdue Univ., West Lafayette, IN)

Older adults often have difficulty understanding speech in background noise, and this difficulty may be associated with cognitive processing demand. According to the effortfulness hypothesis, even sub-clinical age-related changes in hearing may increase cognitive demand for speech understanding, making listening in noise more effortful for older adults even when recognition performance is comparable to that of younger listeners. Separating speech from background noise requires both segregating target from masking signals and selectively attending to the target while ignoring maskers. While both segregation and selection may demand cognitive resources, it is not known whether both mechanisms interact with age to the same degree. To address this question, younger and older adults listened to and repeated sentences presented in quiet and under conditions that put relatively more emphasis on segregation (energetic masking using speech-shaped broad-band noise) or selection (informational masking using two-talker babble) or are cognitively demanding without masking (synthetic speech). Masked stimuli were equally intelligible based on prior research, so differences in listening effort may be attributed to age and/or masker type. Listening effort was measured behaviorally via traditional rating scales (NASA TLX), and psychophysiologically in terms of autonomic nervous system responses (heart rate, pulse period, and amplitude, and skin conductance).

11:45–12:00 General Discussion

TUESDAY MORNING, 3 DECEMBER 2013 — CONTINENTAL 4, 8:30 A.M. TO 11:30 A.M.

Session 2aSP

Signal Processing in Acoustics, Underwater Acoustics, Structural Acoustics and Vibration, and Annual Buacoustics: Time Reversal for Localization and Focusing of Sound

Brian E. Anderson, Chair
Geophys. Group, Los Alamos National Lab., M.S. D443, Los Alamos, NM 87545

Invited Papers

8:30

2aSP1. Time-reversal-based underwater acoustic communication. Hee-Chun Song (Scripps Inst. of Oceanogr., 9500 Gilman Dr., La Jolla, CA 92093-0238, hcsong@mpl.ucsd.edu)

Time reversal (TR) exploits spatial diversity to achieve spatial and temporal focusing in complex environments. Over the last decade its concept has been applied successfully to underwater acoustic communications in time-varying multipath environments with robustness and computational simplicity, as an alternative to conventional adaptive multichannel equalization. Temporal focusing (pulse compression) mitigates the intersymbol interference (ISI) and subsequent channel equalization removes the residual ISI, thus providing nearly optimal performance in theory. The spatial focusing capability facilitates multi-user communications without an explicit use of time, frequency, or code division, while an adaptive time reversal approach can further reduce the crosstalk among users. TR communications can be easily extended to time-varying channels using a block-based approach with periodic channel updates. This talk will present an overview of TR communications and recent advances including bidirectional equalization, single- versus multi-carrier approach, and communications with autonomous vehicles such as an AUV or Glider.

8:50

2aSP2. Time reversal methods for high frequency multiple-input/multiple-output underwater acoustic communication. Aijun Song and Mohsen Badiey (School of Marine Sci. and Policy, Univ. of Delaware, 114C Robinson Hall,Newark, DE 19716, ajsong@udel.edu)

As a sound focusing technique, time reversal has been utilized in underwater acoustic communication since the 1990s. Here, we particularly study the usage of time reversal methods to separate sounds from different sound sources for high frequency transmissions (greater than 10 kHz), in addition to sound focusing. The separation of sounds creates multiple parallel acoustic links between multiple transducers and a receiver array, thus, providing increased data rates between the source and receiver for communication purposes. It is referred to as multiple-input/multiple-output (MIMO) acoustic communication. Multiple at-sea experiments have been conducted to demonstrate high frequency MIMO signaling and their communication performance. A number of signal processing techniques have been developed in the time reversal framework to enhance the sound focusing and separation that are needed in high frequency acoustic MIMO communication. For example, multi-stage interference cancellation methods have been devised to suppress the co-channel interference and to ensure the sound separation. In this talk, we will show their effectiveness in the presence of water column variability based on our experimental data. An acoustic communication channel model will also be used to explain the impact of environmental variability.

9:10

2aSP3. Time reversal communication over doubly spread channels. Wen-Jun Zeng (Dept. of Electron. Eng., City Univ. of Hong Kong, Kowloon, Hong Kong 100084, Hong Kong, cengwj06@mails.tsinghua.edu.cn) and Xue Jiang (Dept. of Elec. and Comput. Eng., McMaster Univ., Hamilton, ON, Canada)

Conventional time reversal can mitigate multipath delay dispersion by temporal focusing. But it is not applicable to time-varying channels with a Doppler spread. Although recently time reversal communication has been adapted to time-variant channels, the modified technique requires frequent channel updates to track channel variations and cannot handle large Doppler spread, which means that it cannot achieve frequency focusing. In this paper, two time reversal receivers for underwater acoustic communications over doubly spread channels are proposed. The proposed approach, which can be interpreted as time-frequency channel matching, is based on the channel spreading function rather than impulse response adopted by the existing techniques; this leads to much less frequent channel updates. Unlike existing methods that only correct a single Doppler shift, the proposed approach uses a rake-like structure to compensate for multiple Doppler shifts and hence can eliminate severe Doppler spread induced by temporal channel variations. Simulation results verify the effectiveness of the proposed approach, indicating that it can simultaneously counteract delay and Doppler spreads, achieving both temporal and frequency focusing.

9:30

2aSP4. Advances in biomedical applications of time reversal acoustic focusing of ultrasound. Alexander Sutin (, Artann Labs., 711 Hudson St., Hoboken, NJ 07030, asutin@stevens.edu), Yegor Sinelnikov, and Armen Sarvazyan (, Artann Labs., West Trenton, NJ)

Time reversal acoustics (TRA) is one of the most efficient methods of ultrasound focusing in heterogeneous composite biological media, especially inside reverberating cavities, such as the skull. In this talk, we will overview several recently developed therapeutic applications of TRA focusing of ultrasound including enhanced drug delivery to brain tumors, generation of focal regions of complex shape tailored to the geometry of the target lesion and TRA dynamic focusing, that allows to maintain the constant acoustic intensity in the focus regardless the variation of acoustical parameters of the media. We are currently developing several new medical applications of TRA, such as remote charging of batteries in internal implants and leadless energizing deep brain stimulators, which are based on the possibility to remotely generate an electrical signal in tissue using TRA principles. These applications employ a TRA focusing system with wireless electromagnetic feedback from the tiny implanted piezotransducer acting as a beacon. The acoustic energy is accurately focused at the piezotransducer generating required electrical signal while providing minimum exposure of surrounding tissues to ultrasound energy. Possibility of remote generation of electrical signals in tissue with amplitudes reaching tens of volts was demonstrated. [Work supported by NIH R21 CA164935-01.]

9:50

2aSP5. Invariants of the time reversal operator and characterization of solid media: An overview. Claire Prada (Institut Langevin, CNRS ESPCI, 1 rue Jussieu, Paris 75005, France, claire.prada-julia@espci.fr)

The invariants of the time reversal operators for solid media have been the object of several studies in the past 15 years in the context of multi-element array imaging. These invariants are obtained from the decomposition of the array response matrix. Their analysis was applied to the detection of flaws using bulk waves, or Rayleigh and Lamb guided more. It was also applied to the characterization of shells using the radiation of circumferential guided modes. A review of the different solutions that have been proposed to improve this analysis will be given. Then, selected experimental examples of time reversal invariants will be discussed.

10:10–10:25 Break

10:25

2aSP6. Evaluation of concrete carbonation using time reversal and nonlinear acoustics. Pierre-yves Le Bas (Geophys. group, EES-17, Los Alamos National Lab., MS D446, Los Alamos, NM 87545, pylb@lanl.gov), Cedric Payan (Aix Marseille Univ., LMA CNRS UPR 7051, Marseille, France), Timothy J. Ulrich (Geophys. group, EES-17, Los Alamos National Lab., Los Alamos, NM), and Vincent Garnier (Aix Marseille Univ., LMA CNRS UPR 7051, Aix, France)

Carbonation of concrete can lead to corrosion of rebar and degradation of structures, including nuclear plant walls and reactors and casks for storage of nuclear waste. Carbonation has been shown to decrease the natural nonlinearity of concrete. Using the time reversal nonlinear elastic diagnostic (TREND) at different frequencies allows for probing at different depths (half a wavelength). By looking at

the evolution of the nonlinear response with frequency we can estimate the depth of carbonation. This study will present experimental results aimed at determining the depth of carbonation on medium scale samples ($25\times50\times10$ cm and $75\times100\times20$ cm). The samples have been prepared using a protocol known to induce carbonation down to a controlled depth. Several samples are available with carbonation at 0, 1, 2, and 4 cm. We will present the results of the experiments analyzed using several techniques to quantify nonlinearity, namely, pulse inversion and Scaled Subtraction Method (SSM).

Contributed Papers

10:45

2aSP7. Applying an old appealing idea to modern seismology: Time reversal to characterize earthquakes. Carene Larmat, Robert A. Guyer, Paul A. Johnson (EES-17, Los Alamos National Lab., P.O. Box 1663, M.S. D452, Los Alamos, NM 87545, carene@lanl.gov), and Jean-Paul Montagner (Seismology Lab., Institut de Physique du Globe de Paris, Paris, Paris Cedex 05, France)

Wave physics is one domain where reversing time is possible and has led to interesting applications. In acoustics, Parvulescu and Clay (1965) used what they termed a "matched signal technique" to beat multi-reverberation in the shallow sea. In seismology, McMechan (1982) demonstrated the feasibility of what he termed "wavefield extrapolation" to locate seismic sources. Since then, other concepts and applications, all related to time-reversal, have often been proved to be successful where other techniques have failed. This success is due to the inherent ability of time-reversal to function well in complex propagation media as well as the remarkable robustness of the method with sparse receiver coverage. The key aspect of time-reversal for future applications in seismology is that it relies on no a priori assumption about the source. This allows automatic location of earthquakes and the study of seismic events for which the assumption of point source breaks down. This is the case of big earthquakes (Mw >8) for which the rupture length and source duration extend to hundreds of kilometers and several tens of seconds. We will show an application to the 2011 Japan earthquake, to icequakes related to glaciers motions in Greenland and to seismic tremor with no clear onset.

11:00

2aSP8. A high amplitude non-contact acoustic source: from a proof of concept to the understanding of the mechanisms involved. Marcel C. Remillieux, Brian E. Anderson, Timothy J. Ulrich, and Pierre-Yves Le Bas (Geophys. Group (EES-17), Los Alamos National Lab., Los Alamos, NM 87545, mcr1@lanl.gov)

Recently, the preliminary design of a high-amplitude, non-contact acoustic source was proposed for nondestructive testing applications [Le Bas *et al*., J. Acoust. Soc. Am. **134**, EL52 (2013)]. The design is based on the principle of time reversal, a process to focus energy at a point in space. In the present work, the main physical mechanisms involved in the operation of this device are examined numerically using the finite-element (FE) method. First, a three-dimensional FE model of the device is validated in the frequency domain against experimental data. Subsequently, two-dimensional transient simulations are used to conduct a parametric study on the effect of wall thickness and transducer density on the efficiency of the device. Last, a time reversal numerical experiment is presented. Results from this study can be used to design a more efficient non-contact source. [Work supported by Institutional Support (LDRD) at the Los Alamos National Laboratory.]

11:15

2aSP9. Imaging the orientation of stress corrosion cracking using the three component time reversed elastic nonlinearity diagnostic. Brent O. Reichman, Brian E. Anderson, Timothy J. Ulrich, and Pierre-Yves Le Bas (Geophys. Group (EES-17), Los Alamos National Lab., M.S. D446, Los Alamos, NM 87545, bea@lanl.gov)

Linear acoustic techniques often are not able to locate closed cracks as the acoustic waves pass right through them. Fortunately, nonlinear acoustic techniques may be used in conjunction with time reversal techniques to not only locate a closed crack but also to image it. This presentation will discuss how cracks may be imaged using the three component time reversed nonlinearity diagnostic (3D-TREND). Specifically, the orientation of individual cracks in a 304L stainless steel plate, which resulted from controlled stress corrosion cracking (SCC) experiments, will be presented along with other imaging information about the crack. 3D-TREND is used to create an individual time reversal focus of energy at each inspection point of interest. The use of different frequencies and different excitation signals has been used in an attempt to extract more information about the crack. [This work was supported by the U.S. Dept. of Energy, Fuel Cycle R&D, Used Fuel Disposition (Storage) Campaign].

Session 2aUW

Underwater Acoustics: Sound Propagation Through and Scattering by Internal Waves, Spice, and Finestructure in Shallow Water I: Past, Present, and Future

Steven I. Finette, Cochair
Acoust. Div., Naval Res. Lab., 4555 Overlook Ave. SW, Washington, DC 20375-5320

James Lynch, Cochair
Woods Hole Oceanogr., M.S. # 11, Bigelow 203, Woods Hole, MA 02543

Chair's Introduction—8:55

Invited Papers

9:00

2aUW1. Future shallow water low frequency (100–1000 Hz) acoustic signal propagation physics studies. Marshall H. Orr (The Acoust. Div., The Naval Res. Lab., 4555 Overlook Ave. SW, Washington, DC 20375, rubyspiral@gmail.com)

The measurement and prediction of the phase and amplitude properties of acoustic signals (1 to 1000 Hz) propagating in the dynamic shallow water (30 to 500 m water depths) environment has been a focus of the underwater acoustic research community for nearly 20 years. The majority of the ocean studies have occurred during late spring to summertime oceanic conditions when the sound speed variability was influenced by nonlinear mode 1 internal wavefields. Sound speed variability during the summer-to-winter and winter-to-summer water column transition periods will cause variability in the phase coherent properties of acoustic signals. Few experimental studies explicitly focused on quantifying the variability of phase coherent properties of acoustic signals during these periods have been performed. Illustration of the types of fluid processes that will perturb the sound speed structure during the fall-to-winter transition periods will be presented. Included will be mode 1 and mode 2 internal wave perturbation of the sound speed field as well as the locations and spatial scales of sound speed variability caused by interleaving water masses of varying temperature and salinity (Spice). Fall-to-winter transition sound speed fields will be contrasted with summer time sound speed conditions. [Work supported by the Office of Naval Research.]

9:20

2aUW2. Statistics of internal waves measured during the Shallow Water 2006 experiment. Mohsen Badiey, Lin Wan (College of Earth, Ocean, and Environ., Univ. of Delaware, 261 S. College Ave., Robinson Hall, Newark, DE 19716, badiey@udel.edu), and James F. Lynch (Woods Hole Oceanographic Inst., Woods Hole, MA)

During the Shallow Water Acoustic Experiment 2006 (SW06), detailed measurements of the time-varying ocean environment were made while simultaneously acoustic signals were transmitted between various source and receiver pairs. The time-varying environment induced by internal waves (IW) was recorded by an array of moored thermistor chains, as well as by the attending research vessels. Using a mapping technique described by Badiey *et al*. [J. Acoust. Soc. Am. EL. **134** (2013)], the three-dimensional (3D) temperature field for over a month of IW events was reconstructed. The results of this mapping are used for the statistical analysis of the IW parameters, such as the IW propagation speed, direction, amplitude, coherence length, etc. This paper provides a summary of these results and also examines the implications of the detailed statistics as regards to the acoustic field. The results in this paper could be used as a database for studying the IW generation, propagation, and its impact on the 3D acoustic propagation in waveguides. [Work supported by ONR322OA.]

Contributed Papers

9:40

2aUW3. A perspective of modeling internal wave/acoustic wave interactions at the Naval Research Laboratory 1992-2012. Steven I. Finette (Acoust. Div., Naval Res. Lab., 4555 Overlook Ave. SW, Washington, DC 20375-5320, steven.finette@nrl.navy.mil)

Interest in modeling internal gravity waves in littoral regions was stimulated by the discovery of anomalous transmission loss in the Yellow Sea, predicted by Zhou and Rogers in 1991 to be caused by solitary wave propagation along the acoustic transmission path. Over the past 20 years, we have been modeling acoustic wave/solitary internal wave interactions and an overview and perspective of some of this work, illustrated by examples, will be presented. The relationship between internal waves and the scintillation index, horizontal refraction of acoustic energy, oceanographic waveguide focusing, mode coupling between acoustic and internal wave modes, and the effect of internal waves on acoustic field uncertainty are discussed. Some thoughts on the future of 4-D simulation of acoustic/internal wave interactions and modeling in the context of incomplete environmental knowledge are considered. [Work supported by the Office of Naval Research.]

9:55

2aUW4. Acoustic normal mode fluctuations due to internal waves in the Shallow Water 2006 experiment. Lin Wan, Mohsen Badiey (College of Earth, Ocean, and Environ., Univ. of Delaware, 003 Robinson Hall, Newark, DE 19716, wan@udel.edu), and James F. Lynch (Woods Hole Oceanographic Inst., Woods Hole, MA)

The Shallow Water 2006 (SW06) experiment was a large-scale acoustic experiment conducted on the New Jersey continental shelf in the summer of 2006 with substantial investment from the Office of Naval Research. The main goal of this experiment was to find a detailed understanding of the waveguide during the propagation of broadband acoustic signals in the presence of internal waves (IW). One month of IW events recorded during this experiment has been reconstructed with the aids of densely deployed thermistor strings and ship-borne X-Band radars. The comprehensive IW measurements enabled the study of the correlation between the IW-induced time-varying environment and the acoustic normal mode fluctuations. During this experiment, an L-shaped hydrophone array was moored inside the area with IW measurements. Acoustic sources transmitting for 7.5 min every 30 min starting on the hour were deployed at the shallow end of the across-shelf path and the outer end of the along-shelf path respectively. The acoustic modal fluctuations in modal arrival time, intensity, temporal coherence, and spatial coherence during the aforementioned IW events are analyzed. The relationship between the modal behavior and IW parameters is investigated and possible IW inversion schemes using acoustic measurements are discussed. [Work supported by ONR322OA.]

10:10–10:25 Break

10:25

2aUW5. Simple expressions for the horizontal array coherence length in shallow water acoustics. James Lynch, Arthur Newhall, Timothy Duda (Dept. Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., M.S. # 11, Bigelow 203, Woods Hole, MA 02543, jlynch@whoi.edu), William Siegmann (Dept. Appl. Mathematics, Rensselaer Polytechnic Inst., Troy, NY), and John Colosi (Dept. Oceanogr., Naval Postgrad. School, Monterrey, CA)

The use of simplified "feature models" (geometric idealizations of specific, isolated ocean features) for coastal oceanographic features can allow one to calculate acoustically useful quantities approximately and even generate analytic forms for them. Feature models for coastal fronts, eddies, internal tides, linear and nonlinear internal waves, and spice are presented and the scattering of sound from these objects is calculated. This allows one to estimate the useful quantity L_{coh}, the horizontal coherence length that represents a physical limit for array signal processing. Calculations of L_{coh} and their comparisons with data will be presented. The effects of acoustic multipath propagation, which can compete with the medium scattering in the estimates of L_{coh}, are also estimated using basic models. [Work sponsored by the Office of Naval Research.]

10:40

2aUW6. Impact of fine-scale sound-speed fluctuations on acoustic autocorrelation times in the East China Sea. Peter C. Mignerey and Altan Turgut (Acoust. Div., Naval Res. Lab., Peter Mignerey Code 7160, Washington, DC 20375-5350, peter.mignerey@nrl.navy.mil)

Autocorrelation times of acoustic signals propagating through shallow oceans are largely driven by sound-speed fluctuations. In August 2008, the Transverse Acoustic Variability Experiment obtained measurements in the East China Sea (65–80 m water depth) of fluctuating signals propagating 33 km from a moored source to a bottomed line array. Supporting environmental measurements were obtained by a towed conductivity-temperature-depth chain. For time periods without large nonlinear internal waves, the measured internal-wave power spectrum shows excess energy at high wavenumbers in comparison with the shallow-water internal-wave model of Levine. Likewise the associated sound-speed fluctuation spectrum exhibits high-wavenumber components in excess of a simple power law. Autocorrelation times of measured 300 Hz acoustic signals were compared with simulated times obtained using a parabolic-equation model to propagate acoustic fields through sound-speed fluctuations driven by linear internal-wave displacements. Results of the comparison are that median measured autocorrelation times (115 s) are shorter than simulations (300 s) driven by spectra with depleted high-wavenumber components. Simple frozen ocean simulations that translate the environment at 0.6 m/s produce autocorrelation times close to the data. [Work supported by the Office of Naval Research.]

10:55

2aUW7. Time reversal of modal arrivals for broadband signals in horizontally stratified shallow water due to internal waves. Mohsen Badiey and Jing Luo (College of Earth, Ocean, and Environ., Univ. of Delaware, 261 S. College Ave., Robinson Hall, Newark, DE 19716, badiey@udel.edu)

Shallow water waveguides in the presence of internal waves can make a time-varying, horizontally stratified medium that can significantly affect broadband pulse propagation. In this paper, we analyze a unique data set obtained during the passage of an internal wave event in a region where a broadband sound source (270 to 330 Hz) was received by an L-shaped hydrophone array about 20 km away in 80 meters on the New Jersey continental shelf. During the time 20:30 and 22:07 GMT, on 17 August 2006, an approaching IW affected the dispersion characteristic of the broadband LFM chirp signal while passing an acoustic track in shallow water waveguide. Modal behavior is examined before and after the internal wave front crossed the source-receiver track. While dispersion characteristics of the signals changed, modal arrival time reversal occurred. The corresponding group and phase velocities that signals experienced during this phenomenon are analyzed using the theory of horizontal rays and vertical modes. These results have motivated theoretical and modeling studies of the waveguide behavior since it was first reported in 2010. [Work supported by ONR322OA.]

11:10

2aUW8. Measured three-dimensional effects of mode-1 and mode-2 nonlinear internal waves on broadband acoustic wave propagation in shallow water. Altan Turgut, Peter C. Mignerey, and Marshall H. Orr (Naval Res. Lab, Acoust. Div., Code 7161, Washington, DC 20375, altan.turgut@nrl.navy.mil)

Horizontal shadowing effects and frequency shifts of acoustic intensity level curves were measured with a bottomed horizontal array in the East China Sea during the summer of 2008. Low-frequency acoustic pulses were transmitted by two fixed sources at 33 km (270–330 Hz LFM) and 20 km (450–550 Hz LFM) range. Strong shadowing effects were observed when mode-1 nonlinear internal wave fronts were nearly parallel to the acoustic propagation path. Numerical studies indicated that shadowing effects are more complex for mode-2 nonlinear internal waves due to acoustic-mode dependent focusing and defocusing. These effects were further analyzed using 3-D PE simulations for more dynamic mode-2 nonlinear internal waves observed on the US New Jersey Shelf. The shadowing was less pronounced for internal waves with curved wave-fronts and small amplitudes. However, regular and irregular frequency shifts were still present. The experimental observations coupled with 3-D PE simulations suggest that nonlinear internal waves may be sensed and characterized via low-frequency acoustic signals. [Work supported by the Office of Naval Research.]

11:25

2aUW9. Propagation of broad-band signals in shallow water in the presence of horizontal stratification. Boris Katsnelson (Marine GeoSci., Univ. of Haifa, Mnt, Carmel, Haifa 31905, Israel, katz@phys.vsu.ru), Andrey Malykhin (Phys., Voronezh Univ., Voronezh, Russian Federation), and Alexander Tckhoidze (Marine GeoSci., Univ. of Haifa, Haifa, Israel)

Horizontal stratification in shelf zone of the ocean is provided by existence of coastal wedge, temperature fronts, nonlinear internal waves, slopes, and canyons, where typical scales are up to tenths of kilometer in range and up to tenths minutes in time, for some perturbations spatial scales are essentially different in different directions in horizontal plane. In this case, there is remarkable horizontal refraction in sound propagation and frequency dependence of horizontal ray trajectories. It means that Fourier components of wideband signal propagate along different paths joining source and receiver in the horizontal plane. Distribution of spectral components in horizontal plane has crescent-like shape and restricted by rays, corresponding to boundary frequencies in spectrum. Propagating wide-band signal has additional spectral distortion as a result of different phase shift for spectral components, propagating along different paths. Also there is difference in directions of wave vectors for difference spectral components (tangents to horizontal rays), leading to phenomena similar to spatial dispersion: different directions of phase and group velocities, compression and decompression of pulses, additional time delay of signal, etc. Mentioned phenomena are considered for models of coastal wedge and temperature fronts. Analytical estimations are presented, as well as results of numerical modeling.

TUESDAY MORNING, 3 DECEMBER 2013 UNION SQUARE 14, 9:00 A.M. TO 10:30 A.M.

Meeting of the Standards Committee Plenary Group

to be held jointly with the meetings of the

ANSI-Accredited U.S. Technical Advisory Groups (TAGs) for: ISO/TC 43, Acoustics, ISO/TC 43/SC 1, Noise, ISO/TC 43/SC 3, Underwater acoustics ISO/TC 108, Mechanical vibration, shock and condition monitoring, ISO/TC 108/SC 2, Measurement and evaluation of mechanical vibration and shock as applied to machines, vehicles and structures, ISO/TC 108/SC 3, Use and calibration of vibration and shock measuring instruments, ISO/TC 108/SC 4, Human exposure to mechanical vibration and shock, ISO/TC 108/SC 5, Condition monitoring and diagnostics of machine systems, and IEC/TC 29, Electroacoustics

P. D. Schomer, Chair, U.S. Technical Advisory Group (TAG) for ISO/TC 43 Acoustics and ISO/TC 43/SC 1 Noise,
Schomer and Associates, 2117 Robert Drive, Champaign, IL 61821

M. A. Bahtiarian, Acting Chair, U.S. Technical Advisory Group (TAG) for ISO/TC 43/SC 3 Underwater acoustics
Noise Control Engineering, Inc., 799 Middlesex Turnpike, Billerica MA 01821

D. J. Evans, Chair of the U.S. Technical Advisory Group (TAG) for ISO/TC 108 Mechanical vibration, shock and condition monitoring, and ISO/TC 108/SC 3 Use and calibration of vibration and shock measuring devices
National Institute of Standards and Technology (NIST), 100 Bureau Drive, Stop 8220, Gaithersburg, MD 20899

W. C. Foiles, Chair of the U.S. Technical Advisory Group (TAG) for ISO/TC 108/SC 2 Measurement and evaluation of mechanical vibration and shock as applied to machines, vehicles and structures
BP America, 501 Westlake Park Boulevard, Houston TX 77079

D. D. Reynolds, Chair, U.S. Technical Advisory Group (TAG) for ISO/TC 108/SC 4 Human exposure to mechanical vibration and shock
3939 Briar Crest Court, Las Vegas, NV 89120

D. J. Vendittis, Chair of the U.S. Technical Advisory Group (TAG) for ISO/TC 108/SC 5 Condition monitoring and diagnostics of machine systems
701 Northeast Harbour Terrace, Boca Raton, FL 33431

P. J. Battenberg, U.S. Technical Advisor (TA) for IEC/TC 29, Electroacoustics
3M Personal Safety Division, Detection Solutions, 1060 Corporate Center Drive Oconomowoc WI 53066

The reports of the Chairs of these TAGs will not be presented at any other S Committee meeting.

The meeting of the Standards Committee Plenary Group will precede the meetings of the Accredited Standards Committees S1, S2, S3, S3/SC 1, and S12, which are scheduled to take place in the following sequence:

Tuesday, 3 December 2013	10:45 a.m. - 12:00 noon	ASC S2, Mechanical Vibration & Shock
Tuesday, 3 December 2013	1:45 p.m. - 3:15 p.m.	ASC S1, Acoustics
Tuesday, 3 December 2013	3:30 p.m. - 5:00 p.m.	ASC S12, Noise
Wednesday, 4 December 2013	9:00 a.m. - 10:30 a.m.	ASC S3, Bioacoustics
Wednesday, 4 December 2013	10:45 a.m. - 12:00 noon	ASC S3/SC 1, Animal Bioacoustics

Discussion at the Standards Committee Plenary Group meeting will consist of national items relevant to all S Committees and U.S. TAGs.

The U.S. Technical Advisory Group (TAG) Chairs for the various international Technical Committees and Subcommittees under ISO and IEC, which are parallel to S1, S2, S3 and S12 are as follows:

U.S. TAG Chair/Vice Chair	TC or SC	U.S. Parallel Committee
ISO		
P.D. Schomer, Chair	**ISO/TC 43** Acoustics	ASC S1 and ASC S3
P.D. Schomer, Chair	**ISO/TC 43/SC1** Noise	ASC S12
M.A. Bahtiarian, Acting Chair	**ISO/TC 43/SC 3**, Underwater acoustics	ASC S1, ASC S3/SC 1 and ASC S12
D.J. Evans, Chair	**ISO/TC 108** Mechanical vibration, shock and condition monitoring	ASC S2
W.C. Foiles, Co-Chair	**ISO/TC 108/SC2** Measurement and evaluation of mechanical vibration and shock as applied to machines, vehiclesand structures	ASC S2
D.J. Evans, Chair	**ISO/TC 108/SC3** Use and calibration of vibration and shock measuring instruments	ASC S2
D.D. Reynolds, Chair	**ISO/TC 108/SC4** Human exposure to mechanical vibration and shock	ASC S3
D.J. Vendittis, Chair	**ISO/TC 108/SC5** Condition monitoring and diagnostics of machine systems	ASC S2
IEC		
P.J. Battenberg, U.S. TA	**IEC/TC 29** Electroacoustics	ASC S1 and ASC S3

TUESDAY MORNING, 3 DECEMBER 2013 UNION SQUARE 14, 10:45 A.M. TO 12:00 NOON

Meeting of Accredited Standards Committee (ASC) S2 Mechanical Vibration and Shock

A.T Herfat, Chair ASC S2
Emerson Climate Technologies, Inc., 1675 West Campbell Road, PO Box 669, Sidney, OH 45365- 0669

C.F. Gaumond, Vice Chair ASC S2
Naval Research Laboratory, Code 7142, 4555 Overlook Ave. SW, Washington DC 20375-5320

Accredited Standards Committee S2 on Mechanical Vibration and Shock. Working group chairs will report on the status of various shock and vibration standards currently under development. Consideration will be given to new standards that might be needed over the next few years. Open discussion of committee reports is encouraged.

People interested in attending the meeting of the TAG for ISO/TC 108, Mechanical vibration, shock and condition monitoring, and four of its subcommittees, take note - that meeting will be held in conjunction with the Standards Plenary meeting at 9:00 a.m. on Tuesday, 3 December 2013.

Scope of S2: Standards, specification, methods of measurement and test, and terminology in the field of mechanical vibration and shock, and condition monitoring and diagnostics of machines, including the effects of exposure to mechanical vibration and shock on humans, including those aspects which pertain to biological safety, tolerance and comfort.

Session 2pAA

Architectural Acoustics, Noise, and ASA Committee on Standards: Innovations and Challenges in Healthcare Acoustics

Erica Ryherd, Cochair
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Acoustics By Design, Inc., 124 Fulton St. East, Second Fl., Grand Rapids, MI 49503

Chair's Introduction—1:00

Invited Papers

1:05

2pAA1. Research methods: Beta-testing Facilities Guideline Institute's acoustical criteria—A case study of continual improvement in healthcare facilities. Kurt Rockstroh (Board of Directors, Facility Guidelines Inst., Falcon Pier, Boston, MA 55801, kurtr@steffian.com)

Continual improvement processes (CIP) present opportunities and challenges for researchers. In healthcare, FGI manages a four-year building-code-development cycle that is continually improved by research and sets standards that others follow (USGBC-LEED, ICC). Innovative, accelerated research methods are needed to continually improve healthcare environments during a period of rapid and profound change, so FGI's CIP method seeks to synchronize the work of code writers and independent researchers. One useful research method is to subject draft requirements to "beta-testing." FGI leaders (led by the presenter) did this to test new acoustical criteria on a 641,000 "hospital of the future" wing at Tufts University's Baystate Medical Center in Springfield MA (a major hospital and Level 1 trauma center). Planning began in 2006 (using draft #1 of the acoustical guidelines); occupancy began in late 2012 (after the guidelines' publication, Jan 2010). But only now, after a full year of occupancy, can the full cost and benefit of the improvements be measured using Federal HCAHPS (a new procedure required by the Affordable Care Act that impacts hospital reimbursements). "Beta test" results were factored into the forthcoming FGI 2014 edition. The "beta test" is the subject of independent, funded research currently underway.

1:25

2pAA2. New healthcare safety risk assessment toolkit includes acoustics and noise control design measures for error prevention in medication safety zones. Mandy Kachur (Soundscape Engineering LLC, 317 S Div. St. #170, Ann Arbor, MI 48104, mkachur@soundscapeengineering.com), Xiaobo Quan (The Ctr. for Health Design, Concord, CA), Daniel M. Horan (Cavanaugh Tocci Assoc., Inc., Sudbury, MA), and David M. Sykes (Acoust. Res. Council, Lincoln, MA)

The Facilities Guidelines Institute (FGI) manages the *Guidelines for Design and Construction of Hospitals and Outpatient Facilities*, which is used by the Joint Commission, many federal agencies, and authorities having jurisdiction in 42 states. The forthcoming 2014 edition calls for completion of a Safety Risk Assessment (SRA) to guide healthcare facility planning and design teams through systematic evaluation of safety issues in the built environment. In 2012, FGI and the federal Agency for Healthcare Research and Quality commissioned The Center for Health Design to oversee a three year effort to develop, test, and disseminate an SRA toolkit. The toolkit will consist of an online facility design questionnaire, supporting whitepapers and guidelines, and education for the healthcare community. The Acoustics Research Council was invited to contribute by integrating noise control language into the medication safety segment of the online questionnaire tool. The result is a question that specifically addresses acoustics and noise control design for error prevention in medication safety zones. In May 2013, the questionnaire content was finalized, and it will be validated, integrated, and evaluated during the subsequent two years. The co-authors are project participants.

1:45

2pAA3. New healthcare acoustics subcommittee: Overview and call for participation. Gary Madaras (Making Hospitals Quiet, Chicago Metallic, 4849 S. Austin Ave., Chicago, IL 60638, DoctorSonics@aol.com)

The Acoustical Society of America (ASA) Technical Committee on Architectural Acoustics (TCAA) formed a new Healthcare Acoustics subcommittee at the 165th meeting in Montreal. The 1st meeting of the Healthcare Acoustics subcommittee will be held at the 166th meeting in San Francisco. This presentation will provide an overview of the purpose of the subcommittee and introduce possible initiatives on which the subcommittee may want to initially focus. *Examples*: 1. Despite financial incentive to make their hospitals quieter, most healthcare providers cannot significantly increase their "quiet-at-night" (HCAHPS) scores in their older or newer inpatient facilities. The national "quiet-at-night" (HCAHPS) score remains the lowest of all patient experience quality indicators. Changing things may require a significant paradigm shift in the architectural and acoustical communities. 2. The Facility Guidelines Institute (FGI) has developed its new *Guidelines for Design and Construction of Residential Health, Care, and Support Facilities* to be published in early 2014. Those Guidelines for eldercare facilities will include terms and concepts such as "nature sounds," "positive auditory distraction," "auditory landmark," "music," "quiet rooms," "quietly operating," etc., that have yet be fully defined acoustically. Please attend and participate.

2:05

2pAA4. Value based education for healthcare design. Edward Logsdon (D. L. Adams Associates, Inc., 1536 Ogden St., Denver, CO 80218, elogsdon@dlaa.com and Kenric Van Wyck (Acoustics By Design, Grand Rapids, MI 49503)

Accelerating the understanding and adoption of the FGI Guidelines is an essential part of the continuous improvement process—especially at a time when healthcare is coping with so much rapid change, and while the FGI Guidelines are becoming more accepted internationally. As part of the process to improve the acoustical environment in healthcare facilities, there is also the continual need to educate architects and designers concerning sound and vibration design criteria included in the FGI Guidelines. Designers need to understand how to apply the criteria, proposed changes to the guidelines, and how the changes will benefit the hospital by improving the patient experience. Using professional social networking, electronic, and face-to-face continuing education combined with research based design, professionals and the public can be educated on the merits of improvement in the acoustical design of healthcare facilities. Web-based education will attract a much wider audience worldwide and further international acceptance. Project case studies where FGI Guidelines were applied will identify the cost impact while increasing awareness and understanding.

2:20

2pAA5. Top ten research needs in the decade ahead. David M. Sykes and William J. Cavanaugh (Architectural Acoust., Rensselaer Polytechnic Inst., 31 Baker Farm Rd., Lincoln, MA 01773, david.sykes@remington-partners.com)

Independent, third-party research co-led by teams including medical personnel, engineers, and scientists is key to raising the bar on acoustical performance in healthcare facilities. While links between noise and health have been officially ignored in the United States for three decades, research over the past eight years in healthcare environments has significantly advanced understanding and created a small community of funding organizations and interested researchers. To make further progress, researchers must demonstrate clear links between noise/sound and patient outcomes as well as the performance effectiveness of healthcare professionals on critical factors such as error rates. Achieving useful, translational results requires: innovative research methods, collaborative funding mechanisms, and a clear focus on outcomes. With no Federal agency providing organized peer-review, it is essential for the acoustics research community to both agree on a research agenda and organize peer-review procedures that will enable continued progress among a widely distributed population of researchers working within the constraints of limited funding. The presenter worked with FGI to draft a "top 10" research agenda. He co-chairs the FGI Acoustics Working Group, edits the Springer Verlag "Quiet series," and has experimented with collaborative research funding mechanisms over the past three decades.

Contributed Paper

2:40

2pAA6. Acoustic quality of buildings: Contributions to the workers health of the health area. Marta R. Macedo (Coordenação de Saúde do Trabalhador, Fundação Oswaldo Cruz, Av. Brasil 4365, Pavilhão Carlos Augusto da Silva, Sala 202, Rua Comandante Rubens Silva 90, ap. 203, bl 1, Rio de Janeiro, Rio de Janeiro 21040-360, Brazil, mribeiro@fiocruz.br), Marcia S. Almeida, Liliane R. Teixeira (Escola Nacional de Saúde Pública, Fundação Oswaldo Cruz, Rio de Janeiro, Rio de Janeiro, Brazil), Ana Paula Gama, Stephanie Livia S. Silva, Olga Dick (Coordenação de Saúde do Trabalhador, Fundação Oswaldo Cruz, Rio de Janeiro, Brazil), Isabele C. Costa, Denise Torreao (Escola Nacional de Saúde Pública, Fundação Oswaldo Cruz, Rio de Janeiro, Brazil), Paulo Roberto L. Jorge (Coordenação de Saúde do Trabalhador, Fundação Oswaldo Cruz, Rio de Janeiro, Brazil), Paulo Marcelo S. Dias, Daniel Valente, and Diane R. Valente (Escola Nacional de Saúde Pública, Fundação Oswaldo Cruz, Rio de Janeiro, Brazil)

This work presents a case study carried through in a research center, education, assistance, and technological development in the areas of the health of the woman, child, and adolescent, situated in Rio de Janeiro, where pathologies of average and high complexity are taken care of. It intended detect environmental problems that could affect the health of workers of the institution, in order to support the elaboration of the action plan to mitigate urgent questions and to develop an architectonical project for a new center. The environments and processes of work had been evaluated, through a participative approach, being visited all the sectors of the center, carried through interviews half-structuralized with workers of all the sectors and promoted debates that had pointed the noise as one of the main factors of bother and stress. The observation in loco and analyze of the project allowed to detect that the architectural design and the disrespect of basic acoustics recommendations had contributed for this picture. Measurements carried through in some pointed areas as uncomfortable indicated to have exposition to sound pressure level above 75 dB(A). However, the resolution of the many of identified acoustic problems will have to wait the construction of the new headquarters.

2:55–3:10 Break

Invited Papers

3:10

2pAA7. Why alarm fatigue is a pivotal issue that affects the acoustical design of healthcare facilities. Paul Barach MD (Res. Committee, Facility Guidelines Inst., 31 Baker Farm Rd., Lincoln, MA 01773, pbarach@gmail.com)

The U.S. FDA and Joint Commission designated "alarm fatigue" the "#1 priority in healthcare technology" in 2011–2012, acknowledging that this acoustical problem results in hundreds of patient deaths and thousands of injuries. The healthcare facilities industry has been slow to recognize that "alarm fatigue" is partly a facility design issue: i.e., a cacophony of recurrent noises from myriad uncorrelated medical devices, set at maximum loudness, occurring in hard-walled, reverberant spaces (such as patient rooms, ORs, and ICUs) produce elevated stress, sleep impairment, disorientation, and dangerously irrational, potentially deadly behavior. "Alarm fatigue" has been addressed as a human factors problem elsewhere: e.g., nuclear plant control rooms (after Three-Mile Island) and aircraft cockpits. In healthcare, it is imperative to engage architects, designers, acoustical engineers facility engineering staffs, and clinicians, who represent the "first line of defense" as the medical device industry requires 5–10 years to implement solutions. The presenter co-led a delegation of 12 distinguished members of the acoustics profession to the national summit on "alarm fatigue," Washington DC, 2011 and has co-authored peer-reviewed medical journal articles and a forthcoming FGI white paper on the subject. This presentation focuses on solutions, challenges, and the research roadmap needed to address "alarm fatigue."

3:30

2pAA8. The healthcare acoustics research team: Bridging the gap between architecture, engineering, and medicine. Erica Ryherd (Woodruff School of Mech. Eng., Georgia Inst. of Technol., Mech. Eng., Atlanta, GA 30332-0405, erica.ryherd@me.gatech.edu), Jeremy Ackerman (Dept. of Emergency Medicine, Emory Univ., Atlanta, GA), Craig Zimring (College of Architecture, Georgia Inst. of Technol., Atlanta, GA), and Kerstin Persson Waye (Occupational and Environ. Medicine, Gothenburg Univ., Gothenburg, Sweden)

Healthy soundscapes are paramount to the missions of hospitals: patients need to sleep and heal without environmental stressors; staff, patients, and family need to communicate accurately but privately; staff need to be able to localize alarms and calls for help. This talk discusses recent findings from the Healthcare Acoustics Research Team (HART), an international, interdisciplinary collaboration of specialists in architecture, engineering, medicine, nursing, and psychology. Members of the HART network are actively engaged in research in the United States and Sweden, having worked in more than a dozen hospitals and a broad range of unit types including adult and neonatal intensive care, emergency, operating, outpatient, long-term care, mother-baby, and others. Highlights will include projects relating noise and room acoustic measures to staff and patient response in addition to studies evaluating impacts of acoustic retrofits. Results show that effective hospital soundscapes require a complex choreography of architectural layout, acoustic design, and administrative processes that is only beginning to be fully understood.

3:50

2pAA9. Acoustic comfort in healthcare facilities—What is it? and What does it mean to both patients and medical staff? Kenneth P. Roy (Bldg. Products Technol. Lab, Armstrong World Industries, 2500 Columbia Ave., Lancaster, PA 17603, kproy@armstrong.com)

Acoustic comfort is all about meeting the acoustic needs of the patient, family, professional staff in hospitals, clinics, pharmacies, etc. These needs include low distraction and annoyance for patients to fully enjoy a "healing environment", high levels of speech intelligibility to support communications and to help reduce medical errors, and adequate speech privacy to meet HIPAA and other privacy needs. How to design for these, and how to verify performance—these are the key issue to be addressed. Architectural choices need to be made with a view for acoustic comfort in addition to all the other relevant design factors. Facility evaluations as part of the commissioning process need to include both objective and subjective surveys to push forward evidence based design approaches for others to take advantage of in future designs. All of these issues are part of the ongoing discussions and field studies that we have been having, and will continue to have.

4:10

2pAA10. Analysis of granular hospital sound level measurements. Benjamin Davenny (Acentech Inc., Cambridge, MA), Aaron Betit (Acentech Inc., 601 South Figueroa St., Ste. 4050, Los Angeles, CA 90017, abetit@acentech.com), and William Yoder (Acentech Inc., Cambridge, MA)

Remote sound monitoring systems were deployed in several locations in a hospital to measure sound levels due to building system and activity noise. In addition to 5 min interval data, granular sound levels were recorded several times per second to provide flexibility in data analysis. Analysis of this information will be presented.

4:30

2pAA11. Estimation of noise induced hearing loss because of indoor and outdoor environment noise factors in Turkish healthcare facilities: A survey of hospitals in Turkey. Filiz Kocyigit (Architecture, Atilim Univ., İncek, Ankara, Turkey, filizbk@gmail.com)

This article aims to evaluate effect of indoor and outdoor environment noise factors in healthcare facilities which affect the work area due to noise induced hearing loss, and also to determine their relationship with the architectural design of the building. As a case study; noise levels, in five state and private healthcare centers, including medical schools, research hospitals, and state hospitals in Turkey were measured. They were compared with similar healthcare centers in the United States. Results include equivalent sound pressure levels (Leq for 5 min from 20 spots in each area), and Lmax–Lmin evaluated as a function of location, frequency, time, and days of the week. Research showed that no location was in compliance with current World Health Organization Guidelines, and a review of

objective data indicated that this was true of hospitals throughout the world. Noise induced hearing loss on continuously users had been estimated with these results in selected hospitals. Data gathered at various hospitals for last decay indicate a trend of increasing noise levels during daytime and nighttime hours. The implications of these results were significant for patients, visitors, and hospital staff.

4:50

2pAA12. Renovation of neonatal intensive care unit per the Facilities Guideline Institute's 2010 Acoustical Design Guidelines. Erik Miller-Klein (SSA Acoust., LLP, 222 Etruria St, Ste. 100, Seattle, WA 98109, erik@ssaacoustics.com)

We did an evaluation of the acoustical conditions of the existing neonatal intensive care unit at the St. Joseph's Hospital in Tacoma, Washington, and issued detailed renovation recommendations that were integrated into the final design. At the conclusion of construction, our team was able to evaluate the improvements of the acoustical conditions per the 2010 FGI Guidelines and the original performance. These included noise impacts from door closures, curtains, alarms, and the conversations.

5:10–6:10 Panel Discussion

TUESDAY AFTERNOON, 3 DECEMBER 2013 UNION SQUARE 23/24, 1:00 P.M. TO 3:20 P.M.

Session 2pABa

Animal Bioacoustics and Acoustical Oceanography: Broadening Applications of Tags to Study Animal Bioacoustics II

Marla M. Holt, Cochair
NOAA NMFS NWFSC, 2725 Montlake Blvd. East, Seattle, WA 98112

Alison K. Stimpert, Cochair
Dept. of Oceanogr., Naval Postgrad. School, Monterey, CA 93943

Invited Papers

1:00

2pABa1. Conservation applications of baseline acoustic tag data from right whales. Susan Parks (Dept. of Biology, Syracuse Univ., 114 Life Sci. Complex, Syracuse, NY 13244, sparks@syr.edu), Douglas P. Nowacek (Nicholas School of the Environ. and Pratt School of Eng., Duke Univ. Marine Lab., Beaufort, NC), Mark Johnson, and Peter L. Tyack (Sea Mammal Res. Unit, Scottish Oceans Inst., School of Biology, Univ. of St. Andrews, St. Andrews, Scotland, United Kingdom)

Data collected from archival acoustic tags have dramatically improved our understanding of marine mammal subsurface behavior. Some of the earliest tag deployments were made on the endangered North Atlantic right whale (*Eubalaena glacialis*) to better understand their behavior and to contribute to their conservation. Over the past 15 years, the database of acoustic tag records collected from a diverse cross section of the population has served as a valuable resource for research. Acoustic tag data have provided critical behavioral information including details on individual calling behavior in designated critical habitat feeding grounds, behavioral responses to controlled exposures of acoustic signals including conspecific and man-made signals, acoustic responses to vessel noise, and ground truth data for passive acoustic monitoring efforts. Current studies continue to utilize tag data to shed light on the individual, seasonal, and regional variation in acoustic behavior of right whales, particularly focused on the shallow water coastal habitats of the calving grounds off of Florida and Georgia and in the migration corridor off the Eastern United States. These studies have contributed data that are crucial for targeted conservation efforts and highlight the value of long-term databases of tag data in baleen whale research.

1:20

2pABa2. Controlled sound exposure experiments to measure marine mammal reactions to sound: Southern California behavioral response study. Brandon L. Southall (SEA, Inc., 9099 Soquel Dr., Ste. 8, Aptos, CA 95003, Brandon.Southall@sea-inc.net), John Calambokidis (Cascadia Res. Collective, Olympia, WA), Moretti David (Naval Undersea Warfare Ctr., Newport, RI), Jay Barlow (Southwest Fisheries Sci. Ctr., La Jolla, CA), Stacy DeRuiter (CREEM The Observatory, St. Andrews, Scotland, United Kingdom), Jeremy Goldbogen (Cascadia Res. Collective, Olympia, WA), Ari Friedlaender (Long Marine Lab., Univ. of California, Santa Cruz, Santa Cruz, CA), Elliott Hazen (NOAA-PFEL, Pacific Grove, CA), Alison Stimpert (Naval Postgrad. School, Monterey, CA), Arranz Patricia (Sea Mammal Res. Unit, Scottish Oceans Inst., St. Andrews, Scotland, United Kingdom), Erin Falcone, Greg Schorr, Annie Douglass (Cascadia Res. Collective, Olympia, WA), Chris Kyburg (SPAWAR Systems Ctr., San Diego, CA), and Peter Tyack (Sea Mammal Res. Unit, Scottish Oceans Inst., St. Andrews, Scotland, United Kingdom)

SOCAL-BRS is an inter-disciplinary collaboration designed to increase understanding of marine mammal behavior and provide a robust scientific basis for estimating risk and minimizing effects of mid-frequency military sonar systems. Data were collected using visual observations, passive acoustic monitoring, animal-attached acoustic and movement tags, photo ID, biopsy, and controlled sound exposure experiments on over 20 cetacean species in biologically important areas throughout the southern California Bight. Ninety-six individuals of ten species were tagged with six tag types, including two species [Baird's beaked whale (*Berardius bairdii*), Risso's dolphin (*Grampus griseus*)] not previously studied with such tools. Fifty-six controlled CEEs were conducted using protocols and protective measures to ensure animals were not harmed. Simulated sonar signals were projected through a deployed sound source and changes in vocal, diving, and horizontal movement behavior were measured. Results demonstrate that Cuvier's beaked whales (*Ziphius cavirostris*) react most strongly to simulated sonar exposures with clear changes in vocal and diving behavior and avoidance responses at low received sound levels. Blue whale (*Balaenoptera musculus*) responses are more variable, depending on complex interactions of exposure and behavioral conditions. Ongoing efforts include expanding sample sizes in other species using simulated sounds and the novel inclusion of operational mid-frequency sonars.

1:40

2pABa3. Noise design tradeoffs for a general-purpose broadband acoustic recording tag. William C. Burgess (Greeneridge Sci., Inc., 6060 Graham Hill Rd. Stop F, Felton, CA 95018, burgess@greeneridge.com), Susanna B. Blackwell (Greeneridge Science, Inc., Aptos, CA), and Patrick Dexter (Greeneridge Science, Inc., Ojai, CA)

To design instruments for general-purpose use invokes both blessings and curses; blessings because a flexible product may leverage its development effort across many different applications, but curses because it may not be perfect for any of them. Designing for noise performance of broadband acoustic recording tags epitomizes this tradeoff. Lower self noise typically requires more power and a larger transducer, and may come at the expense of clipping strong sounds, all of which impact tag applications. Higher self noise, however, reduces detection range under quiet conditions and diminishes utility for noise monitoring. The electronic design of the Acousonde^{TM} acoustic/ultrasonic recording tag navigates this challenge using two acoustic channels with very different gains (29–49 versus 14–34 dB), noise (minimum 40 versus 70 dB re 1 μPa^2/Hz at 1 kHz), and bandwidths (42 versus 9.3 kHz). Mechanically, design for hydrodynamics reduces turbulent flow noise especially once the tag aligns with flow direction due to self-orienting. As a result of these design elements, the Acousonde's noise performance is comparable to that of a much larger recording instrument while preserving capability as an animal tag. [Development supported by ONR.]

2:00

2pABa4. The next generation of multi-sensor acoustic tags: Sensors, applications, and attachments. Douglas Nowacek (Nicholas School of the Environ. and Pratt School of Eng., Duke Univ. Marine Lab., 135 Duke Marine Lab Rd., BEAUFORT, NC 28516, dpn3@duke.edu), Matthew Bowers (Marine Sci. and Conservation, Duke Univ. Marine Lab, BEAUFORT, NC), Andrew Cannon (1900 Eng., Greenville, SC), Mark Hindell (Inst. of Marine and Antarctic Sci., Univ. of Tasmania, Hobart, TAS, Australia), Laurens E. Howle (Mech. Eng. and Mater. Sci., Duke Univ., Durham, NC), Mark M. Murray (Mech. Eng., US Naval Acad., Annapolis, MD), Dan Rittschof (Marine Sci. and Conservation, Duke Univ. Marine Lab, Beaufort, NC), K. Alex Shorter, and Michael Moore (Biology Dept., Woods Hole Oceanographic Inst, Woods Hole, MA)

From Kooyman's 1963 wind-up kitchen timer TDR, multi-sensor tags have evolved significantly over the last twenty years. These advancements, including high fidelity acoustics, have been driven by improved sensing and electronics technology, and resulted in highly integrated mechatronics systems for the study of free ranging animals. In the next decade, these tags will continue to improve, and promising work has begun in three key areas: (i) new sensors; (ii) expanding uses of existing sensors; and (iii) increasing attachment duration and reliability. The addition of rapid acquisition GPS and the inclusion of gyroscopes to separate the dynamic acceleration of the animal from gravitational acceleration, are underway but not widely available to the community. Existing sensors could be used for more and different applications, e.g., measuring ambient ocean noise. Tags attached to pinnipeds in the Southern Ocean, for example, could provide noise measurements from remote areas. Finally, attachment duration has been limiting for cetaceans because the suction cups typically used do not reliably stay attached for more than a day. We will present data on engineering efforts to improve attachments: (i) improved tag hydrodynamics; (ii) incorporating bio-compatible glues; and (iii) micro structuring tag components to utilize hydrostatic forces and enhance adhesion.

2:20

2pABa5. The relationship between vessel traffic and noise levels received by killer whales. Juliana Houghton (School of Aquatic & Fishery Sci., Univ. of Washington, Box 355020, Seattle, WA 98195, stephj5@uw.edu), Marla Holt (NOAA/NMFS/Northwest Fisheries Sci. Ctr., Seattle, WA), Deborah Giles (Dept. of Wildlife, Fish, and Conservation Biology, Univ. of California, Davis, CA), Candice Emmons, Brad Hanson (NOAA/NMFS/Northwest Fisheries Sci. Ctr., Seattle, WA), Jeff Hogan (Cascadia Res. Collective, Olympia, WA), Trevor Branch, and Glenn VanBlaricom (School of Aquatic & Fishery Sci., Univ. of Washington, Seattle, WA)

Cetaceans that rely on their acoustic environment for key life history strategies are susceptible to noise effects from anthropogenic use such as ecotourism. Endangered Southern Resident killer whales (SRKW) are the primary target for vessel-based whale-watching in the Salish Sea. Vessel interactions and associated noise have been identified as potential stressors for SRKW. Previous research has indicated that both stressors negatively impact SRKW; however, there is a missing link between vessel characteristics/behavior and noise levels actually received by individual whales. To investigate this relationship, data were collected concurrently using mobile remote sensing survey equipment packages and digital acoustic recording tags. This allowed us to obtain precise geo-referenced vessel data and noise levels received by the whales. We used linear regression to summarize patterns in vessel characteristics and relate them to received noise levels. Received noise levels (RNL) were correlated with the number of vessels. RNL also increased when larger vessels were present or when vessels were traveling at relatively high speed. These findings facilitate improved understanding of the contributions of vessel characteristics to the noise levels received by individual cetaceans. Results from this study can be used to refine existing vessel regulations in order to better manage SRKW to recovery.

2:35

2pABa6. Use of an animal-borne active acoustic tag to conduct minimally-invasive behavioral response studies. Selene Fregosi, Holger Klinck (Cooperative Inst. for Marine Resources Studies, Oregon State Univ. and NOAA Pacific Marine Environ. Lab., Hatfield Marine Sci. Ctr., 2030 SE Marine Sci. Dr., Newport, OR 97365, selene.fregosi@noaa.gov), Markus Horning (Marine Mammal Inst., Oregon State Univ., Newport, OR), David K. Mellinger (Cooperative Inst. for Marine Resources Studies, Oregon State Univ. and NOAA Pacific Marine Environ. Lab., Newport, OR), Daniel P. Costa (Dept. of Ecology and Evolutionary Biology, Univ. of California, Santa Cruz, Santa Cruz, CA), David A. Mann (Loggerhead Instruments, Sarasota, FL), Kenneth Sexton (The Sexton Co., Salem, OR), and Luis Huckstadt (Dept. of Ecology and Evolutionary Biology, Univ. of California, Santa Cruz, Santa Cruz, CA)

A pilot study was conducted to evaluate the potential of animal-borne active and passive acoustic tags for conducting minimally-invasive behavioral response studies on pinnipeds. A prototype tag was developed and tested on juvenile northern elephant seals (*Mirounga angustirostris*) using translocation experiments at Año Nuevo State Park, CA, USA, in spring 2012. The principal scientific questions of this pilot study were (1) do low-intensity sounds emitted by an animal-borne tag elicit behavioral responses, and (2) are potential animal responses related to signal content (e.g., threatening vs non-threatening)? Preliminary results indicate that (1) low-intensity sounds emitted by animal-borne tags elicit distinct behavioral responses, (2) these responses appear related to signal content, and (3) the responses may differ based on depth, bathymetry, and location. The results of the study show the promise of this approach as a minimally invasive and cost-effective method to investigate animal responses to underwater sounds, as well as a method to develop mitigation strategies. We are currently in the process of improving the tag design for future field efforts with the goal to increase the sample size, range of acoustic stimuli, and age/sex classes of tagged seals. [Funding from NOAA/NMFS Ocean Acoustics Program.]

2:50

2pABa7. Putting tags in the researcher's toolkit: An examination of the strengths, limitations, and added-value from animal tagging. Robert Gisiner (NAVFAC EXWC EV, US Navy, 1000 23rd Ave., Port Hueneme, CA 93043, bob.gisiner@navy.mil)

The US Navy, through the Office of Naval Research and other offices, has focused on improving animal tags, reducing cost, and increasing availability. From data-rich packages like video and acoustic dataloggers to simple location-only tags, tags provide a variety of new data to studies of marine animals and their ecosystems. Tags realize their full potential when calibrated or validated against other existing alternative sensor systems like visual surveys, photo-identification, genetics, and acoustic monitoring. When tag cost, cost of delivery and recovery or monitoring are weighed against the data uniquely available from tags, an integrated data collection strategy involving animal tagging can be developed to generate the best data at the optimal total cost for a given research or resource management scenario.

3:05–3:20 Panel Discussion

Session 2pABb

Animal Bioacoustics: Noise Impacts on Marine Life

Michael A. Stocker, Chair
Ocean Conservation Research, P.O. Box 559, Lagunitas, CA 94938

Contributed Papers

3:30

2pABb1. Passive acoustic monitoring for marine mammals during Navy explosives training events off the coast of Virginia Beach, Virginia. Cara F. Hotchkin, Mandy Shoemaker, Anurag Kumar (Naval Facilities Eng. Command, Atlantic, 6506 Hampton Blvd., Norfolk, VA 23508, cara.hotchkin@navy.mil), Carl Hager (U.S. Naval Acad., Annapolis, MD), David MacDuffee, Jene Nissen, and Ronald Filipowicz (U.S. Fleet Forces Command, Norfolk, VA)

Navy training events involving the use of explosives pose a potential threat to marine mammals. This study used passive acoustic and visual monitoring data to evaluate marine mammals' behavioral responses to noise from explosive events. Monitoring was conducted during five training events in the Virginia Capes (VACAPES) Range Complex during August/September of 2009–2012. Passive acoustic monitoring methods ranged from a single hydrophone to an array of sonobuoys monitored in real time. Visual monitoring effort over the five events totaled approximately 34 h (day before events: 10.1 h; days of events: 22.3 h; day after events: 1.5 h), yielding a total of 27 marine mammal sightings. Approximately 54 h of acoustic data were collected before, during, and after the five events. Behavioral changes were evaluated based on analysis of vocalizations detected before, during, and after explosions and concurrent data from visual sightings. For time periods with both visual and acoustic monitoring data, detection methods were compared to evaluate effectiveness. Continuing use and evaluation of both visual and passive acoustic methods for monitoring of explosive training events will improve our knowledge of potential impact resulting from explosive events and help improve management and conservation of marine mammals.

3:45

2pABb2. Use of Automated passive acoustic monitoring methods for monitoring for marine mammals in conjunction with US Navy Mid-frequency Active Sonar training events. Stephen W. Martin, Roanne A. Manzano-Roth, and Brian M. Matsuyama (SSC PAC, 53560 Hull St., Code 71510, San Diego, CA 92152, steve.w.martin@navy.mil)

Automated passive acoustic detection, classification, and localization (DCL) methods are employed to deal with large volumes of acoustic data to support estimating the sound pressure levels (SPLs) that marine mammals are exposed to from mid-frequency active sonar (MFAS) during US Naval training events. These methods are applied to a training event involving MFAS conducted February 2012 in Hawaiian waters with thirty one hydrophones of data collected continuously over an 11 day period. The automated methods detect and determine locations of marine mammals, specifically minke and beaked whales, and the times of the MFAS transmissions utilizing custom C++ algorithms. Streamlined manual validation methods are employed which utilize custom MATLAB display routines. Animal locations uncertainties are addressed for the two different species. Once the transmitting ship and animal locations are determined acoustic propagation modeling is utilized to estimate the sound pressure levels (in dB re 1 micro Pascal) that an animal, or group of animals, were exposed to. Surface ducted propagation conditions can result in species such as beaked whales being exposed to over 30 dB higher SPL's when they return to the surface to breathe compared to when at depth foraging.

4:00

2pABb3. Impact of underwater explosions on cetaceans. Simone Baumann-Pickering (Scripps Inst. of Oceanogr., Univ. of California San Diego, 9500 Gilman Dr., La Jolla, CA 92093, sbaumann@ucsd.edu), Amanda J. Debich, Ana Širović (Scripps Inst. of Oceanogr., Univ. of California San Diego, San Diego, CA), James V. Carretta (Southwest Fisheries Sci. Ctr., National Oceanic and Atmospheric Administration, San Diego, CA), Jennifer S. Trickey, Rohen Gresalfi (Scripps Inst. of Oceanogr., Univ. of California San Diego, San Diego, CA), Marie A. Roch (Dept. of Comput. Sci., San Diego State Univ., San Diego, CA), Sean M. Wiggins, and John A. Hildebrand (Scripps Inst. of Oceanogr., Univ. of California San Diego, San Diego, CA)

Use of seal bombs to deter sea lions from being caught in nets and preying on catch is a common practice for a number of fisheries. Purse seine fisheries in Southern California target primarily squid, but also scombrids and baitfish such as sardine and anchovy, while set gillnet fisheries' primary catch are halibut and white seabass. All of these fisheries use seal bombs as deterrents. Continuous passive acoustic recordings at several sites in the Southern California Bight collected since 2007 revealed an extensive use of smaller explosives, most likely seal bombs, during nighttime hours with a seasonal occurrence matching fishery activities. During several months of the year they were used all night, every night. The median occurrence of explosions when detected was 8 per hour; however, during periods of high fishing effort they reached up to 480 explosions per hour. From behavioral response and opportunistic studies we know that beaked whales as well as endangered blue whales react negatively to anthropogenic sound sources. We are testing the hypothesis that these underwater explosions have a suppressive effect on the acoustic behavior and therefore the communication and foraging of cetaceans, possibly leading to impacts on the individual fitness and overall population health.

4:15

2pABb4. Monitoring of marine mammal occurrence and acoustic behaviors in relation to mid-frequency active sonar using autonomous recorders deployed off the undersea warfare training range, Florida. Thomas F. Norris, Julie Oswald, Tina M. Yack, Elizabeth Ferguson (Bio-Waves, Inc., 144 W. D St., Ste. #205, Encinitas, CA 92024, thomas.f.norris@bio-waves.net), Anurag Kumar (Naval Facilities Eng. Command Atlantic, U.S. Navy, Norfolk, VA), Jene Nissen (U.S. Fleet Forces Command, U.S. Navy, Norfolk, VA), and Joel Bell (Naval Facilities Eng. Command Atlantic, U.S. Navy, Norfolk, VA)

Passive acoustic data were collected from nine Marine Autonomous Recording Units (MARUs) deployed 60–150 km in an area that coincides with the U.S. Navy's planned Undersea Warfare Training Range (USWTR) off Jacksonville FL. MARUs were deployed for 26 days during fall 2009, and 37 days in winter 2009–2010. Data were manually reviewed for marine mammal vocalization events, man-made noise, and mid-frequency active sonar events, which were logged using TRITON software. Seasonal and diel patterns were characterized qualitatively. Patterns and probabilities of vocalization events by species, or species groups, were related to sonar events. Vocalizations were detected for minke whales, North Atlantic right

2p TUE. PM

whales, sei whales, humpback whales, sperm whales, the blackfish group, and delphinids. Minke whale pulse-trains occurred almost continuously during the winter deployment but were absent in fall. Right whale events occurred mostly during winter at shallow-water sites, but unexpectedly were also detected at deep-water sites. Sperm whale events occurred exclusively near the continental shelf break and exhibited a strong diel pattern. Minke whale events had a strong negative relationship with sonar events. These results provide an initial assessment of marine mammal occurrence within the Navy's planned USWTR, and provide new information on vocalization events in relation to sonar.

4:30

2pABb5. Vocalization behaviors of minke whales in relation to sonar in the planned Undersea Warfare Training Range off Jacksonville, Florida. Talia Dominello, Thomas Norris, Tina Yack, Elizabeth Ferguson, Cory Hom-Weaver (Bio-Waves Inc., 364 2nd St. #3, Encinitas, CA 92024, talia.dominello@bio-waves.net), Anurag Kumar (Naval Facilities Eng. Command Atlantic, Norfolk, VA), Jene Nissen (U.S. Fleet Forces Command, Norfolk, VA), and Joel Bell (Naval Facilities Eng. Command Atlantic, Norfolk, VA)

Nine Marine Autonomous Recording Units (MARU's) were deployed in a rectangular array at a site coinciding with the United States (U.S.) Navy's planned Undersea Warfare Training Range (USWTR) approximately 60–150 km offshore Jacksonville, FL (13 September to 8 October and 3 December to 8 January, 2009–2010) at shallow, mid-depth, and deep sites (45, 183, 305 m). Data were reviewed in detail using TRITON (Wiggins, 2007). Event logs were created for each day at every site. Custom-written MATLAB scripts were used to calculate the probability of minke whale vocalization events occurring in the presence and in the absence of mid-frequency sonar. Minke whale vocalization events were completely absent in the fall deployment period, but occurred almost continuously during the winter deployment, indicating a strong seasonal pattern of occurrence. Minke whale vocalizations were detected most frequently at deep-water sites, and only at low levels ($<$0.03% of time) at shallow-water sites. Results of the probability analysis indicated a strong negative correlation to sonar. Minke whale vocalization events were greatly reduced, or completely ceased, during most days with nearly continuous sonar events during an approximate 3-day period. To our knowledge, such changes in acoustic behaviors of minke whales in relation to sonar have not been reported before.

4:45

2pABb6. Behavioral responses of California sea lions (*Zalophus californianus*) to controlled exposures of mid-frequency sonar signals. Dorian S. Houser (Dept. of Conservation and Biological Res., National Marine Mammal Foundation, 2240 Shelter Island Dr., San Diego, CA 92106, dorian.houser@nmmfoundation.org), Stephen W. Martin, and James J. Finneran (US Navy Marine Mammal Program, SSC Pacific, San Diego, CA)

Acoustic dose-response functions can be used to explore the relationship between anthropogenic noise exposure and changes in marine mammal behavior. Fifteen sea lions participated in a controlled exposure study to determine the relationship between the received sound pressure level (SPL) of a mid-frequency sonar signal (1-s duration, 3250-3450 Hz) and behavioral deviations from a trained behavior. Sea lions performed 10 control trials followed by 10 exposure trials within an open-water enclosure. Acoustic playbacks occurred once during each exposure trial when the sea lion crossed the middle of the enclosure. Received levels, ranging from 125 to 185 dB re 1 μPa (rms) SPL, were randomly assigned but were consistent across all trials for each individual. Blind scoring of behavioral responses was performed for all trials. A canonical correlation analysis indicated that cessation of the trained behavior, haul-out, a change in respiration rate, and prolonged submergence were reliable response indicators. Sea lions showed both an increased responsiveness and severity of response with increasing received SPL. No habituation to repeated exposures was observed, but age was a significant factor affecting the dose-response relationship. Response patterns and factors affecting behavioral responses were different from those observed in bottlenose dolphins and are indicative of species-specific sensitivities.

5:00

2pABb7. The characteristics of boat noise in marine mammal habitats. Christine Erbe (Ctr. for Marine Sci. & Technol., Curtin Univ., Kent St., Bentley, WA 6102, Australia, c.erbe@curtin.edu.au)

Environmental management of underwater noise tends to focus on sources related to offshore oil and gas exploration and navy operations. However, in many, specifically near-shore habitats, boat noise prevails, yet remains largely unregulated. Underwater noise of personal watercraft (jet skis) was recorded in Queensland, and consisted of broadband energy between 100 Hz and 10 kHz attributed to the vibrating bubble cloud generated by the jet stream, overlain with frequency-modulated tonals corresponding to impeller blade rates and harmonics. Broadband monopole source levels were 149, 137, and 122dB re 1 μPa @ 1m (5th, 50th, and 95th percentiles). Underwater noise of zodiacs (inflatable boats with outboard motors) operated by whale-watching companies was recorded in southern British Columbia. At slow cruising speeds (10 km/h), underwater noise peaked between 50 and 300 Hz; at fast traveling speeds (55 km/h), underwater noise peaked between 100 and 3000 Hz, exhibiting strong propeller blade rate tonals at all speeds. Broadband source levels increased with speed from 126 to 170dB re 1 μPa @ 1m according to SL = 107 + 32 log10(-speed/km/h). Even though noise levels from jetskis are lower than those of propeller-driven boats, it is not necessarily the broadband source level that correlates with the bioacoustic impact on marine fauna.

5:15

2pABb8. Complex masking scenarios in Arctic environments. Jillian Sills (Ocean Sci., Univ. of California at Santa Cruz, 100 Shaffer Rd., Santa Cruz, CA 95050, jmsills@ucsc.edu), Colleen Reichmuth (Inst. of Marine Sci., Univ. of California at Santa Cruz, Santa Cruz, CA), and Brandon L. Southall (Southall Environmental Associates, Aptos, CA)

Critical ratios obtained using octave-band noise and narrowband signals provide a useful first approximation for understanding the effects of noise on hearing. When considering realistic listening scenarios, it may be necessary to examine the effects of spectrally complex, time-varying noise sources on an animal's ability to detect relevant signals. In the case of Arctic seals, the increasing prevalence of seismic exploration makes an examination of masking by impulsive sounds particularly relevant. However, the characteristics of received sounds from airgun operations vary dramatically depending on the seismic source, environmental parameters, and distance. In order to determine the potential for auditory masking by airguns, we developed a paradigm to quantify the influence of spectral and temporal variations in typical seismic noise on signal detectability. This method calls for calculation of detection probabilities for seals listening for the same signal embedded at different time windows within a background of distant airgun noise. We believe this approach will enable an experimental assessment of masking potential by impulsive noise as distance between the receiver and source is increased. Such an assessment will aid in determining the extent to which standard critical ratio data can be reasonably applied in complex masking scenarios. [Work supported by OGP-JIP.]

Session 2pBA

Biomedical Acoustics: Nanobubbles, Nanoparticles, and Nanodroplets for Biomedical Acoustics Applications

Jonathan Mamou, Cochair
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Invited Papers

1:15

2pBA1. Ultrasound-targeted delivery of systemically administered therapeutic nanoparticles. Kelsie Timbie, Caitlin Burke (Biomedical Eng., Univ. of Virginia, Box 800759, Health System, Charlottesville, VA 22908, rprice@virginia.edu), Elizabeth Nance, Graeme Woodworth (Ctr. for Nanomedicine, Johns Hopkins Univ., Baltimore, MD), Grady W. Miller (Biomedical Eng., Univ. of Virginia, Charlottesville, VA), Justin Hanes (Ctr. for Nanomedicine, Johns Hopkins Univ., Baltimore, MD), and Richard J. Price (Biomedical Eng., Univ. of Virginia, Charlottesville, VA)

The ultrasound (US)-targeted delivery of systemically administered drug and gene-bearing nanoparticles has emerged to become a robust area of investigation with clear clinical potential. Such approaches typically entail the concurrent injection of contrast agent microbubbles (MBs) and nanoparticles, followed by the application of US to the region of interest. US-activated MBs disrupt the surrounding microvessel, permitting nanoparticle delivery with precise spatial localization. Our group has previously shown that US-targeted nanoparticle delivery can amplify collateral artery growth, that the binding of nanoparticles to MBs enhances nanoparticle delivery, that non-viral gene nanocarrier transfection is dependent on both MB diameter and US pressure, and that solid tumor growth can be controlled by the US-targeted delivery of 5 FU nanoparticles. More recent studies center on developing MRI-guided focused ultrasound (FUS) for nanoparticle delivery across the blood brain-barrier (BBB), which is the foremost impediment to drug treatment for most central nervous system diseases. Importantly, densely PEGylated nanoparticles that have been specifically designed to penetrate brain tissue (i.e., brain-penetrating nanoparticles, BPNs) are capable of crossing the BBB after it is opened with FUS and MBs, thereby supporting the use of gene- and drug-bearing BPNs in combination with MR-guided FUS in treating disorders and pathologies of the CNS.

1:35

2pBA2. Ultrasound-mediated stimulation of nanoparticles for therapeutic applications. Tyrone M. Porter (Mech. Eng., Boston Univ., 110 Cummington St., Boston, MA 02215, tmp@bu.edu)

Ultrasound has the unique capability to change the pressure and temperature of tissue in a noninvasive and highly localized manner. Consequently, ultrasound is an attractive tool for enhancing the delivery of drugs to targeted cells and tissues. This talk will review a variety of stimuli-responsive drug carriers (i.e., liposomes, nanoparticles, emulsions, etc.) and the role of ultrasound in facilitating transport and drug delivery.

1:55

2pBA3. Low boiling point perfluorocarbon nanodroplets for biomedical imaging and therapeutics. Paul S. Sheeran (Biomedical Eng., Univ. of North Carolina at Chapel Hill and North Carolina State Univ., Chapel Hill, NC), Terry O. Matsunaga (Radiology, Univ. of Arizona, Tucson, AZ), and Paul A. Dayton (Biomedical Eng., Univ. of North Carolina at Chapel Hill and North Carolina State Univ.,Chapel Hill, NC 27599, padayton@bme.unc.edu)

Perfluorocarbon (PFC) droplets have been examined for biomedical applications for more than a decade. Of particular interest are “phase change” PFC droplets, which can convert from liquid to gaseous phase with an acoustic energy input. In liquid form, these droplets demonstrate stability not capable with microbubbles, but in gaseous form the resultant bubble provides the cavitation source that is crucial for ultrasound imaging or therapeutic mediation. Traditionally, phase-change droplets have utilized PFCs with boiling points close to body temperature, so that they could be readily activated with acoustic energy input. However, the desire reduce these droplets to hundred-nanometer sizes in order to reach the extravascular space has made PFC selection more complicated. Specifically, the influence of surface tension on droplets that are significantly smaller than a micron elevates the boiling point past that of the bulk fluid, making hundred-nanometer sized droplets challengingly difficult to vaporize with diagnostic ultrasound energy levels. Our solution has been a unique approach to condense low boiling point gaseous perfluorocarbons, creating a population of meta-stable PFC nanodroplets. The resulting nanodroplets exhibit stability, yet have very low energy activation requirements. We discuss advantages and disadvantages of this new formulation, and demonstrate applications in biomedical imaging and therapeutics.

2:15

2pBA4. Entropy imaging and nanostructures. Michael Hughes (School of Medicine, Washington Univ., 1632 Ridge Bend Dr., St Louis, MO 63108, mshatctrain@gmail.com), John E. McCarthy (Dept. of Mathematics, Washington Univ., St Louis, MO), Jon N. Marsh, and Samuel A. Wickline (School of Medicine, Washington Univ., St Louis, MO)

Virtually all imaging devices today function by collecting either electromagnetic or acoustic waves and using the energy carried by these waves to determine pixel values to build up what is basically an "energy" picture. However, waves also carry information and this can also be used to determine the pixel values in an image. We have employed several measures of information all of which are based on different forms of entropy. Numerous published studies have demonstrated the advantages of entropy, or "information," imaging over conventional methods in materials characterization and medical imaging. Moreover, the technique is robust as these results were obtained using a variety of imaging systems having different frequency ranges and transducers. Similar results have also been obtained using microwaves. We will present the results of several entropy-imaging-based *in vivo* studies. The first of these were based on a non-gaseous liquid-filled nanoparticle contrast agent used to image tumors. We will also present a study of therapy monitoring of muscular dystrophy conducted without contrast. In both studies entropy images were able to detect changes in backscattered ultrasonic signals that were not detectable using conventional techniques.

2:35

2pBA5. Characterization of liquid-filled nanoparticles for detection and drug-delivery in tumors. Nicolas Taulier, Thomas Payen, Sara Jafari (Parametric Imaging Lab, CNRS - Univ. Pierre and Marie Curie, 15 rue de l'ecole de medecine, Paris 75006, France, lori.bridal@upmc.fr), Jonathan Mamou (F.L. Lizzi Ctr. for Biomedical Eng., Riverside Res., New York, NY), Fattal Elias, Nicolas Tsapis (Institut Galien Paris-Sud, UMR 8612, LabEx LERMIT, Univ Paris-Sud and CNRS, Châtenay-Malabry, France), and Lori Bridal (Parametric Imaging Lab, CNRS - Univ. Pierre and Marie Curie, Paris, France)

Liquid-filled nanoparticles provide long half-life circulation favoring accumulation in the interstitial space of tumors through enhanced permeation and retention. However, these relatively incompressible nanoparticles are much less echogenic than those with a gas-core. We have worked to characterize and optimize the acoustic response of PLGA-shelled, perfluorooctyl bromide (PFOB) core nanoparticles (radii 70 to 200 nm ; shell-thickness-to-radius ratio 0.25 to 1). Acoustic response (attenuation coefficient, ultrasonic velocity, and relative backscattered intensity) was explored *in vitro* for ranges of concentration, shell-thickness, acoustic pressures and pulse durations (20 to 40 MHz). Modification of the surface chemistry of the polymeric shell with fluorescent, pegylated, or/and biotinylated phospholipids was not associated with apparent response modification. Successful incorporation of paclitaxel in the shell has been achieved but currently the thick-shells of these nanoparticles impede ultrasound-triggered delivery. Although work remains to better adapt liquid-filled nanoparticles for detection and delivery, trials demonstrating nanoparticle detection *in vivo* in mice indicate their potential for accumulation, and detection in tumors.

2:55

2pBA6. Ultrasound-mediated delivery of bioactive nanobubbles to vascular tissue. Jonathan T. Sutton, Jason L. Raymond (College of Eng. and Appl. Sci.; Biomedical Eng., Univ. of Cincinnati, 231 Albert Sabin Way, 3940 Cardiovascular Ctr., Cincinnati, OH 45267, suttonjt@mail.uc.edu), Michael C. Verleye (College of Eng.; Dept. of Chemical Eng., Notre Dame Univ., Notre Dame, IN), Gail J. Pyne-Geithman (College of Medicine; Dept. of Neurosurgery, Univ. of Cincinnati, Cincinnati, OH), Jack Rubinstein, and Christy K. Holland (College of Medicine; Internal Medicine, Div. of Cardiovascular Diseases, Univ. of Cincinnati, Cincinnati, OH)

Bubble liposomes (BLs) are under development for ultrasound-triggered release of a potent vasodilator within the vasculature. Nano-sized vesicles facilitate this process by enclosing bubbles to enhance ultrasound image contrast, and deliver therapeutic agents. Assessment of drug delivery in living tissue allows for mechanistic pathways to be revealed. In this study, we used a novel *ex vivo* model to assess the vascular effects of ultrasound-mediated delivery of a bioactive gas-nitric oxide (NO)-from nanobubble liposomes. Porcine carotid arteries were excised post-mortem and mounted in physiologic buffer. Vascular tone was assessed in real time by coupling the artery to an isometric force transducer. NO-loaded BLs were infused into the lumen of the artery, which was exposed to 1-MHz pulsed ultrasound, while acoustic cavitation emissions were monitored. Changes in vascular tone were concurrently measured and compared to control and sham NO exposures. Our results demonstrate that ultrasound-triggered NO release from BLs induces potent vasorelaxation within porcine carotid arteries. This approach is a valuable mechanistic tool to assess the bioeffects that NO elicits within the vasculature upon release from BLs exposed to 1-MHz ultrasound.

Contributed Papers

3:15

2pBA7. Acoustic droplet vaporization is initiated by superharmonic focusing. Oleksandr Shpak (Phys. of Fluids, Univ. of Twente, Witbreuksweg 379-404, Enschede 7522ZA, Netherlands, o.shpak@utwente.nl), Martin Verweij (Dept. of Imaging Sci. and Technol., Delft Univ. of Technol., Delft, Netherlands), Rik Vos, Nico de Jong (Biomedical Eng., Erasmus MC, Rotterdam, Netherlands), Detlef Lohse, and Michel Versluis (Phys. of Fluids, Univ. of Twente, Enschede, Netherlands)

Acoustically sensitive emulsion droplets composed of a liquid perfluorocarbon have the potential to be a highly efficient system for local drug delivery, embolotherapy or for tumor imaging. The physical mechanisms underlying the acoustic activation of these phase-change emulsions into a bubbly dispersion, termed acoustic droplet vaporization, have not been well understood. The droplets have a very high activation threshold, its frequency dependence does not comply with homogeneous nucleation theory and focusing spots have been observed. We showed that acoustic droplet vaporization is initiated by a combination of two phenomena: highly nonlinear distortion of the acoustic wave before it hits the droplet, and focusing of the distorted wave by the droplet itself. At high excitation pressures, nonlinear distortion causes significant superharmonics with wavelengths below the diameter of the droplet. Because these superharmonics strongly contribute to the focusing effect, the mechanism also explains pressure thresholding effects. In an accompanying paper, mathematical modeling aspects are presented. A proposed model is validated with experimental data captured with an ultra high-speed camera on the positions of the nucleation spots.

Moreover, the presented mechanism explains the hitherto counterintuitive dependence of the nucleation threshold on the ultrasound frequency.

3:30

2pBA8. Subwavelength droplets in nonlinear ultrasound fields: Simulation of focusing effects. Martin D. Verweij (Acoust. Wavefield Imaging, Delft Univ. of Technol., P.O. Box 5046, Delft 2600GA, Netherlands, m.d.verweij@tudelft.nl), Oleksandr Sphak (Phys. of Fluids Group, Univ. of Twente, Enschede, Netherlands), Hendrik J. Vos, Nico de Jong (Biomedical Eng., Erasmus MC, Rotterdam, Netherlands), Detlef Lohse, and Michel Versluis (Phys. of Fluids Group, Univ. of Twente, Enschede, Netherlands)

Ultrasound can trigger the evaporation of tiny droplets of emulsified, superheated fluids like perfluorocarbon. This acoustic droplet vaporization (ADV) effect is important because of its potential medical applications. For example, drug-loaded nanodroplets can penetrate the vessel wall and subsequently release their therapeutic load through acoustic activation. Until now, the medical application of ADV has been limited by lack of understanding of the acoustic activation mechanism. In an accompanying paper, a mechanism is proposed that can fully explain the experimentally observed phenomena such as a frequency dependent pressure threshold. The mechanism involves focusing of higher harmonics of the nonlinear activation field. In the current presentation it will be explained how the focusing effect is simulated. A particular problem here is that typical droplet sizes are in the order of micrometers, while typical wavelength sizes are in the order of hundreds of micrometers. This difference in scale will render the traditional analytic solution numerically useless. The problem is avoided by using appropriate expansions of the functions involved. Numerical results are presented that show a focusing effect inside the droplet for a range of incident harmonics. These results have enabled the demonstration of the mechanism in the accompanying paper.

3:45

2pBA9. Characterization of nanometric ultrasound contrast agents with a liquid perfluorocarbon core. Ksenia Astafyeva (Lab. of Parametric Imaging, Pierre & Marie Curie Univ., 15 rue de l'ecole de medecine, Paris 75013, France, ksenia.astafyeva@gmail.com), Jean-Marc Conoir, Matthieu Guédra (Inst. of Jean Le Rond d'Alembert, Paris, France), Elias Fattal (Institut Galien, Univ. Paris South, Chatenay-Malabri, France), Christine Pepin, Ange Polidori (Univ. of Avignon, Avignon, France), Nicolas Taulier (Lab. of Parametric Imaging, Pierre & Marie Curie Univ., Paris, France), Jean-Louis Thomas (Inst. of NanoSci., Paris, France), Nicolas Tsapis (Institut Galien, Univ. Paris South, Chatenay-Malabri, France), Tony Valier-Brasier (Inst. of Jean Le Rond d'Alembert, Paris, France), and Wladimir Urbach (Lab. of Parametric Imaging, Pierre & Marie Curie Univ., Paris, France)

In this work, we propose a new type of nanometric ultrasound contrast agents (nUCA) with a liquid core and we model their acoustic propagation through their dilute solutions. These capsules have a shell made of a biocompatible polymer or fluorinated surfactants and a liquid perfluorocarbon core to undergo a high lifetime. The capsules are small enough (from 100 nm to 1 μm) to pass through the tumor endothelium, and they remain stable *in vitro* for several months. Ultrasound attenuation and speed of sound measurements through dilute suspensions of nUCA were carried out from 3 to 90 MHz at various temperatures and concentrations. The acoustic propagation was modeled by combining (i) a dilatational mode taking into account the radial oscillations of the capsules, and (ii) a translational mode of oscillations induced by viscoinertial interaction with the continuous phase. The model makes possible to fit with good accuracy the experiments using values compatible with literature data. Moreover, it reveals information about unknown parameters of the shell: for instance, the viscoelastic shell has to be described as a Maxwell rheological medium. [This work was supported by Emergence-UPMC program.]

4:00–4:15 Break

4:15

2pBA10. Ultrasonic propagation in suspensions of encapsulated compressible nanoparticles. Matthieu Guédra (Institut Jean le Rond d'Alembert, Université Pierre et Marie Curie - Paris 6, 4 Pl. Jussieu, Paris 75252, France, matthieu.guedra@dalembert.upmc.fr), Ksenia Astafyeva (Laboratoire d'Imagerie Paramétrique, Université Pierre et Marie Curie - Paris 6, Paris, France), Jean-Marc Conoir, François Coulouvrat (Institut Jean le Rond d'Alembert, Université Pierre et Marie Curie - Paris 6, Paris, France), Nicolas Taulier (Laboratoire d'Imagerie Paramétrique, Université Pierre et Marie Curie - Paris 6, Paris, France), Jean-Louis Thomas (Institut des NanoSci. de Paris, Université Pierre et Marie Curie - Paris 6, Paris, France), Wladimir Urbach (Laboratoire d'Imagerie Paramétrique & Laboratoire de Physique Statistique de l'ENS, Université Pierre et Marie Curie - Paris 6, Paris, France), and Tony Valier-Brasier (Institut Jean le Rond d'Alembert, Université Pierre et Marie Curie - Paris 6, Paris, France)

Dispersion and absorption are examined for dilute suspensions of encapsulated droplets of nanometric size, with typical radii around 100 nm. This new generation of contrast agents is designed for targeted delivery of drugs. Compared to standard contrast agents used for imaging, particles are of smaller size to pass the endothelial barrier, their shell made up of biocompatible material is stiffer to undergo a longer time life and they have a liquid (PFC) instead of a gaseous core. Ultrasound propagation of these suspensions is modeled by combining (i) a dilatational mode of oscillation assuming a compressible shell with a visco-elastic behavior of Kelvin-Voigt or Maxwell type (relaxation), (ii) a translational mode of oscillation induced by visco-inertial interaction with the ambient fluid, and (iii) polydispersion in terms of radius and shell thickness. Influence of the various effects will be examined. Experimental measurements of the dispersion and absorption properties of nanodroplets solutions over the 1–100 MHz frequency range are performed for various temperatures and concentrations. They allow to fit with good accuracy the model properties and estimate some unknown mechanical properties of the shell of the nanodroplets. [Work supported by programme Emergence-UPMC—project NACUNAT and by Inserm/Plan Cancer—project NABUCCO.]

4:30

2pBA11. Dual perfluorocarbon nanodroplets enhance high intensity focused ultrasound heating and extend therapeutic window *in vivo*. Linsey C. Phillips, Paul S. Sheeran, Connor Puett (Joint Dept. of Biomedical Eng., UNC at Chapel Hill, and NC State Univ., CB 7575, Univ. of North Carolina, Chapel Hill, NC 27599, linsey@email.unc.edu), Kelsie F. Timbie, Richard J. Price (Biomedical Eng., Univ. of Virginia, Charlottesville, VA), G. Wilson Miller (Radiology, Univ. of Virginia, Charlottesville, VA), and Paul A. Dayton (Joint Dept. of Biomedical Eng., UNC at Chapel Hill, and NC State Univ., Chapel Hill, NC)

Perfluorocarbon microbubbles are known to enhance high intensity focused ultrasound (HIFU) ablation by cavitation. However, they can result in superficial skin heating, minimizing their clinical translation. Perfluorocarbon nanodroplets activate only at the higher pressures present at the acoustic focus. We hypothesized that a mixed perfluorocarbon nanodroplet formulation would minimize surface heating while still enhancing ablation. Tissue-mimicking phantoms containing microbubbles or nanodroplets were sonicated (1 MHz, 15 W, 60 s) to assess heating and lesion formation *in vitro*. Microbubbles or nanodroplets were injected into rats ($n = 3$) and HIFU (1 MHz, 15 W, 15 s) was focused into each liver while under MRI guidance. Temperature throughout the liver was tracked by MR thermometry. *In vitro*, microbubbles caused excess surface heating during HIFU, whereas nanodroplets did not. *In vivo*, microbubbles typically circulate for less than 15 min. In comparison, the nanodroplets remained viable in circulation for at least 96 min. HIFU lesions of consistent volume were produced during this time and reached the same maximal temperature rise (Δ550 °C). In the absence of nanodroplets or microbubbles, significantly greater power (25 W) and twice as much time (30 s) was required to generate an ablation lesion in the liver. These results demonstrate the ability of nanodroplets to more safely shorten ablation procedures.

4:45

2pBA12. Use of micro- and nano-sized inertial cavitation nuclei to trigger and map drug release from cavitation-sensitive liposomes. Susan M. Graham, Rachel S. Myers, James Choi, Miriam Bazan-Peregrino (Inst. of Biomedical Eng., Dept. of Eng. Sci., Univ. of Oxford, ORCRB, Oxford OX3 7DQ, United Kingdom, susan.graham@eng.ox.ac.uk), Leonard Seymour (Dept. of Oncology, Univ. of Oxford, Oxford, United Kingdom), Robert Carlisle, and Constantin C. Coussios (Inst. of Biomedical Eng., Dept. of Eng. Sci., Univ. of Oxford, Oxford, United Kingdom)

Encapsulation of cytotoxic drugs into liposomes enhances pharmacokinetics and improves passive accumulation in tumors. However, stable liposomes have limited drug release, and thus action, at the target site. This inefficient and unpredictable drug release is compounded by a lack of low-cost, non-invasive methods to map release in real time. We present a new liposomal vehicle that is exclusively triggered by inertial cavitation. Ultrasound exposure of these liposomes in the absence of SonoVue® provided no increase in drug release, whilst with SonoVue® at inertial cavitation pressure levels a substantial (30%) and significant ($p < 0.001$) increase was observed *in vitro*. A 16-fold increase in the level of drug release within tumors was similarly observed in the presence of inertial cavitation following intravenous delivery. Passive acoustic mapping of inertial cavitation sources during delivery was also found to correlate strongly with the presence of release. However, variability in tumor perfusion indicated that uneven distribution of micron-sized SonoVue® may limit this approach. Nano-scale cavitation nuclei, which may more readily co-localize with 140 nm liposomes, were thus developed and showed similar cavitation energies to SonoVue® *in vitro*. These nano-nuclei may ultimately provide a more reliable and uniform way to trigger drug release *in vivo*.

5:00

2pBA13. Time lapse observation of phase change nano droplet after vaporization stimulated by ultrasound. Kenji Takehara, Takashi Azuma (BioEng., The Univ. of Tokyo, 7-3-1 Hongou, Bunkyou-ku, Tokyo 113-8656, Japan, k.take531@gmail.com), Kiyoshi Yoshinaka (National Inst. of Adv. Industrial Sci. and Technol., Tsukuba, Ibaraki, Japan), Satoshi Yamaguchi (Chemistry and BioTechnol., The Univ. of Tokyo, Bunkyou-ku, Tokyo, Japan), Miyuki Maezawa (Olympus Corp., Shinjuku-ku, Tokyo, Japan), Ichirou Sakuma (Precision Eng., The Univ. of Tokyo, Bunkyou-ku, Tokyo, Japan), Teruyuki Nagamune (Chemistry and BioTechnol., The Univ. of Tokyo, Bunkyou-ku, Tokyo, Japan), Shu Takagi, and Yoichirou Matsumoto (Mech. Eng., The Univ. of Tokyo, Bunkyou-ku, Tokyo, Japan)

Small enough to permeate through tumor blood vessel, and can be detect by ultrasound, phase change nano droplet (PCND) have been studied as contrast agents and therapeutic sensitizer for career. To investigate performance for these purpose, we investigated physical behavior of PCND, especially a lifetime of microbubbles generated by ultrasound stimulation. To investigate bubble's behavior after phase change, we observed a time-lapse change of bubbles population with following method. Focused transducer with a frequency of 3.3 MHz was placed in a water bath of 37 degrees. At the focal point, polyacrylamide gel including PCND was placed. Focused hydrophone was placed perpendicularly to the direction of ultrasound propagation. Two kind of ultrasound pulse wave was used; phase change pulse at the beginning and observation pulse at every 500 μs. The amplitude of scattering signal (SS) reflects the sum of scattering cross-section of bubbles. The time lapse observation of PCND after phase change showed two kinds of behavior, quick and slow decay of scattering signal from the bubbles. The unique increase of SS from the phase-changed bubbles in the monitoring phase was observed. We improved the experimental setup to measure bubble's population properly from various directions using 1-D array transducer and will present the result.

5:15

2pBA14. Detection of unique acoustic signatures for phase-change contrast agents used in medical imaging and therapy. Paul S. Sheeran, Karl H. Martin (Joint Dept. of Biomedical Eng., Univ. of North Carolina and North Carolina State Univ., 10 DuxforDC., Durham, NC 27703, pssheeran@gmail.com), Jordan N. Hjelmquist (Dept. of Biomedical Eng., North Carolina State Univ., Raleigh, NC), Terry O. Matsunaga (Dept. of Medical Imaging, Univ. of Arizona, Tucson, AZ), and Paul A. Dayton (Joint Dept. of Biomedical Eng., Univ. of North Carolina and North Carolina State Univ., Chapel Hill, NC)

Phase-change contrast agents (PCCAs) provide a dynamic platform to approach problems in medical ultrasound (US). Upon US-mediated activation, the liquid core vaporizes and expands to produce a gas bubble ideal for US imaging and therapy. In this study, we demonstrate through underlying theory, high-speed microscopy, and US interrogation that PCCAs composed of highly volatile perfluorocarbons (PFCs) exhibit unique acoustic behavior that can be differentiated from tissue and standard microbubble contrast agents. Experimental results show that when activated with short pulses PCCAs will over-expand ($6.3\times$ to $7.7\times$ the droplet diameter, PFC dependent) due to momentum of expansion, and undergo unforced, under-damped radial oscillation while settling to a final bubble diameter ($5.1\times$ to $5.5\times$ the droplet diameter). Oscillation frequency is inversely related to droplet size—near 100 kHz for droplets ≥ 4 μm in diameter, and 2.5 MHz for droplets near 500 nm. Results from *in vitro* vessel phantoms using confocal piston transducers with an "activate high" (8 MHz, 2 cycles), "listen low" (1 MHz) scheme show that droplet-specific signals can be detected in both time and frequency domain, and that the magnitude of the acoustic "signature" increases with PFC volatility. These signatures may aid in development of droplet-specific detection techniques.

5:30

2pBA15. Study of mechanism of sonoporation using lipid bilayer and surface-modified microbubble. Kodai Hirose, Takashi Azuma (Mech. Eng., The Univ. of Tokyo, 7-3-1, Hongo, Bunkyo-ku, Tokyo 113-8656, Japan, khirose@fel.t.u-tokyo.ac.jp), Kiyoshi Yoshinaka (National Inst. of Adv. Industrial Sci. and Technol., Tsukuba, Japan), Akira Sasaki, Shu Takagi, and Yoichiro Matsumoto (Mech. Eng., The Univ. of Tokyo, Tokyo, Japan)

Microbubble enhanced sonoporation is one of gene therapies and expected to be safe, less invasive, and controllability of treatment area, but its induction rate is very low and its mechanism remains to be explained. The objective is to analyze the mechanism and obtain an optimal design. Because microbubble density distribution was unstable during sonication and ultrasound intensity on the cell surface was affected by distribution change through attenuation change during propagation, microbubble distribution should be controlled and localized only near the membrane to realize high reproducibility. An artificial lipid bilayer modified by biotin-avidin to bind microbubbles was used for this purpose. We have three steps: build lipid bilayer, binding with microbubbles, and introduction of sonication and observation system. We built the lipid bilayer with a diameter of 1 mm using Black Lipid Membrane method. With a capacitance measurement with impedance analyzer (NF, ZM2375), the lipid bilayer thickness was confirmed and its duration is more than an hour. Then, we introduced the sonication and observation system to the lipid bilayer, and we are observing a behavior of the lipid bilayer under irradiation of ultrasound. Afterward we will observe that of a lipid bilayer modified of microbubbles.

5:45

2pBA16. Optimization of acoustic parameters and nanodroplet concentration for spatially controlled, reduced energy high intensity focused ultrasound ablation. Andrew C. Puett (Biomedical Eng., UNC, 6317 S Bradley Overlook, Wilmington, NC 28403, connorpuett@gmail.com), Linsey C. Phillips, Paul S. Sheeran, and Paul A. Dayton (Biomedical Eng., UNC, Chapel Hill, NC)

Background: Perfluorocarbon (PFC)-nanodroplets (ND) provide cavitation sites when vaporized to microbubbles by acoustic energy and lower the power required to ablate tissue by high intensity focused ultrasound (HIFU). However, control over ablation can be problematic. This study explored vaporization, ablation, and PFC-ND concentration *in vitro* to optimize the acoustic pressure (intensity) and insonation time required for spatially controlled HIFU enhancement. Methods: HIFU (continuous wave; 1MHz; 5–20 s; 2–4 MPa) was applied to albumin-acrylamide gels containing PFC-agent (1:1 mix of volatile decafluorobutane and more stable dodecafluoropentane at 10^5-10^7 ND/mL). Controlled ablation was defined as the production of cigar-shaped lesions centered at the acoustic focus. Results: Vaporization field change from "cigar" to "tadpole" began at 5×10^5, 2.5×10^6, and 1×10^7 ND/mL using 4, 3, and 2 MPa, respectively. The volumes of the microbubble clouds (8–200 mm^3) and ablation lesions (1–135 mm^3) within them were dependent on acoustic intensity, insonation time, and PFC-ND concentration. Conclusions: Ablation within microbubble clouds of predictable size, shape, and location can be generated in gels containing PFC-ND using intensities $\leq$650 W/cm^2. Also, pressures and insonation times can be selected to achieve an ablation lesion of desired size for a given PFC-ND concentration. Demonstrating control is an important step toward developing a useful clinical tool.

6:00

2pBA17. Design and characterization of biocompatible perfluorocarbon nanodroplets for theragnostic application. Lucie Somaglino, Ksenia Astafyeva (Laboratoire Imagerie Paramétrique, UMR 7623, CNRS-UPMC, 15, rue de l'école de médecine, EscA 2ème ét., Paris 75005, France, lucie.somaglino@yahoo.fr), Stéphane Desgranges, Ange Polidori, Christiane Contino-Pepin (Institut des Biomolécules Max Mousseron, UMR 5247, CNRS-Université d'Avignon et des Pays de Vaucluse, Avignon, France), Wladimir Urbach, and Nicolas Taulier (Laboratoire Imagerie Paramétrique, UMR 7623, CNRS-UPMC, Paris, France)

We have developed stable emulsions ($\geq$ 3 months) made of nanodroplets (nD) of perfluorocarbon (PFC) dispersed in water to serve as theragnostic agent. nD are stabilized by in house fluorinated surfactants, named FTAC, which chemical structure can be modified to tune their properties. We have characterized nD size distributions (mean diameters from 200 to 600 nm), density, adiabatic compressibility, interfacial tension. US properties of the emulsions have been investigated such as attenuation. Lastly, ultrasonic signals backscattered by nanoemulsions were studied and compared with water to extract signal to noise ratio (SNR), by emitting single negative pulses at $\approx$40 MHz. At similar mean/mode diameters, we showed a strong dependence in SNR values (i) with size distribution, altered by the nature of the surfactant or by a centrifugation/filtration process, (ii) with core nature, and (iii) with nD volume fraction. Besides, hydrophobic drugs such as a thalidomide derivative (with anti-angiogenic properties), have been encapsulated by addition of 10% of triacetin in the nD core. So as to study drug release from nD, a dedicated setup of US cavitation was designed. Cavitation was generated in controlled conditions using a 1 MHz focused transducer (US bursts: 12-22 MPa peak) in a thin wall container immersed in water which temperature and degassed level were kept constant. The generation of cavitation in nD solutions resulted in a strong and reproducible SNR decrease.

TUESDAY AFTERNOON, 3 DECEMBER 2013 — MASON, 1:30 P.M. TO 3:10 P.M.

Session 2pEA

Engineering Acoustics, Acoustical Oceanography, and Animal Bioacoustics: Bearing Measurement Methods for Small Wideband Sonars

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

Chair's Introduction—1:30

Invited Papers

1:35

2pEA1. Use of Bessel side lobe modulation of frequency modulated pulse bearing estimation. Kenneth M. Walsh (K + M Engineering Ltd., 51 Bayberry Ln., Middletown, RI 02842, kwalsh4@mindspring.com)

The beam pattern of a circular transducer is a set of frequency dependent Bessel functions. By using a broad band FM pulse and a 1–3 composite transducer, it appears that the modulation in amplitude and phase due to a reception of the high frequency components can be used to estimate the echo's angle off the transducer axis. The 1–3 composite transducer has a simple amplitude and phase structure.

1:55

2pEA2. Modeling of bio-inspired broadband sonar for high-resolution angular imaging. Jason E. Gaudette (Adv. Acoust. Systems Div., NUWC Div. Newport, 1176 Howell St., B1371/3, Newport, RI, jason.e.gaudette@navy.mil) and James A. Simmons (Dept. of Neurosci., Brown Univ., Providence, RI)

Echolocating mammals perceive images of targets with hyper-resolution and navigate seamlessly through obstacles in complex acoustic environments. The biological solution to imaging with sound is vastly different from man-made sonar. The most prominent difference is that instead of imaging with narrow beams, bats ensonify a large spatial region and exploit broadband echo information to acoustically focus with about one degree of angular resolution. Angular localization may therefore be redefined as a spectral pattern matching problem. By imaging with wider beams, this remarkable performance requires only a single broadband transmitter and two receive elements. Our computational modeling work has led to new insight into the salient spatial information encoded by the bat's auditory system. Although theoretically not required, spatial localization performance increases with the aid of highly complex baffle structures such as those found in biological sonar. Replicating these bio-inspired baffle structures and acoustic processing techniques in man-made systems can reduce sonar array aperture requirements by a factor of 100 or more for a variety of both aerial and underwater acoustic sensing applications. Recent modeling results are presented along with progress toward the design of a compact bio-inspired sonar system for high-resolution imaging. [Work supported by ONR and NUWC Newport.]

2:15

2pEA3. Front-looking and side-looking receiving beams for biosonar imaging and flight guidance. James A. Simmons (Dept. of Neurosci., Brown Univ., Providence, RI 02912, james_simmons@brown.edu)

The biosonar broadcasts of big brown bats are very broadly beamed over $\pm 60^\circ$-120°, thus ensonifying virtually all of the objects in the surrounding sonar scene. However, masking-release results indicate that off-side clutter is rejected outside of a significantly narrower beam of $\pm 15^\circ$. Recent experiments establish that clutter interference with target imaging is prevented by sensing the lower amplitude of FM2 relative to FM1 in clutter echoes and forming poorly-focused images of the clutter. Removal of clutter from perception does not address the need to guide flight in complex surroundings bounded by clutter such as vegetation. A separate mechanism used for guidance may be located in the midbrain, where exclusively contralaterally sensitive neurons respond to target range and direction, the ingredients for following the flow of surrounding objects as they slide past the flying bat. A review of neurophysiological results from different laboratories suggests that the big brown bat's biosonar system may contain different receiver pathways—a forward-looking system for target imaging and classification, and a side-looking system for following the surrounding clutter. [Work supported by ONR and NSF.]

2:35

2pEA4. Improving direction-sensing by multibeam sonar. Gerard Llort-Pujol (Image and Information Processing Dept., Institut Mines Télécom - Télécom Bretagne, Brest, France), Kenneth G. Foote (Woods Hole Oceanographic Inst., 98 Water St., Woods Hole, MA 02543, kfoote@whoi.edu), and Christophe Sintes (Image and Information Processing Dept., Institut Mines Télécom - Télécom Bretagne, Brest, France)

The complexity of multibeam sonar systems makes their beamforming susceptible to amplitude and phase distortion, e.g., due to environmental changes. A form of calibration is usually performed *in situ* over a flat area to ensure flatness in the resulting sonar image. However, lack of detailed knowledge of individual channel performance prevents application of an amplitude-weighting function such as the Blackman or Hamming type, which could otherwise be used to reduce the impact of sidelobes without damaging signal quality. Two radically different solutions are proposed: application of the Vernier principle [G. Llort-Pujol, Oceans 2006 MTS/IEEE Conf. Proc. (Quebec City, Canada, 2008); C. Sintes *et al*., Oceans 2011 MTS/IEEE Conf. Proc. (Waikoloa, HI, 2011)] to interferometry performed at low grazing angles, and performance of a standard-target calibration [K. G. Foote *et al*., J. Acoust. Soc. Am. **117**, 2013 (2005)] to measure the two-way sensitivity of individual channels directly. The two methods, which are also applicable to sidescan sonar, are elaborated.

Contributed Paper

2:55

2pEA5. Bearing measurements using a compact wideband sensor by forming three dipoles. David A. Brown (BTech Acoustics LLC, ATMC, 151 Martine St., Fall River, MA 02723, dbAcoustics@cox.net)

This paper summaries the development of a compact sensor for determining the bearing angle of an incoming signal using trinary acoustic dipoles. The approach is a kin to the using orthogonal dipoles an omni-reference, which is common in DIFAR sonobuoy. Symmetrically processing three dipoles can offer advantages in sensitivity and extended bandwidth. Analytical and experimental results on the TRI-phase Bearing Estimator, or TRIBE, are presented.

Session 2pED

Education in Acoustics: Take 5's

Jack Dostal, Chair
Phys., Wake Forest Univ., P.O. Box 7507, Winston-Salem, NC 27109

For a Take-Five session no abstract is required. We invite you to bring your favorite acoustics teaching ideas. Choose from the following: short demonstrations, teaching devices, or videos. The intent is to share teaching ideas with your colleagues. If possible, bring a brief, descriptive handout with enough copies for distribution. Spontaneous inspirations are also welcome. You sign up at the door for a five-minute slot before the session starts. If you have more than one demo, sign-up for two consecutive slots.

TUESDAY AFTERNOON, 3 DECEMBER 2013 CONTINENTAL 1, 1:00 P.M. TO 3:50 P.M.

Session 2pMU

Musical Acoustics and Signal Processing in Acoustics: Digital Musical Instruments

Edgar J. Berdahl, Chair
Music, Louisiana State Univ., 102 New Music Bldg., Baton Rouge, LA 70803

Invited Papers

1:00

2pMU1. Interaction design and the active experience of music. David Wessel (Music CNMAT, Univ. of California Berkeley, 1750 Arch St., Berkeley, CA 94709, davidwessel@me.com)

Music search engines, play list generators, streaming audio, and portable players have taken much of the focus of music technology. The emphasis is on delivery and experiencing music is by playback, playback while jogging or while working about the house, and sadly even while studying. In this talk in the hope of providing an antidote I will examine the role of bodily action in the experience of music and the importance of human computer interaction design in the development of computationally based musical instruments. Central are gestural interfaces and their mapping to musical material. Special emphasis will be given to designing for expression, for exploration and discovery, and to a musical practice that involves a coordinated balance of software development and daily bodily engagement with one's instrument.

1:20

2pMU2. LinnStrument and other new expressive musical controllers. Roger Linn (Roger Linn Design, 1147 Keith Ave., Berkeley, CA 94708, rl@rogerlinndesign.com)

Roger Linn will demonstrate his LinnStrument, a controller for musical performance that captures three dimensions of movement for each touch, polyphonically, in order to provide more expressive control of software music synthesis. In addition, he will show videos of other similar new instruments and compare the unique approaches taken by each designer.

1:40

2pMU3. Sound synthesis for a brain stethoscope. Chris Chafe, Juan-Pablo Caceres, and Michael Iorga (Stanford Univ., CCRMA/Music, Stanford, CA 94305, cc@ccrma.stanford.edu)

Exploratory ascultation of brain signals has been prototyped in a project involving neurologists, real-time EEG and techniques for computer-based sound synthesis. In a manner similar to using a stethoscope, the listener can manipulate the location being listened to. Sounds which are heard are sonifications of electrode signals. We present a method for exploring sounds from arrays of sensors as sounds which are useful for distinguishing brain states. The approach maps brain wave signals to modulations characteristic of human voice. Computer-synthesized voices "sing" the dynamics of wakefulness, sleep, seizures, and other states. The goal of the project is to create a recognizable inventory of such vocal "performances" and allow the user to probe source locations in the sensor array in real time.

2:00

2pMU4. Grafting acoustic instruments and signal processing: Creative control and augmented expressivity. Dan Overholt (Media Technol., Aalborg Univ. Copenhagen, A.C. Meyers Vaenge 15, Copenhagen SV. 2450, Denmark, dano@create.aau.dk)

In this study, work is presented on a hybrid acoustic/electric violin. The instrument has embedded processing that provides real-time simulation of acoustic body models using DSP techniques able to gradually transform a given body model into another, including extrapolations beyond the models to explore interesting new timbres. Models can include everything from various violin bodies to guitars, sitars with their sympathetic strings, and even physically impossible acoustic bodies. The development also presents several practical approaches to sensor augmentation and gestural playing techniques that can be applied to bowed-string and other acoustic instruments, in order to provide immediate creative control over the possibilities offered by DSP. The study has focused on augmenting the expressivity of the violin toward finding novel timbral possibilities, rather than a goal of simulating prior acoustic violins with high fidelity. The opportunity to control a virtually malleable body while playing, i.e., a model that changes reverberant resonances in response to player input, results in interesting audio effects. Other common audio effects can also be employed and simultaneously controlled via the musician's movements. For example, gestural tilting of the instrument is tracked via an embedded Inertial Measurement Unit (IMU), which can be assigned to alter parameters such as the wet/dry mix of an octave-doubler or other effect, further augmenting the expressivity of the player.

2:20

2pMU5. Saxophone fingering identification. Tamara Smyth (Music, Univ. of California San Diego, 9500 Gilman Dr., MC 0099, La Jolla, CA 92093-0099, trsmyth@ucsd.edu) and Marjan Rouhipour (Computing Sci., Simon Fraser Univ., Surrey, BC, Canada)

The focus of this work is to identify the tonehole configuration or "fingering'' applied by a player during performance, using only the signal recorded at the bell. Because of a player can use alternate fingerings/overblowing to produce a given frequency, detecting the sounding pitch only reduces the possible candidates—it does not produce a unique result. Several recordings of a professional saxophonist playing notes using all fingerings are considered, and several higher level features are explored for distinguishing between a fundamental and an overblown note. In the latter case, it is observed that during the attack portion of the note, the spectral centroid is usually lower, there is greater inharmonicity and increased pitch instability. Combining these heuristics with the detection of subharmonics has yielded excellent results in detecting overblown notes. With the possible fingerings being greatly reduced by this preprocessing, more computationally expensive statistical methods may be employed for a more accurate estimation of the actual fingering applied. To this end, the recorded sound is calibrated to that produced by a reed model coupled to a waveguide that is informed by an acoustic measurement of the player's saxophone configured with each usable fingering.

2:40

2pMU6. The Faust Synthesis Toolkit: A set of linear and nonlinear physical models for the Faust programming language. Romain Michon (Dept. of Music, Ctr. for Comput. Res. in Music and Acoust., 660 Lomita Court, Stanford, CA 94305-8180, rmichon@ccrma.stanford.edu)

The Faust Synthesis ToolKit is a set of virtual musical instruments written in the Faust programming language and based on waveguide algorithms and on modal synthesis. Most of them were inspired by instruments implemented in the Synthesis ToolKit (STK) and the program SynthBuilder. Our attention has partly been focused on the pedagogical aspect of the implemented objects. Indeed, we tried to make the Faust code of each object as optimized and as expressive as possible. Some of the instruments in the Faust-STK use nonlinear allpass filters to create interesting and new behaviors. Also, a few of them were modified in order to use gesture data to control the performance. A demonstration of this kind of use is done in the Pure Data program. Finally, the results of some performance tests of the generated C++ code are presented.

3:00

2pMU7. Drawn to sound: An audio visual musical instrument using custom electronics and magnetometer. John Granzow and Hongchan Choi (CCRMA, Stanford Univ., 660 Lomita Dr., Stanford, CA 94305, johknee5@gmail.com)

Drawings are amplified through a resonant surface and transmitted via microphone to custom software hosted on an embedded linux computer. An HMC-5883L magnetometer is used to modulate the signal according to the position of the pencil (equipped with magnetic sleeve). 3D vectors are derived from the magnetometer using a custom Arduino library. We project this vector into a 2D plane to get magnitude or distance between the sensor and the magnetized pencil as well as the heading angle. This implementation gives us 2D polar coordinates such that the position of the pencil can be used to vary the audio output. The transformation of the raw drawing sound is excited with proximity to the magnetometer due to the sensors exponentially increasing sensitivity to the magnetic field. Resulting drawings often contain both visually motivated marks as well as gestures that are made for sound such as dark scribbled regions where a desired timbre or pitch shift is repeated throughout a performance. This presentation will discuss hardware design, the implementation of our custom software and circuitry, how these components combine for a compelling performance platform as well as areas where we seek improvement.

Contributed Papers

3:20

2pMU8. The Stingray embedded acoustic instrument. Edgar J. Berdahl (Music, Louisiana State Univ., 102 New Music Bldg., Baton Rouge, LA 70803, eberdahl@ccrma.stanford.edu)

Many traditional acoustic musical instruments are convenient to use: a performer picks one up, provides an energetic excitation, and the resulting sound radiates immediately. No wires, protocols, or software updates are ever required. The same cannot be said for the vast majority of prior digital musical instruments. The present work addresses how to endow a digital musical instrument with the convenience, look, feel, and personality of a traditional acoustic musical instrument via enclosure prototyping techniques, audio amplification, and embedded computation. For example, the Stingray is an embedded acoustic instrument. Although its battery needs to be occasionally charged, it otherwise can give the impression of a traditional acoustic musical instrument. The control inputs for the Stingray include a piano keyboard and force-feedback motorized faders. The faders allow the performer to interact expressively with the sound, while the piano keyboard enables the precise selection of notes. If desired, the Stingray can be programmed using physical models, including models of hypothetical acoustic instruments that would be infeasible to build physically. In this configuration, the Stingray expressively transforms the performer's gestures into radiated sound in an energy-conserving manner.

3:35

2pMU9. Towards a bendable circuit model of the Casio SK-1 keyboard. Kurt J. Werner (CCRMA, Stanford Univ., 223 Ayrshire Farm Ln., Apt. 205, Stanford, CA 94305, kwerner@ccrma.stanford.edu)

The Casio SK-1 keyboard was introduced in 1985 and synthesizes sampled and built-in sounds via pulse-code modulation and additive ("harmonic") synthesis. Initially important as one of the first home keyboards with sampling capabilities, the SK-1 has become one of the most popular instruments for circuit bending, the process of creatively modifying or augmenting sound-producing electronic devices. I create a parameterized component-level software model of the analog circuitry of the Casio SK-1, with applications to archiving and preserving its historic sound, expanding its basic behavior through circuit-bent modifications and extensions, and providing circuit benders with a resource for informed bending. Throughout, special attention is paid to creating models in terms of the circuit's component values. The SK-1's Percussion, Bass, and Chord Filters are modeled and analyzed in continuous-time as transfer functions and discretized via the bilinear transform. The non-linear (including diodes and transistors) Envelope/Pitch Mixing Circuit and Melody Filter are analyzed with linearizing simplifications to elucidate design intent, and modeled as ordinary differential equations to capture their behavior accurately. A review of techniques for the numerical solution of ordinary differential equations follows. A model of the internal speaker's impulse response and estimated static non-linearity rounds out the project.

Session 2pNS

Noise, Physical Acoustics, and Structural Acoustics and Vibration: Launch Vehicle Acoustics

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

Tracianne B. Neilsen, Cochair
Brigham Young Univ., N311 ESC, Provo, UT 84602

R. Jeremy Kenny, Cochair
NASA, M.S. ER42, Bldg. 4203, Marshall Space Flight Ctr., Huntsville, AL 35812

Invited Papers

1:00

2pNS1. Far-field acoustic modeling of rocket noise to determine community impacts. Michael M. James, Alexandria R. Salton, and Micah Downing (Blue Ridge Research and Consulting, LLC, 15 W. Walnut St., Ste. C, ASHEVILLE, NC 28801, michael.james@blueridgeresearch.com)

The emerging commercial space market is generating interest in commercial launch site ("spaceport") development around the United States. FAA regulations require all new spaceports to acquire a launch site operator's license, which is considered a Federal action subject to environmental review. Potential noise impacts are evaluated based on FAA Order 1050.1E, Change 1, Environmental Impacts: Policies and procedures, which include the assessment of DNL and may be supplemented with additional acoustical metrics. These supplemental metrics may range from speech interference to structural damage impacts. Extensive studies and research have examined the appropriateness of these metrics in relation to aircraft operations. However, the differences between these acoustic sources and operational modes stress the need for computer models and impact criteria specific to launch vehicles. Further measurements and research are needed to improve rocket source characterization, long-range sound propagation of high amplitude waveforms through complex atmosphere, and environmental and community impacts. The evolving nature of the regulatory environment surrounding rocket noise warrants a renewed focus on appropriate noise modeling and impact criteria to determine potential conflicts with launch noise.

1:20

2pNS2. Use of a large microphone array to identify noise sources during a rocket engine test firing and a rocket launch. Jayanta Panda (Experimental Aero-Phys. Branch, NASA Ames Res. Ctr., M.S. 260-1, Moffet Field, CA 94035, jayanta.panda-1@nasa.gov), Robert N. Mosher, and Barry J. Porter (Experimental Aero-Phys. Branch, Aerosp. Computing, Inc., Mountain View, CA)

A 70 microphone, 10 ft×10 ft, microphone phased array was built for use in the harsh environment of rocket launches. The array was setup at NASA Wallops launch pad 0A during a static test firing of Orbital Sciences' Antares engines, and again during the first launch of Antares vehicle. It was placed 400 ft away from the pad, and was hoisted on a scissor lift 40 ft above ground. The data sets provided unprecedented insights into rocket noise sources. The duct exit was found to be the primary source during the static test firing; the large amount of water injected beneath the nozzle exit quenched all other sources. The noise maps during launch were found to be time-dependent. As the engines came to full power and became louder, the primary source switched from the duct inlet to the duct exit. Further elevation of the vehicle caused spilling of the hot plume, resulting in a distributed noise map covering most of the pad. As the entire plume emerged from the duct, and the on-deck water system came to full power, the plume itself became the loudest noise source. These noise maps will help to improve the sound suppression system for future launches.

1:40

2pNS3. Acoustic measurements of the Epsilon rocket at liftoff. Tatsuya Ishii, Seiji Tsutsumi, Kyoichi Ui, Hideshi Oinuma (JAXA, 7-44-1 Jindaiji-higashi-machi, Chofu, Tokyo 182-8522, Japan, ishii.tatsuya@jaxa.jp), Yutaka Ishii (Bruel & Kjaer Japan, Tokyo, Japan), and Kei Wada (Science Service, Inc., Tokyo, Japan)

Launch vehicles generate intense acoustic field caused by the enormous thrust power during liftoff, and this acoustic field leads to harmful payload vibration. Japan Aerospace Exploration Agency (JAXA) plans to launch a new solid propellant rocket, Epsilon, in 2013. This three-stage rocket utilizes a reliable first stage motor of the JAXA's H-2 rocket, SRB-A. Since the SRB-A is expected to cause excessive acoustic load, a countermeasure was required to mitigate the acoustic feedback to the vehicle. Computational works proposed a launch pad structure to attenuate the Mach wave radiation from the plume, and the acoustic wave generated by the plume impinging to the flame deflector. In the previous ASA conference, the authors introduced scale model tests using 1:42 scale rocket motors and the launch pad models. The scale model tests clarified the acoustic benefit of the launch pad structure. The tested geometry of the launch pad structure was adopted to the full-scale structure. Acoustic measurements are planned in the first launch of the Epsilon rocket

in order to evaluate the acoustic influence with this newly constructed launch pad structure. The measurement setup and a brief review of the measurements (if possible) are discussed.

2:00

2pNS4. Scale model thruster acoustic measurement results. Magda B. Vargas and Robert J. Kenny (MSFC, NASA, M.S. ER42, Bldg. 4203, Marshall Space Flight Ctr., Huntsville, AL 35812, magda.b.vargas@nasa.gov)

The Space Launch System (SLS) Scale Model Acoustic Test (SMAT) is a 5% scale representation of the SLS vehicle, mobile launcher, tower, and launch pad trench. The SLS launch propulsion system will be comprised of the Rocket Assisted Take-Off (RATO) motors representing the solid boosters and four Gas Hydrogen (GH2) thrusters representing the core engines. The GH2 thrusters were tested in a horizontal configuration in order to characterize their performance. In phase 1, a single thruster was fired to determine the engine performance parameters necessary for scaling a single engine. A cluster configuration, consisting of the four thrusters, was tested in phase 2 to integrate the system and determine their combined performance. Acoustic and overpressure data was collected during both test phases in order to characterize the system's acoustic performance. The results from the single thruster and 4-thuster system are discussed and compared.

2:20

2pNS5. Near-field/far-field study of the end-effects regime produced by large area ratio nozzles. Raymundo M. Rojo, Charles E. Tinney, Woutijn J. Baars (Aerosp., Univ. of Texas at Austin, 701 28th St., Apt. 407, Austin, TX 78712, raymundo.rojo46@gmail.com), and Joseph H. Ruf (NASA Marshall Space Flight Cente, Huntsville, AL)

Vibro-acoustic loads emanating from large area ratio rocket nozzles during start-up can be catastrophic to the launch system and payload. This study quantifies a particular feature referred to as the "end-effects regime", which is considered the largest source of vibro-acoustic loading during start-up [Nave and Coffey, AIAA Paper 1973-1284]. In this experiment, data acquired during the start-up sequence of several full-scale rocket engines are compared to the laboratory-scale measurements of a thrust-optimized parabolic-contour nozzle conducted in a fully anechoic chamber. The laboratory studies encompass both static and dynamic wall pressures measured inside the nozzle, as well as far-field acoustic surveys. The event produced during the "end-effects regime" was successfully reproduced in the sub-scale model, and was characterized in terms of its mean, variance, and skewness, as well as the spectral properties of the signal obtained by way of time-frequency analyses. The intensity and characteristic frequency of the event of interest are discussed through a comparison of the nominal values for the full-scale and sub-scale system and whether they obey with standard scaling laws.

2:40

2pNS6. Evaluation of Japanese current primary launch vehicle liftoff acoustic environment change due to launch pad facility modifications. Hiroki Ashida, Makoto Hirai (Aerosp. Systems, Mitsubishi Heavy Industries, Ltd., 10, Oye-cho, Minato-ku, Nagoya City, Aichi 455-8515, Japan, hiroki1_ashida@mhi.co.jp), Keita Terashima, and Takumi Ujino (Space Transportation Mission Directorate, Japan Aerosp. Exploration Agency, Ibaraki, Japan)

H-IIA, Japanese primary launch vehicle, has been successfully launched 21 flights with a success rate of 95.4%. During 12 years of its operational phase, extensive acoustic measurements on the vehicle and on the ground have been conducted to refine conventional prediction methods and to evaluate the effect of vehicle/pad configuration changes. In this presentation, the evaluation of the effects of major pad configuration modification on H-IIA liftoff acoustic environment is presented. The effect of water injection is also evaluated.

3:00–3:15 Break

3:15

2pNS7. Numerical study on acoustic generation of a supersonic jet impinging to deflectors. Seiji Tsutsumi, Ryoji Takaki (JEDI, JAXA, 3-1-1 Yoshinodai, Chuuou, Sagamihara, Kanagawa 252-5210, Japan, tsutsumi.seiji@jaxa.jp), Koji Okamoto (Dept. of Adv. Energy, Univ. of Tokyo, Kashiwa, Japan), and Susumu Teramoto (Dept. of Aeronautics and Astronautics, Univ. of Tokyo, Bunkyo-ku, Japan)

Acoustic wave generated from a $M = 1.8$ ideally expanded jet impinging on flame deflectors is investigated numerically in order to obtain the knowledge of flame deflector design to mitigate the acoustic loading on launch systems and payloads. Mechanism of acoustic wave generated from a 45-degree-inclined flat plate placed 5D downstream from the nozzle exit is analyzed first, and it is revealed that source of the acoustic wave is located where shock waves is formed due to the jet impingement. In this study, effect of the deflector shape will be discussed for better understanding the acoustic generation mechanism.

3:35

2pNS8. Towards jet acoustic prediction within the Launch Ascent and Vehicle Aerodynamics framework. Jeffrey A. Housman (Appl. Modeling and Simulation Branch, NASA Ames Res. Ctr., M.S. N-258, Moffett Field, CA 94035, jeffrey.a.housman@nasa.gov), Christoph Brehm (Appl. Modeling and Simulation Branch, Sci. and Technol. Corp., Moffett Field, CA), and Cetin Kiris (Appl. Modeling and Simulation Branch, NASA Ames Res. Ctr., Moffett Field, CA)

Understanding the acoustic environment generated during lift-off is critical for successfully designing new space vehicles. In order for modeling and simulation tools to effectively assist in the development of the vehicles, validation must be performed on simplified model problems. In this paper, time-accurate implicit large eddy and detached eddy simulations coupled with a linear acoustic propagation method are applied to a Mach 1.8 perfectly expanded jet impinging on a flat plate at 45 degrees. The Launch Ascent and Vehicle Aerodynamics (LAVA) code used to simulate this problem is a high-fidelity unsteady simulation tool for modeling fluid dynamics, conjugate heat transfer, and acoustics. A detailed description of the linear acoustic propagation tool is provided. The narrow band far-field

sound pressure levels predicted using LAVA are compared to existing experimental data. POD and spectral methods are applied to analyze the noise sources due to coherent flow structures and jet impingement. Grid and time-step sensitivity studies are performed to assess the spatial and temporal requirements for accurate jet acoustic simulation. Sensitivity of the predicted far-field sound pressure levels to position of the acoustic propagation surface is also assessed.

3:55

2pNS9. Intensity-based approach to characterize near-field acoustic environments of space flight vehicles. Michael M. James, Alexandria R. Salton (Blue Ridge Research and Consulting, LLC, 15 W. Walnut St., Ste. C, Asheville, NC 28801, michael.james@blueridgeresearch.com), Kent L. Gee, and Tracianne B. Neilsen (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT)

An intensity-based measurement probe has been developed to measure the magnitude, directivity, and spectral content of near-field rocket source noise. An array of the intensity-based measurement probes was deployed in a static test firing of the GEM-60 at Alliant Techsystems in Promontory, UT, in 2012. The probes were positioned along the shear layer of the rocket motor plume to enable a comparison of the resultant source characterization obtained via sound power type measurement approaches to traditional acoustic measurement methodologies. The measurement results demonstrate that the intensity-based probes advance the measurement and characterization of the near-field acoustic environment of rockets. Moreover, intensity-based acoustic data provides an important role in formulating more realistic sound source models, improving acoustic load estimations, and aiding in the development of the next-generation space flight vehicles via improved measurements of noise near the rocket plume.

4:15

2pNS10. Methods for estimating acoustic intensity in rocket noise fields. Derek C. Thomas, Benjamin Y. Christensen, and Kent L. Gee (Dept. of Phys. and Astronomy, Brigham Young Univ., N223 ESC, Provo, UT 84602, derekcthomas@gmail.com)

The acoustic field produced by launch vehicles is difficult to measure and characterize. Acoustic intensity measurements provide more information per measurement location than pressure measurements and are therefore interesting for the characterization of rocket noise fields. The extreme environment associated with a rocket requires a robust intensity probe while the large size of the source and the high-amplitude, highly nonlinear behavior of the system produce a signal with significant low and high frequency components. Thus, the probe must also provide accurate results over a large frequency band. The bandwidth limitations of the standard method for estimating acoustic intensity, the p-p finite difference method, motivated the development of an alternative method for intensity estimation. The new phase and amplitude gradient estimation (PAGE) method will be presented and compared to the standard p-p method. Specific features of the rocket noise field that can be leveraged to improve the bandwidth of the intensity estimates will also be discussed.

4:35

2pNS11. Acoustic intensity estimates from a solid rocket motor test firing. Benjamin Christensen, Derek C. Thomas, and Kent L. Gee (Phys. and Astronomy, Brigham Young Univ., 539E 300N, Provo, UT 84606, ukeben@gmail.com)

Acoustic measurements of a static GEM-60 solid motor test firing were taken as part of a continuing effort to characterize the aeroacoustic source regions and noise environment around launch vehicles. Multiple 2D intensity probes, consisting of four coplanar microphones, were used in the measurement. Two intensity estimation techniques have been applied to the data: the finite difference p-p method, and the new phase and amplitude gradient estimation (PAGE) method. We will present and compare results from both methods and compare to measurements made at past test firings. It appears that the PAGE method for estimating acoustic intensity provides usable results over a larger frequency bandwidth.

Session 2pPA

Physical Acoustics and Structural Acoustics and Vibration: Acoustics of Pile Driving: Measurements, Models, and Mitigation II

Kevin M. Lee, Cochair
Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758

Karl-Heinz Elmer, Cochair
OffNoise-Solutions GmbH, Leinstr. 36, Neustadt a. Rbge. 31535, Germany

Invited Papers

1:30

2pPA1. Effects of pile driving on fishes. Arthur N. Popper (Univ. of Maryland, Biology/Psych. Bldg, College Park, MD 20742, apopper@umd.edu), Michelle B. Halvorsen (Battelle-Pacific Northwest National Lab., Sequim, WA), Thomas J. Carlson (ProBioSound, Holmes Beach, FL), Michael E. Smith (Univ. of Western Kentucky, Bowling Green, KY), and Brandon M. Casper (Naval Submarine Medical Res. Lab., Groton, CT)

We examined the physiological effects of impact pile driving on fishes using a specially designed tube that allows replication of the far field acoustic conditions of impulsive stimuli. Studies show that the received signal levels needed to result in onset of effects is a combination of single strike sound exposure level (SELss) and cumulative sound exposure level (SELcum), although not in an equal energy relationship. In contrast to current interim regulations, which indicate that the onset of physiological effects occurs at 187 dB SELcum, our experimental results for six species of fishes showed that the onset of physiological effects, none of which produced mortality, was at about 207 dB SELcum. This onset SELcum had to be at least 7–10 dB higher to result in effects that could potentially be mortal. Additional studies showed that fishes can recover from the effects of pile driving and that a fish species without a swim bladder showed no effects, at least up to an SELcum of 216 dB. Investigations on the effects of pile driving on sensory hair cells of the inner ear, an analog for hearing loss, showed that damage only occurred at SELcum that are substantially higher than onset of other effects.

1:50

2pPA2. A Monte Carlo approach to determining marine mammal exposure risk to long term marine piling operations. Paul A. Lepper (School of Electron., Elec. and Systems Eng., Loughborough Univ., Loughborough LE113TU, United Kingdom, p.a.lepper@lboro.ac.uk), Stephen P. Robinson, Pete D. Theobald, and Tanja Pangerc (Acoust. Group, National Physical Lab., Teddington, United Kingdom)

The expansion of offshore renewable developments, primarily offshore wind, has led to widespread use of large scale percussive piling for foundation construction. In UK waters alone, over 900 foundations installed up to 2013, mostly mono-piles with extensive upscaling of developments planned for the next decade. Pile diameters range from a few meters to greater than 6 m and penetration depths of 20–30 m. These piles are typically percussively driven with several thousand hammer strikes over periods of several hours with individual strike hammer energies in excess of 1900 kJ occasionally used and reported per strike underwater Sound Exposure Level source levels of 215 dB re 1 $\mu Pa^2 s\text{-}m^2$. Potential exists for injury to occur from cumulative sound exposure to repetitive but lower level signals at greater range. If simple receptor behaviors are assumed (static, fleeing, etc.) exposure over time to an entire pile construction sequence can be estimated. These models have been extended using a Monte Carlo approach to model long term, entire wind farm construction scenarios with repetitive foundation construction periods of 24–36 h. The statistical distribution of exposure risk is modeled as well as analysis of the sensitivity of behavioral responses to potential impact effects such as habitat exclusion.

2:10

2pPA3. Noise mitigation systems (NMS) for reducing pile driving noise: Experiences with the "big bubble curtain" relating to noise reduction. Michael A. Bellmann and Patrick Remmers (itap GmbH, Marie-Curie-Str. 8, Oldenburg 26160, Germany, bellmann@itap.de)

For the offshore wind farm Borkum West II in the German North Sea the Noise Mitigation System (NMS) "Big Bubble Curtain" was used during pile driving activities. Within this project systematically variations of different influencing factors on noise reductions such as air volume, nozzle hose sizes, distance of nozzle hoses, etc., were investigated. Additionally the "Big Bubble Curtain" is currently in use for different other OWF in the German North Sea. Therefore, the "Big Bubble Curtain—BBC" is at the moment one of the most investigated NMS under offshore condition. Within this presentation, experiences and results of the above listed projects will be shown and discussed.

2:30

2pPA4. Evaluation of hydro sound and vibration measurements during the use of the Hydro-Sound-Damper (HSD) at the wind farm "London Array". Benedikt Bruns (Technische Universität Braunschweig, Beethovenstrasse 51 b, Braunschweig 38106, Germany, b.bruns@tu-bs.de)

Since some years a noise prevention concept for the protection of marine animals exists in Germany. Based on that, the acoustic underwater noise from the pile driving at offshore wind farms is required to be less than 160 dB (SEL) at a distance of 750 m. This value, however, is often exceeded so that the use of a soundproofing system is necessary. The Hydro-Sound-Damper (HSD) is a new, versatile method to reduce the noise during offshore pile driving. To achieve this, elements of different sizes and materials are used, which are fixed to fishing nets. The principle of operation and the effectiveness of these HSD elements were investigated in the laboratory and in situ under offshore conditions at the world's largest offshore wind farm "London Array." During the offshore tests thorough measurements were performed which metered the propagation of the hydro sound and the vibrations of the sea floor at various distances and directions. The evaluation of these data led to very promising results concerning underwater noise reduction. This article describes the theory and implementation of the HSD at "London Array" and focuses on the interpretation of the data from the hydro sound and vibration measurements.

Contributed Papers

2:50

2pPA5. Efficient application of encapsulated bubbles and foam elements to mitigate offshore piling noise. Karl-Heinz Elmer (OffNoise-Solutions GmbH, Leinstr. 36, Neustadt a. Rbge. 31535, Germany, karl-heinz.elmer@t-online.de)

The very high noise levels of offshore piling noise during the installation of offshore wind converters are dangerous to marine life like harbor porpoises and fishes. Encapsulated bubbles and foam elements are successfully used to reduce the very high noise levels. There are different physical effects such as impedance mismatch, resonance and scattering effects and material damping effects that are responsible to the measured noise reduction between about 10 dB (SEL) and 23 dB (SEL). The radiation of underwater noise from the pile surface, the propagation and the reflections of the radiated waves are investigated together with the influence of encapsulated bubbles and foam elements in the near field of the pile. Only a very small part of the impact energy is radiated directly from the pile into the surrounding water. Most of the hammer energy is driven into the ground. A small part from this is radiated indirectly from the ground into the water. The efficiency of encapsulated bubbles and foam elements in underwater noise mitigation is investigated by theoretical studies, laboratory measurements, and by offshore measurements with different elements and combinations.

3:05

2pPA6. Dependence of resonance frequencies and attenuation for large encapsulated bubbles on bubble wall thickness and bubble fill-material. Gregory Enenstein, Preston S. Wilson, and Kevin M. Lee (Mech. Eng. Dept. and Appl. Res. Lab., The Univ. of Texas at Austin, 4700 W Guadalupe St #A-437, Austin, TX 78751, gregenenstein@gmail.com)

Arrays of large encapsulated bubbles are currently under development for the purpose abating low-frequency anthropogenic underwater noise from various sources including marine pile driving and oil and gas exploration and production. An existing predictive model [Church, J. Acoust. Soc. Am. **97**, 1510–1521 (1995)], which was originally intended to describe propagation through suspensions of microbubbles used as ultrasound contrast agents, was previously found to be in good agreement with resonance frequency and attenuation measurements using large encapsulated with radii on the order of 10 cm [Lee *et al*., J. Acoust. Soc. Am **132**, 2039 (2012); Lee and Wilson, Proceedings of Meetings on Acoustics **19**, 075048 (2013)]. For the current study, both laboratory and lake experiments were performed on large encapsulated bubbles to investigate the dependence of the bubbles' resonance frequencies and attenuation on the bubble wall thickness. Additionally, laboratory measurements were made to investigate the effects on encapsulated bubble resonance frequencies and damping using bubble fill-materials other than air, and a lake experiment was then performed to relate these effects on the damping to the attenuation provided by arrays of such bubbles. [Work supported by AdBm Technologies.]

3:20–3:35 Break

3:35

2pPA7. Acoustic measurements and modeling of air-filled, underwater resonator cavities. Laura Tseng, Kevin M. Lee (Appl. Res. Laboratories: The Univ. of Texas at Austin, 10,000 Burnet Rd., Austin, TX 78758, Ltseng@utexas.edu), Preston S. Wilson (Mech. Eng., The Univ. of Texas at Austin, Austin, TX), and Mark S. Wochner (AdBm Technologies, Austin, TX)

This paper investigates the near-resonance acoustical properties of submerged air-filled resonators intended for use in an underwater noise abatement system. These resonators are a potential alternative to encapsulated bubbles. The resonators are similar to Helmholtz resonators in shape and design, but without a neck, consisting of underwater inverted air-filled cavities with rigid walls. A finite element model was developed to investigate the acoustic behavior of the resonators near their resonance frequencies, and based on the results of the model, physical realizations of the resonators were designed and fabricated for testing. Experiments were performed with the resonators in a closed water-filled tank operated in the long wavelength limit [J. Acoust. Soc. Am. **132**, 2039 (2012)], where their resonance frequencies and Q-factors were measured using the technique described by Leighton *et al*., [J. Acoust. Soc. Am. **112**, 1366–1376 (2002)]. Comparison between the results from the measurements and modeling will be discussed. [Work supported by Shell Global Solutions.]

3:50

2pPA8. Radiated sound from a scale-model pile submerged in a two--layer medium. Kevin M. Lee, Todd A. Hay, Taylor W. Weaver, Preston S. Wilson, and Mark F. Hamilton (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, klee@arlut.utexas.edu)

Underwater noise due to both marine pile driving and offshore wind farm operation is not only radiated directly from the pile into the water, but also from the seabed surrounding the pile. While there is much interest in mitigating the noise from these activities, a better understanding of the source mechanisms and propagation is needed to determine optimal strategies for noise abatement. A recent analytical model of the acoustic field radiated by submerged piles includes radiation from the pile directly into the water and into a stratified viscoelastic sediment as well as propagation into a shallow water waveguide from both the direct and sediment radiation paths [Hay *et al*., Proceedings of Meetings on Acoustics **19,** 070038 (2013)]. As a step towards validating this model, scale-model experiments were conducted in the high kilohertz frequency range with a model pile consisting of a mechanically excited metallic tube inserted into a laboratory tank filled with two stratified layers to simulate the water/sediment interface. Measurements of the acoustic field in the experiment are compared with the model predictions, and the relevance of these results to implementing noise abatement strategies will be discussed. [Work supported by ARL:UT IR&D.]

4:05

2pPA9. On the resonance frequency of an ideal arbitrarily shaped bubble. Kyle S. Spratt, Kevin M. Lee (Appl. Res. Labs., The Univ. of Texas at Austin, 4307 Caswell Ave., Apt. E, Austin, TX 78751, sprattkyle@gmail.com), Preston S. Wilson (Dept. of Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), and Mark S. Wochner (AdBm Technologies, Austin, TX)

Large encapsulated bubbles have recently been described for use in abating low-frequency anthropogenic underwater noise [J. Acoust. Soc. Am. **130**, 3325–3332 (2011)], and the use of encapsulation allows for the possibility of bubbles that are nonspherical in their equilibrium state. For the purpose of more accurately determining such bubbles' resonance frequencies, a lumped-element model of the linear oscillation of an ideal, arbitrarily shaped gas bubble in an incompressible liquid is presented. The corresponding boundary-value problem required to predict the resonance frequency of the bubble is seen to be equivalent to a classic problem in electrostatics [J. Acoust. Soc. Am. **25**, 536–537 (1953)]. Predictions made for the resonance frequency of prolate and oblate spheroidal bubbles using this model are tested against a finite-element model of the full acoustic scattering problem. Particular attention is then paid to the case of an ideal toroidal bubble of arbitrary thickness, and predictions made for the resonance frequency of such a bubble using the lumped-element approach are compared to a finite-element model of the full acoustic scattering problem as well as to existing approximate models for the dynamics of thin toroidal bubbles. [Work supported by AdBm Technologies, LLC and the ARL:UT McKinney Fellowship in Acoustics.]

4:20

2pPA10. The effect of sediment stiffness on the piling pulse—Results from a wave-equation analysis model. Michael A. Wood and Victor F. Humphrey (Fluid Dynam.. and Acoust. Group, Inst. of Sound and Vib. Res., Faculty of Eng. and the Environment, Univ. of Southampton, Southampton, Hampshire SO17 1BJ, United Kingdom, maw1v07@soton.ac.uk)

The rapid expansion of wind farms in UK waters has led to increased concern of the anthropogenic noise emitted into the sea due to piling. The piling process generates high levels of noise capable of propagating over long distances in the water column; it is important that this noise can be predicted, and the likely environmental impact determined. The model presented in this work comprises a method known as Wave Equation Analysis of Piles. This technique models the stress-wave as it propagates along the pile using a finite-difference approach. Additionally, the plastic nature of the sediment is modeled at both the pile wall and pile toe. Previous work has shown that by considering the radial expansion of the pile, the results of this model may be coupled to an acoustic model. The effect of the sediment on the pile motion has previously received little attention. The results show that although the first downward-going pulse is independent of the sediment parameters, reflected pulses are affected. The calculations show that there exists an impedance condition based on the soil parameters: for low soil stiffnesses the pulse is inverted at the toe end, whereas for higher stiffnesses no inversion is seen.

4:35

2pPA11. Physical models and improvement of bubble curtain for the suppression of underwater noise from a pile drive. Alexander Sutin and Hady Salloum (Stevens Inst. of Technol., 711 Hudson St., Hoboken, NJ 07030, asutin@stevens.edu)

Man-made sounds in ocean and inland water environments become biologically significant when they affect the ability of animals and fish to survive and reproduce. Extremely strong sounds produced by pile driving can highly exceed the environmental safety limits and application of bubble curtain is one of the most effective methods for sound suppression. We present several physical models explaining sound suppression by bubble curtain and discuss the methods of improving bubble curtain efficiency. The physical models of sound suppression by a bubble curtain include: (a) Estimation of bubble curtain impedance leading to decreasing of the pile drive acoustic coupling with surrounding bubbly water and (b) theoretical model for the estimation of sound attenuation by resonance bubbles. The developed models were analyzed for the optimization of the pile drive sound suppression. Several methods for generating a bubble curtain with small bubbles and bubbles with varied sizes are considered. We also suggest a way for improving the efficiency of bubble curtains by increasing the lifetime of the bubbles using bubble coating. Coated micro bubbles are widely used as ultrasound contrast agents in cardiology. One of the simplest ways for micro bubble coating is the passing of bubbles through oil.

TUESDAY AFTERNOON, 3 DECEMBER 2013 CONTINENTAL 2/3, 1:00 P.M. TO 5:00 P.M.

Session 2pPP

Psychological and Physiological Acoustics: Building a Stairway to Hearing

Sridhar Kalluri, Chair
Starkey Hearing Res. Ctr., Starkey Hearing Technologies, 2150 Shattuck Ave., Ste. 408, Berkeley, CA 94704

Contributed Papers

1:00

2pPP1. Testing and extending the Woodworth model. William M. Hartmann (Phys. & Astronomy, Michigan State Univ., East Lansing, MI 48824, hartman2@msu.edu) and Neil L. Aaronson (Natural Sci.& Mathematics, Richard Stockton College of NJ, High Bridge, NJ)

The Woodworth model and formula for interaural time difference is frequently used as a standard in physiological and psychoacoustical studies of binaural hearing for humans and other animals. It is a frequency-independent, ray-tracing spherical head model that is expected to agree with an exact diffraction model in the high-frequency limit. The predictions by the Woodworth model for antipodal ears and for incident plane waves are compared with the predictions of the exact model as a function of frequency to quantify the discrepancy when the frequency is not high. In a second calculation, the Woodworth model is extended to arbitrary ear angles, both for plane-wave incidence and for finite point-source distance. This extended Woodworth model leads to different formulas in six different regions defined by ear angle and source distance. It is noted that the characteristic cusp in Woodworth's well-known function comes from ignoring the longer of the two paths around the head in circumstances when the longer path is actually important. This error can be readily corrected.

1:15

2pPP2. Minimum audible angle at the subjective front during listener's active and passive head rotation. Yôiti Suzuki (Res. Inst. of Elec. Commun., Tohoku Univ., Katahira 2-1-1, Aoba-ku, Sendai 981-0942, Japan, yoh@riec.tohoku.ac.jp), Akio Honda (Tohoku Fukushi Univ., Sendai, Japan), Kagesho Ohba, Shuichi Sakamoto (Res. Inst. of Elec. Commun., Tohoku Univ., Sendai, Japan), and Yukio Iwaya (Tohoku Gakuin Univ., Sendai, Japan)

Listener's head movement, particularly horizontal rotation, effectively improves sound localization acuity (Wallach, 1939; Thurlow, 1967; Kawaura, 1989). However, few findings have been obtained concerning sound localization during head rotation. In the present study, we directly investigated the minimum audible angle (MAA) at the front during horizontal rotation. A sound stimulus (30-ms noise burst) was presented from a loudspeaker of a circular array (r = 1.1 m), with a loudspeaker separation of 2.5 degrees. The listener, sitting at the center of the circle, was asked to answer whether the sound stimulus was presented from the left or right of the subjective front (2AFC). We designed three listening conditions, static, active rotation and passive rotation. In the static condition, listeners were asked to keep their heads still. For the active rotation condition, listeners were asked to rotate their heads. Meanwhile, for the passive rotation condition, listeners sitting on a revolving chair were rotated by an experimenter. In the latter two conditions, the test stimulus was triggered during head movement. Results showed the MAA to deteriorate significantly in the two rotation conditions. This implies that the improvements in sound localization due to head motion could be explained by the multiple-look model (Viemeister, 1991).

1:30

2pPP3. Difference of the perceived auditory space between walking and passive self-motion. Shuichi Sakamoto (Res. Inst. of Elec. Commun., Tohoku Univ., 2-1-1 Katahira, Aoba-ku, Sendai, Miyagi 980-8577, Japan, saka@ais.riec.tohoku.ac.jp), Hideaki Terashima (Graduate School of Information Sci., Tohoku Univ., Sendai, Japan), Wataru Teramoto (Dept. of Comput. Sci. and Systems Eng., Muroran Inst. of Technol., Muroran, Japan), Yôiti Suzuki (Res. Inst. of Elec. Commun., Tohoku Univ., Sendai, Japan), and Jiro Gyoba (Graduate School of Arts and Letters, Tohoku Univ., Sendai, Japan)

We are investigating how auditory space was represented during linear self-motion (Teramoto *et al*., 2013). Several studies have suggested that whether the listener's motion is active or passive affected sound localization (Hirahara *et al*., 2013). In the present study, therefore, we set up three conditions: active motion condition, passive motion condition, and no motion condition. In active motion condition, observers were walking straight ahead. In passive motion condition, observers were transported forward by a robotic wheelchair. During the self-motion, a short noise burst was presented from one of the loudspeakers which were aligned parallel to the traveling direction when the listener's coronal plane reached the location of one of the speakers (null point). The listeners indicated the direction in which the sound was perceived relative to their coronal plane (i.e., a two-alternative forced-choice task). The results of experiment showed that the sound position aligned with the subjective coronal plane was displaced compared with the null point. However, there was no significant difference between auditory space in active and passive motion conditions. This result suggests only action of the kinetic system during self-motion and planning and execution of voluntary movement would not affect perceived auditory space.

1:45

2pPP4. Sound source localization from tactile aids for unilateral cochlear implant users. Xuan Zhong, Shuai Wang, Michael Dorman, and William Yost (Dept. of Speech and Hearing Sci., Arizona State Univ., 975 S Myrtle Ave., Lattie F Coor Hall 2211, Tempe, AZ 85287, xuan.zhong@asu.edu)

The present research asks whether two tactile aids with directional microphones, by providing additional inter-channel level information and etc., could help unilateral cochlear implant (CI) localize sound sources. For normal hearing subjects, sound source localization based on tactile vibration cues alone can be as accurate as auditory localization in the frontal horizontal plane (Gescheider, 1970). CI users may as well benefit from additional tactile aids just as normal hearing people do. The current study uses two bone-anchored hearing aids (BAHA) as sources of tactile vibration. The two BAHAs, bonded together by a special gadget to maintain a particular distance and angle, both have directional microphones, and are programed so that one point to the front-left side and the other to the front-right side. Unilateral CI users voluntarily participated in the experimental study. Wide band noise stimuli were presented at 65 dB SPL. The subjects hold one BAHA in each hand and do localization tasks with (1) CI only and (2) with CI and tactile sensation combined. Preliminary data shows CI users can get some benefit from the additional information provided by tactile aids in 360 degree localization (45 degree spacing) on the horizontal plane.

2:00

2pPP5. Characterization of the available feedback gain margin at two device microphone locations, in the fossa triangularis and Behind the Ear, for the light-based contact hearing device. Suzanne C. Levy, Daniel J. Freed (EarLens Corp., 200 Chesapeak Dr., Redwood City, CA 94063, slevy@earlenscorp.com), and Sunil Puria (EarLens Corp., Stanford, CA)

Assistive devices compensate for hearing impairment by amplifying sounds with gain, which is limited by acoustic feedback. The light-based Contact Hearing Device (CHD) provides amplification to 10 kHz by mechanically vibrating the umbo with a wireless Tympanic-Contact Actuator (TCA). Driving the eardrum mechanically generates a pressure wave, which travels laterally down the ear canal and produces feedback. Placing the microphone in the fossa triangularis (FT) may preserve more natural acoustic cues than the BTE location, although it may reduce available feedback gain margin (FGM). Thirteen subjects with bilateral mild-to-severe hearing impairment were fit with CHDs (26 ears). The TCA was driven with light-pulses and the feedback pressure was measured at the FT and above the pinna at the BTE microphone locations. The mean FGM varied from 32 to 48 dB and 38 to 60 dB for the FT and BTE microphone locations, respectively. FGM was lowest in the 3–5 kHz range and highest at about 7 kHz (below 1 kHz FGM is not measurable). The STD of FGM varied from 5.3 to 13 dB due individual anatomies. A microphone at the BTE has 6 to 12 dB additional FGM over the FT location, and allows a broader inclusion range to fit patients with amplification to 10 kHz.

2:15

2pPP6. A long-overdue review of empirical uncertainties in the "fatigue" found through simultaneous dichotic loudness balance. Lance Nizami (Independent Res. Scholar, Wilkie Way, Palo Alto, CA 94306, nizamii2@att.net)

SDLB uses equalization of the loudnesses of stimuli at the two ears (one stimulus intermittent, one constant) to measure the alleged "fatigue" over time of the loudness of the constant stimulus (Hood, 1950). Hood found 50 dB of "fatigue", and SDLB remained influential for over a quarter-century. However, the interpretation of SDLB was long questioned; recently, a novel model uniting the physiology and the behavior emerged (Nizami 2012, Int. Soc. Psychophys., Ottawa, Canada), and others independently re-measured "fatigue". Classically, the stimulus waveforms at the two ears were similar, sometimes identical, permitting two equalization techniques when the stimuli coincided in time: centering of the sound between the ears (lateralization), or matching the loudness contributions from each ear (loudness-matching). Further, "fatigue" was habitually expressed as across-listeners averages. But careful scrutiny reveals that (1) over the "fatiguing" duration, lateralization may give way to loudness-matching, (2) dedicated loudness-matching may nonetheless yield only half as much "fatigue" as lateralization, and (3) the standard deviation of "fatigue" can be half its mean value, such that some listeners would not have "fatigued." In sum, the magnitude of "fatigue" is remarkably uncertain, and is likely to remain so until auditory physiology is compellingly integrated into explanations of SDLB.

2:30

2pPP7. Influence of measurement method and context of presentation on the loudness difference between increasing and decreasing intensity sounds. Emmanuel Ponsot, Patrick Susini (IRCAM, 1 Pl. Igor Stravinsky, Paris, France, patrick.susini@ircam.fr), and Sabine Meunier (LMA, CNRS, UPR 7051, Aix-Marseille Univ, Centrale Marseille, Marseille, France)

Four experiments were conducted to assess the loudness of both increasing and decreasing intensity sounds using different methods and context of presentation as between-subjects factors. In Exp 1 and Exp 2, loudness was assessed directly by using magnitude estimation procedures, with increasing and decreasing sounds presented respectively either in the same block or in separate blocks. In the other two experiments, loudness was measured by the mean of pairwise comparisons. While increasing and decreasing sounds were compared with each other in Exp 3, they were compared respectively with constant-intensity sounds in Exp 4. Two-intervals, 2AFC interleaved-adaptive procedures were used to prevent from potential biases. As a result, very similar trends were observed in the four experiments. In particular, the loudness difference between increasing and decreasing sounds always felt within the same range: decreasing intensity sounds need to be about 3 dB louder that increasing sounds to be perceived with equal loudness. This study thus indicates that this loudness asymmetry actually corresponds to a true perceptual effect and is not due to any experimental bias, since a clear consistency across the results was found using different measurement methods and context of presentation.

2:45–3:00 Break

3:00

2pPP8. Effect of sound duration on loudness estimates of increasing and decreasing intensity sounds. Emmanuel Ponsot, Anne-Laure Verneil, and Patrick Susini (IRCAM, 1 Pl. Igor Stravinsky, Paris 75004, France, ponsot@ircam.fr)

The influence of sound duration on global loudness of non-stationary stimuli was investigated. Loudness of 2 and 6-s increasing and decreasing intensity sounds with different ranges of intensity-variation was assessed using a magnitude estimation procedure. Results once again uphold the existence of a loudness difference between the two patterns: while they only differ in their temporal profile, increasing sounds were perceived louder than decreasing sounds. In addition, global loudness estimates were increased with duration for the two types of sounds, and a small but significant interaction occurred between type and duration. A contrast analysis revealed that while global loudness of increasing and decreasing sounds raised with duration in a similar way in the case of low and moderate intensities (below 75 dB SPL), global loudness was significantly more affected by duration with increasing than with decreasing intensity profiles for high-intensity stimuli. This result suggests the existence of an underlying memory process combined with a "peak-end rule" as being responsible for the loudness asymmetry typically observed between the two types of intensity-pattern being judged [Susini *et al*. (2010). "End level bias on direct loudness ratings of increasing sounds" J. Acoust. Soc. Am. EL **128**(4), 163–168].

3:15

2pPP9. A novel bispectral approach to study cochlear non-linearities in transient evoked otoacoustic emissions. Gabriella Tognola, Alessia Paglialonga, Emma Chiaramello, and Stefano Moriconi (Inst. of Electronics Computer, and Telecommun. Eng. ISIB, CNR It. Natl. Res. Council, Piazza L. da Vinci, 32, Milan 20133, Italy, gabriella.tognola@polimi.it)

A new approach, based on bispectral analysis, is proposed to study non-linearity in cochlear active mechanisms, as evaluated in transient evoked otoacoustic emissions (TEOAEs). Based on an ad hoc formulation of the bispectral analysis, specifically developed in this study to fit the particular features of TEOAEs, this innovative method detects non-linearity by extracting quadratic frequency couplings (QFCs) in TEOAE recordings. The method directly estimates non-linear TEOAE components, which may not be detected by conventional spectral analysis and can thus provide important information useful for better modeling cochlear active mechanisms and to detect sub clinical cochlear dysfunctions. The method was characterized with synthesized TEOAEs as a function of the main TEOAE parameters and then used to analyze TEOAEs recorded in normal hearing adults and full-term neonates to: (i) obtain normative data; (ii) to evaluate test retest repeatability of non-linear interaction mechanisms; and (iii) to investigate the influence of stimulus intensity on non-linear interaction mechanisms. Results revealed that most of the energy of non-linear components is located in the 1–4 kHz range and that the test retest repeatability of non-linear interaction mechanisms is high. The factor that most affects non-linearities is stimulus intensity.

3:30

2pPP10. Finite element model of feed-forward/feed-backward amplification in the mouse cochlea. Joris Soons (Biomedical Phys., Univ. of Antwerp, 496 Lomita Mall, Stanford, California 94305, jsoons@stanford.edu), Sunil Puria, and Charles R. Steele (Mech. Eng., Stanford Univ., Stanford, CA)

Thousands of hair cells in the organ of Corti, situated along the basilar membrane (BM), detect displacements due to sound input. For low input sounds, these displacements are amplified by active outer hair cells (OHCs). A proposed theory is the feed-forward/feed-backward mechanism for the OHC amplification where an expanding hair cell gives a forward push through the Deiters Cells and a backward pull on the BM through the Phalangeal process. Previously this was implemented mathematically using WKB theory (Yoon *et al*. 2011, Biophys. J.). In the present work, we explicitly modeled this as a Y-shaped arrangement of the OHC-DeitersCell-PhallangealProcess to form a building block using beam elements in a finite element formulation. These Y-shaped blocks were chained together to construct a single-row organ-of-Corti model from the base to apex, coupled to the BM and scalae fluid, of a mouse cochlea. The OHC force is proportional to the shear on the BM. For a 10 kHz stapes input tone, the passive BM reaches a peak gain of about 26 dB. For the active case the BM gain increases to 58 dB and shifts apically by about 0.6 mm. These results are consistent with physiological measurements in several other living animals.

3:45

2pPP11. Should the acceptable noise level be considered to be an acceptable noise range? Jonas Brännström, Lucas Holm, Tobias Kastberg (LogopeDC, phoniatrics, and Audiol., Lund Univ., Clinical Sci. Lund, Lund SE-22185, Sweden, jonas.brannstrom@med.lu.se), and Steen J. Olsen (Dept. of Otorhinolaryngology, Head and Neck Surgery, Univ. Hospital, Rigshospitalet, Copenhagen, Denmark)

The acceptable noise level (ANL) test is used to quantify the amount of competing background noise (BNL) that a listener is willing to accept when listening to speech at the most comfortable level (MCL). ANL is calculated by subtracting the BNL from the MCL. Most studies show large intersubject ANL variability and a few also demonstrate large intrasubject variability. Very few predictor variables for ANL have been identified and it has been proposed that the ANL depends on an inherent characteristic of the listener. However, some of the variability seems to depend on poor precision of the ANL test. After removing the effect of poor precision, some variability still remains. One possible explanation for these findings may be that the ANL is not a single level but a range of levels. Using recent data, this presentation examines the notion of an acceptable noise range.

4:00

2pPP12. Effects of speakers' language background on speech perception in adults. Mark A. Dame, Harisadhan Patra, Petula C. Vaz (Audiol. & Speech Pathol., Bloomsburg Univ. of PA, 226 CEH, 400 E 2nd St., Bloomsburg, PA 17815, hpatra@bloomu.edu), and Biswajit Ray (Electronics Eng. Technol., Bloomsburg Univ. of PA, Bloomsburg, PA)

With an increasingly changing sociolinguistic environment in the U.S., clinicians are challenged to accurately evaluate individuals' speech perception under realistic everyday listening conditions. This study investigated how contextual linguistic information, speakers' language background, speech-rate, and background noise individually and interactively affect listeners' ability to recognize sentences. Ten normal-hearing American English native speakers, aged 20–22 years were recruited as listeners. Eight normal-hearing native speakers of four different languages (American English, Chinese-Mandarin, German, and Spanish) were recruited as speakers. These speakers read 120 sentences, 60 high-predictability and 60 low-predictability sentences from the revised Speech in Noise test (Bradlow and Alexander, 2007), which served as test stimuli. The listeners reported the target word of each sentence presented either in quiet or in multitalker babble. Results revealed that noise, high speech-rate, and lack of contextual cues could have significant adverse affects on listeners' test scores. Speakers' language background also had significant effects on listeners' performance. Specifically, listeners had the most difficulty in perceiving the Chinese-Mandarin speakers, followed by the Spanish, and German speakers; more so in multitalker babble than in quiet. Further studies are warranted since results may have implications for audiology diagnosis and rehabilitation to address effective everyday communication.

4:15

2pPP13. Training Mandarin-speaking amusics to recognize pitch direction: Pathway to treat musical disorders in congenital amusia? Fang Liu (Dept. of Linguist and Modern Lang., The Chinese Univ. of Hong Kong, Rm. G36, Leung Kau Kui Bldg., Shatin, N.T., Hong Kong, China, fangliufangliu@gmail.com), Cunmei Jiang (Music College, Shanghai Normal Univ., Shanghai, China), Tom Francart (ExpORL, Dept. of NeuroSci., KU Leuven, Leuven, Belgium), Alice H. Chan (Div. of Linguist and Multilingual Studies, School of Humanities and Social Sci., Nanyang Technolog. Univ., Singapore, Singapore), and Patrick C. Wong (Dept. of Linguist and Modern Lang., The Chinese Univ. of Hong Kong, Hong Kong, China)

Congenital amusia is a lifelong disorder of musical perception and production that has been hypothesized as due to impaired pitch direction recognition. This study investigated whether amusics could be trained to identify pitch direction in speech and music, and if so, whether enhanced pitch direction recognition would benefit musical processing in amusia. Eighteen Mandarin-speaking amusics and 18 matched controls were evaluated using the Montreal Battery of Evaluation of Amusia (Peretz *et al*., 2003) and tested on two psychophysical pitch threshold tasks for identification of pitch direction in the Mandarin syllable /ma/ and its piano tone analog. Subsequently, nine of the eighteen amusics undertook a 10-session training program on pitch direction identification in /ma/ and piano tone. Compared with those untrained, trained amusics demonstrated significantly improved thresholds for pitch direction identification in speech syllables [pretest: 2.92 (4.63) semitones; posttest: 0.16 (0.06)], and marginally significant improvement in pitch direction identification thresholds for piano tones [pretest: 5.20 (4.56); posttest: 0.89 (1.46)]. These findings suggest that pitch sensitivity of individuals with congenital amusia could be improved through auditory training, providing evidence for neural plasticity in the amusic brain, which may lead the way to other rehabilitative programs for treating this musical disorder.

4:30

2pPP14. Neuromagnetic beta-band oscillation for rhythmic processing induced by subjectively accented structure. Takako Fujioka (CCRMA (Ctr. for Comput. Res. in Music and Acoust.), Dept. of Music, Stanford Univ., 660 Lomita Dr., Stanford, CA 94305-8180, takako@ccrma.stanford.edu), Laurel J. Trainor (Dept. of Psych., Neurosci. and Behavior, McMaster Univ., Hamilton, ON, Canada), and Bernhard Ross (Rotman Res. Inst., Baycrest, Toronto, ON, Canada)

Musical rhythm facilitates synchronized body movements and schema-based, predictive timing perception. Our previous magnetoencephalography (MEG) study demonstrated that beta-band (~20 Hz) activity in bilateral auditory cortices shows synchronized modulation that predicts the time point of the next beat (Fujioka *et al*. 2009, 2012). Furthermore, after finger tapping to a different musical meter such as a march or waltz (every 2nd or 3rd beat), the broadband evoked response from auditory cortex differentiates the metric conditions (Fujioka *et al*., 2010). Here we examined how beta-band activity indexed subjective metrical perception during listening to unaccented beats (1) after listening to acoustically accented beats, and (2) after finger tapping with either the left or right index finger. The auditory cortices showed beat-synchronized modulation in line with the previous studies in both march and waltz conditions. However, distinction between down-beat and up-beat positions was stronger in march than in waltz condition, with symmetrical activities in the left and right auditory cortices. This contrast between the two metric conditions was stronger in the side contralateral to the tapping finger in the tapping condition. This suggests that contribution of auditory cortices to metric processing depends on both its timing structure and rhythmic movement in the contralateral hemisphere.

4:45

2pPP15. On the learnability of auditory concepts. Ronaldo Vigo, Mikayla Barcus, Yu Zhang, and Charles Doan (Ohio Univ., 200 Porter Hall, Athens, OH 45701, vigo@ohio.edu)

The field of categorization and concept learning research has been dominated by findings involving visual stimuli. Among these findings is the learning difficulty ordering of the family of category structures associated with visual categorical stimuli consisting of four objects defined over three dimensions. This ordering has been observed numerous times in several rigorous studies (Kruschke, 1992; Love *et al*., 2004; Nosofsky *et al*., 1994; Shepard *et al*., 1961; Vigo, 2011a, 2013a. 2013b) and has been influential in shaping current theories of conceptual behavior. In recent work, we have freed the field from this visual bias by examining the learnability of auditory categorical stimuli that are instances of the aforementioned structures. We found that, in general, for auditory categorical stimuli the learning difficulty ordering of these structures is somewhat different from that of their visual counterparts. However, we also found that this difference may be explained and accurately predicted by a simple encoding mechanism proposed in generalized invariance structure theory (GIST; Vigo, 2013b). We view this result as evidence in support of the proposition that a single basic conceptual system underlies the acquisition of both auditory and visual concepts.

Session 2pSA

Structural Acoustics and Vibration: Computational Structural Acoustics

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Chair's Introduction—1:00

Invited Papers

1:05

2pSA1. Material identification in frequency-domain coupled acoustic-structure interaction using an error in constitutive equation functional. Wilkins Aquino, James Warner, and Manuel Diaz (Civil and Environ. Eng., Duke Univ., Hudson Hall, Durham, NC 27708, wa20@duke.edu)

This work focuses on the inverse identification of linear elastic material parameters in the context of frequency-domain, coupled acoustic-structure interaction. The approach postulates the inverse problem as an optimization problem where the solution is obtained by minimizing a modified error in constitutive equation (MECE) functional. The MECE functional measures the discrepancy in the constitutive equations that connect kinematically admissible strains and dynamically admissible stresses while incorporating the measurement data as an additional quadratic penalty term. The method is formulated generally for the recovery of material properties in a coupled acoustic-structure system using solid displacement and/or fluid pressure measurement data. Numerical results demonstrate that the proposed methodology can identify the spatial distribution of elastic moduli from partial and noisy measurements taken in either the fluid or solid subdomains.

1:25

2pSA2. A parallel Helmholtz solver for acoustics and structural acoustics. Clark R. Dohrmann and Timothy F. Walsh (Computational Solid Mech. & Structural Dynam., Sandia National Labs., P.O. Box 5800, M.S. 0380, Albuquerque, NM 87185-0380, crdohrm@sandia.gov)

In this talk, we present a parallel Helmholtz solver for the solution of acoustic and structural acoustic problems in the frequency domain. The solver is based on overlapping Schwarz domain decomposition concepts for parallelism, and artificial structural damping is introduced in the preconditioner to deal with resonant or near resonant conditions at the subdomain level. An efficient method to solve problems over a range of frequencies is also described. Numerical examples are presented for acoustic and structural acoustic problems, as well as a study of solver performance for higher order finite element discretizations.

1:45

2pSA3. Computational methods for the interior structural acoustics of small spaces. Karl Grosh, Yizeng Li (Dept. of Mech. Eng., Univ. of Michigan, 2350 Hayward St., Ann Arbor, MI 48109-2125, grosh@umich.edu), and Robert Littrell (Baker-Calling, Inc., Santa Monica, CA)

In some biomechanical systems and micro-electro-mechanical systems (MEMS), the interaction of a viscous compressible fluid confined in a space bounded in part by a flexible structure is of central importance. Two specific examples are MEMS microphones (condenser or piezoelectric) and the cochlea. In both the manmade and biological acoustical sensor, the interior space is typically smaller than an acoustic wavelength, and a successful design involves trade-offs between sensitivity, bandwidth, and noise (including thermal, mechanical, electrical, or channel generated noise); the latter two criteria depend critically on the viscous and thermal forces in the system. A direct numeric approach to modeling viscous and thermal effects is often prohibitively expensive, as boundary layers must be resolved in the mesh. In this talk, we will present approximate methods that enable the inclusion of viscothermal effects in a computational framework. In particular, a variational approach amenable to inclusion in a finite element based code will be presented. Results for this method, which retains the accuracy of a Navier-Stokes formulation with the computational cost of a standard scalar acoustic formulation, will be given along with the limitations of the method and a discussion of alternative numerical approaches.

Contributed Papers

2:05

2pSA4. Convolution formulations for non-negative intensity in a plane. Earl G. Williams (Acoust. Div., Naval Res. Lab., Code 7106, 4555 Overlook Ave., Washington, DC, DC 20375, earl.williams@nrl.navy.mil)

New spatial convolution formulas for a variant of the active normal intensity in planar coordinates have been derived that use measured pressure or normal velocity near-field holograms to construct a positive-only (outward) intensity distribution in the plane, quantifying the areas of the vibrating structure that produce radiation to the far-field. This is an extension of the outgoing-only (unipolar) intensity technique recently developed for arbitrary geometries by Steffen Marburg. The method is applied independently to pressure and velocity data measured in a plane close to the surface of a point-driven, unbaffled rectangular plate in the laboratory. It is demonstrated that the sound producing regions of the structure are clearly revealed using the derived formulas and that the spatial resolution is limited to a half-wavelength. A second set of formulas called the hybrid-intensity formulae are also derived which yield a bipolar intensity using a different spatial convolution operator, again using either the measured pressure or velocity. Using the experiment results it is shown that the velocity formula yields the classical active intensity and the pressure formula an interesting hybrid intensity that may be useful for source localization. Computations are fast and carried out in real space without Fourier transforms into wavenumber space. [Work supported by the Office of Naval Research.]

2:20

2pSA5. Minimization of the sound radiated by a curved underwater panel. Micah R. Shepherd and Stephen A. Hambric (Appl. Res. Lab, Penn State Univ., PO Box 30, M.S. 3220B, State College, PA 16801, mrs30@psu.edu)

The sound radiated by a curved underwater panel excited by a point drive is minimized using an evolutionary strategy. The panel thickness is varied in incremental strips along the length of the panel to obtain the optimal configuration. The panel weight and sound radiation are minimized simultaneously using a single weighted objective function. The weighting coefficient is then varied to obtain a Pareto front describing the competing nature of the two objectives. The optimal designs are compared for each weighting coefficient illustrating the tradeoff between minimizing weight and radiated sound power. Normal modes and radiation efficiency curved are also compared. When the optimizer favors minimizing weight, the element thickness is uniformly minimized. Conversely, when the optimizer favors minimizing sound power, the optimal design accepts energy less easily and becomes an inefficient radiator.

2:35

2pSA6. Vibroacoustic modeling of ventilation window system with internal partial partitions. Xiang Yu and Li Cheng (Dept. of Mech. Eng., The Hong Kong Polytechnic Univ., Hung Hom, Kowloon 00000, Hong Kong, li.cheng@polyu.edu.hk)

Domestic buildings exposed to traffic and environmental noise has always been a major concern. Among various noise control measures, ventilation windows offer appealing features by simultaneously allowing good sound attenuation and air ventilation. However, the modeling of such systems is challenging in that the simulation tools should cope with the complexity of the system; to reach reasonably high frequency range; and to offer the flexibility needed for system optimization. In the present study, the sound transmission through a double-glazed ventilation window is investigated using the patch transfer function method along with a handy treatment of the air aperture and micro-perforated elements. Numerical results show the sound insulation performance can be improved with internal partial partitions, which can be either solid or micro-perforated.

2:50–3:05 Break

3:05

2pSA7. Sound field alteration through cavity shape design using a Wavelet-Galerkin Method. Su Zhang (Dept. of Mech. Eng., The Hong Kong Polytechnic Univ., Hong Kong, Hong Kong) and Li Cheng (Dept. of Mech. Eng., The Hong Kong Polytechnic Univ., Hung Hom, Kowloon 00000, Hong Kong, li.cheng@polyu.edu.hk)

The general problem of internal sound field prediction along with the manipulation of the cavity boundary shape to alter the sound pressure distribution is dealt with in this paper. Owing to the compactly supported orthogonal property of the wavelet and its extraordinary fitting capability, Daubechies wavelet scaling function is used as a global basis function to expand the unknown sound pressure under the general Galerkin framework. The proposed formulation is shown to offer high accuracy on the numerical calculation of the sound pressure field with the use of a remarkably small number of messing points. A genetic based shape optimization algorithm is proposed and demonstrated. As an example, an enclosure with an inner rigid acoustic screen is investigated. By optimizing the shape of the screen, the sound pressure level within a chosen area is successfully reduced. Results show the remarkable potentials of the proposed approach as a topology optimal tool for the general inner sound field problems.

3:20

2pSA8. A comparison of perfectly matched layers and infinite elements for exterior Helmholtz problems. Gregory Bunting (Computational Solid Mech. and Structural Dynam., Sandia National Labs., 709 Palomas Dr. NE, Albuquerque, NM 87108, gbunting@purdue.edu), Arun Prakash (School of Civil Eng., Purdue Univ., West Lafayette, IN), and Timothy Walsh (Computational Solid Mech. and Structural Dynam., Sandia National Labs., West Lafayette, Indiana)

Perfectly matched layers and infinite elements are commonly used for finite element simulations of acoustic waves on unbounded domains. Both involve a volumetric discretization around the periphery of an acoustic mesh, which itself surrounds a structure or domain of interest. Infinite elements have been a popular choice for these problems since the 1970s. Perfectly matched layers are a more recent technology that is gaining popularity due to ease of implementation and effectiveness as an absorbing boundary condition. In this study, we present massively parallel implementations of these two techniques, and compare their performance on a set of representative structural-acoustic problems on exterior domains. We compare the ability of these methods to absorb acoustic waves on ellipsoidal domains with waves of relatively high oblique angles of incidence. We also examine the conditioning of the linear systems generated by the two techniques by examining the number of Krylov-iterations needed for convergence to a fixed solver tolerance. [Sandia National Laboratories is a multi-program laboratory managed and operated by Sandia Corporation, a wholly owned subsidiary of Lockheed martin Corporation, for the U.S. Department of Energy's National Nuclear Security Administration under contract DE-AC04-94AL850000.]

3:35

2pSA9. A plane wave method for modeling acoustic variables in cavities. Matthew Kamrath (Graduate Program in Acoust., Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, kamrath64@gmail.com) and Gary Koopmann (KCF Technologies, Inc., State College, PA)

The pressure field within a cavity that contains a homogeneous fluid is modeled in the frequency domain as a superposition of N plane waves with arbitrary orientations and unknown complex amplitudes. Specifying the pressure, the normal velocity, or the specific impedance at N locations produces a system of N linear equations with N unknowns. Then, solving for the complex amplitudes yields an approximation of the pressure and velocity fields inside the cavity. Preliminary comparisons to analytic results are presented.

3:50

2pSA10. Tuned elastic shells with matched acoustic impedance and sound speed in water. Alexey Titovich and Feruza Amirkulova (Mech. and Aerosp. Eng., Rutgers Univ., 98 Brett Rd., Piscataway, NJ 08854, alexey17@eden.rutgers.edu)

Tuning of an elastic shell to match the acoustic properties of the surrounding fluid (water) is done with an internal mechanism consisting of a concentered central mass supported by an axisymmetric distribution of elastic stiffeners. The effective impedance and index are both matched at low frequencies (ka $<$ 1) by adjusting the mass and stiffness of the internal oscillator for a given shell thickness. The subsonic flexural modes of the shell, which are excited by the point attachments of the stiffeners, are ultimately suppressed in the low frequency range with a sufficiently large number of stiffeners. As a result, the scattering cross-section of the tuned shell-stiffener-mass system is negligible compared to the empty shell at frequencies below the resonance of the internal oscillator. An optimal shell thickness exists which maximizes the frequency range of water-like behavior. Furthermore, tuning the effective properties of each such shell in an array is used to achieve acoustic transparency and lensing. Inclusion of damping decreases the magnitude of the internal oscillator resonance without significantly affecting the scattered field. Several planar simulations are presented demonstrating the applications.

4:05

2pSA11. Analytical scattering method of flexural waves considering the evanescent field in a thin plate. Sungjin Cho and Junhong Park (Mech. Eng., Hanyang Univ., Haengdang-dong 17, Seongdong-gu, Seoul 133-791, South Korea, sjcho0407@hanyang.ac.kr)

This paper presents a theoretical solution on analyzing the scattering behaviors of the flexural waves by a circular obstacle medium having different material properties. In deriving the exact solution, the evanescent field together with the scattered field is considered. The modified Bessel function of the second kind is used for the evanescent field. As the evanescent wave is exponentially decaying in all directions near the edge of a circular obstacle medium when the mass density and critical angle are discontinuous, the scattering pattern without evanescent field is difficult to analyze the wave scattering phenomenon near the boundary layer. The resulting equations are solved using symbolic calculation of software MATLAB. The dependence of the scattering behaviors on the mechanical properties of the inner medium is investigated. The influence of the evanescent field on the scattering is also investigated in various configurations. The result shows that the scattering pattern has a strong dependence on the evanescent field in near field of a circular obstacle.

4:20

2pSA12. Influence of cross-sectional discontinuity on the damping characteristics of viscoelastically supported rectangular plates. Jeongwon Park, Sangkeun Ahn (Dept. of Mech. Eng., Hanyang Univ., 222 Wangsimni-ro, Seongdong-gu, Seoul 133-791, South Korea, jwparks@hanyang.ac.kr), Ji Woo Yoo (Res. and Development Div., Hyundai-Kia Motors, Hwaseng, Gyeonggi, South Korea), and Junhong Park (Dept. of Mech. Eng., Hanyang Univ., Seoul, South Korea)

Cross-sectional shape of a structure is one of the important design parameter associated with the vibration responses and resulting sound radiation. In this study, the influence of cross-sectional discontinuity on the vibration characteristics of plate structures was investigated. The variation of cross-sectional geometry in the rectangular plate was modeled as the change of bending stiffness for flexural wave propagation analysis. The complex translational and rotational stiffness at plate edges were used for modeling of the damping at the boundaries. The ratio between the incident and reflected waves from the boundaries was predicted for the flexural waves of different wavelengths to analyze the effect of support stiffness on the vibration damping. Using the wave propagation model, the condition of the viscoelastic boundary properties and the discontinuous flexural stiffness of plate for minimum reflection ratio, i.e., maximum dissipation of the vibration was calculated. Modal damping characteristics of the plate with and without the discontinuity in the cross-section were measured and compared to the predicted reflection ratios. The measured damping ratios on the different boundary conditions showed similar pattern with the predicted vibration energy dissipation at the viscoelastic supports.

4:35

2pSA13. Modal analysis of a vibrating string via electromagnetic field excitation. Anton A. Filyayev (Audio Arts & Acoust., Columbia College Chicago, 33 E. Congress Ave., Chicago, IL 60604, anton.filyayev@loop.colum.edu)

An electromagnetic field (EM) generating device ("EBow") was used to excite a fixed string held in tension between two boundaries on a custom-built apparatus in the style of a monochord. A high-speed camera was used to visualize and record the vibration of the string across the length of the string for every EBow position. The data sets were analyzed to create amplitude vs time graphs. Every excitation position was found to result into a different mode of vibration that emphasized fundamental, second, third, and fourth eigenmodes. Each mode resulted in a distinct displacement pattern, which was found to follow curve fits with R<+>2<+> values ranging from 0.94 to 0.99. The recorded amplitudes also agreed with expected nodal and anti-nodal positions for every observed mode signifying the data were robust. A discrete Fourier series expansion was performed on the polynomial fits and yielded simple second-order expressions (including a damping term) which are in agreement with the expected excitation models for a fixed string.

Session 2pSCa

Speech Communication: Speech Technology

Yi Xu, Chair
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Contributed Papers

1:30

2pSCa1. Measurement of formants in synthetic vowels. Christine H. Shadle, Hosung Nam (Haskins Labs., 300 George St., New Haven, CT 06511, shadle@haskins.yale.edu), and D. H. Whalen (Speech and Hearing Sci., CUNY Graduate Ctr., New York, NY)

The measurement of formant frequencies of vowels is among the most common measurements in speech studies, but is known to be biased by the particular fundamental frequency (F0) exciting the formants, and to be inaccurate for formants close together or for speakers using a high F0. To allow a comparison across multiple measurement techniques, vowels were synthesized using the Klatt synthesizer with known formant values. The synthetic vowels were constructed with five different F1 values and nine different F0 values; formant bandwidths, and higher formant frequencies, were constant. The F0s varied in such a way that the most intense harmonic in F1 or F2 either matched the center frequency or deviated in the range of 3–87 Hz. Manual measurements by four subjects were compared to automatic measures using the LPC Burg algorithm, LP closed-phase covariance, and spectra smoothed cepstrally or by averaging repeated DFT's. Formants were also measured from pruned reassigned spectrograms. Error patterns differ among the methods, but most tracked the frequency of the most intense harmonic; the smallest errors occur with closed-phase covariance and reassigned spectrogram. Implications for such measures on vowels in isolated words of real speech are discussed. [Work supported by NIH-NIDCD grant DC-002717.]

1:45

2pSCa2. A further comparison of fundamental frequency tracking algorithms. Hongbing Hu (Intel Corp., Binghamton, New York), Peter Guzewich, and Stephen Zahorian (Elec. and Comput. Eng., State Univ. of New York at Binghamton, PO Box 6000, Binghamton, NY 13902, zahorian@binghamton.edu)

"Yet another Algorithm for Pitch Tracking -YAAPT" was published in a 2008 JASA paper (Zahorian and Hu), with additional experimental results presented at the fall 2012 ASA meeting in Kansas City. The results presented in both the journal paper and at the fall 2012 meeting indicated that YAAPT generally has lower error rates than other widely used pitch trackers (YIN, PRAAT, and RAPT). However, even YAAPT-created pitch tracks had significant "large" errors (pitch doubling and pitch-halving) for both clean and noisy speech. Recently additional post-processing heuristics have been incorporated to reduce the incidence of these type errors—thus reducing the need for hand correcting pitch tracks for situations where extremely accurate tracks are desired. For the case of an all-voiced track, interpolation through unvoiced intervals has been improved. The updated version of YAAPT is presented along with experimental results. The experiments are conducted with multiple databases, including British English, American English, and Mandarin Chinese. For most conditions evaluated, YAAPT gives better performance than the other fundamental frequency trackers.

2:00

2pSCa3. Automated assessment of English fundamental frequency contours for non-native speakers from China and India. Keelan Evanini and Xinhao Wang (Educational Testing Service, Rosedale Rd., M.S. R-11, Princeton, NJ 08541, kevanini@ets.org)

This study investigates the F0 contours produced by non-native speakers of English from two different countries. Speakers from China ($N = 202$; L1 Mandarin) and India ($N = 230$; multiple L1s) read a paragraph out loud in the context of an assessment of non-native English speaking proficiency; in addition, a control set of native speakers ($N = 85$) read the same paragraph. The words and phonemes in all of the responses were provided with time stamps using forced alignment with the prompt text, and mean normalized F0 values were extracted for each word from each speaker. A model F0 template for the paragraph was generated by taking the mean word-level F0 value across all native speakers, and the correlation between the word-level F0 contour and this template was calculated for each non-native speaker (as in Schwanenflugel *et al.* 2004). This correlation was used as a feature for assessing F0 contours and was correlated with expert human scores of English proficiency. The results demonstrated that the feature performed somewhat better for the speakers from India ($r = 0.390$) than the speakers from China ($r = 0.326$). Furthermore, the correlations improved when function words were removed from the analysis and only content words were considered ($r = 0.396$ and $r = 0.346$).

2:15

2pSCa4. Detecting voice disguise from speech variability: Analysis of three glottal and vocal tract measures. Talal B. Amin (Elec. and Electron. Eng., Nanyang Technolog. Univ., Singapore, Singapore), James S. German (Humanities and Social Sci., Nanyang Technolog. Univ., HSS-03-46, Singapore, Singapore, jsgerman@ntu.edu.sg), and Pina Marziliano (Elec. and Electron. Eng., Nanyang Technolog. Univ., Singapore, Singapore)

The deliberate attempt by speakers to conceal their identity (voice disguise) presents a challenge for forensics and for automated speaker identification systems. Using a database of natural and disguised voices of three professional voice impersonators, we build on earlier findings (Amin *et al*., 2012) by exploring how certain glottal and vocal tract measures, including fundamental frequency (f0), glottal timing (Open Quotient), and vowel formants, are exploited to create novel voice identities. Specifically, we explored whether the amount and type of variation exhibited by impersonators can be used to develop a metric for distinguishing natural from disguised voices. As expected, variation in f0 and Open Quotient was speaker-dependent, and corresponded closely to social attributes (i.e., gender/age) of the voice identities involved. In a novel finding, the effects of voice identity on vowel formants were highly dependent on vowel category, and could not be readily characterized as global modifications to the vowel space (Bradlow *et al*.,1996). We therefore developed a no-reference objective metric for voice disguise that treats formant variability on a vowel-by-vowel basis. This metric consistently assigned high rankings to natural voices (3.3/27 on average). This correlated closely with the subjective disguisedness ratings of 18 naïve listeners, even outperforming them slightly.

2:30

2pSCa5. Glottal articulations in tense vs lax phonation contrasts. Jianjing Kuang (Linguist, Univ. of Pennsylvania, UCLA Campbell Hall 3125, Los Angeles, California 90095, kuangjj@gmail.com) and Patricia Keating (UCLA, Los Angeles, CA)

This study explores the glottal articulations of one type of phonation contrast—the tense vs lax phonation contrasts of three Yi (Loloish) languages—which is interesting because neither phonation type is very different from modal voice, and both are independent of the languages' tonal contrasts. Electroglottographic (EGG) recordings were made in the field, and traditional EGG measures showed many small but significant differences between the phonations. Tense phonation involves more overall contact and briefer but slower changes in contact. Functional Data Analysis was then applied to entire EGG pulse shapes, and the resulting first principal component was found to be mostly strongly related to the phonation contrasts, and correlated with almost all the traditional EGG measures. Unlike the traditional measures, however, this component also captures differences in abruptness of contact. Furthermore, previously-collected perceptual responses from native speakers of one of the languages correlated better with this component than with any other EGG measure or any acoustic measure. The articulatory differences between these tense and lax phonations, involving glottal aperture and how glottal closure is made, are not extreme, but apparently they are consistent enough, and perceptually robust enough, to support this linguistic contrast. [Work supported by NSF.]

2:45–3:00 General Discussion

3:00–3:30 Break

3:30

2pSCa6. The effect of non-linear dimension reduction on Gabor filter bank feature space. Hitesh A. Gupta, Anirudh Raju, and Abeer Alwan (Elec. Eng., Univ. of California Los Angeles, 550 Veteran Ave., Apt. 102, Los Angeles, CA 90024, hiteshag@ucla.edu)

In this paper, we modify the Gabor feature extraction process, while applying the Gabor filters on the power-normalized spectrum and concatenating with power normalized cepstrum coefficients (PNCC), for noise robust large vocabulary continuous speech recognition. In Chang *et al*., ICASSP (2013), a similar Gabor filter bank (GBFB) feature set with multilayer perceptron (MLP) processing (to reduce the feature dimension) has been used with mel frequency cepstrum coefficients showing improvements on Aurora-2 and renoised Wall Street Journal corpora. On a subset of the Aurora-4 database (only male), our method has shown promising results (when using PCA) being 7.9% better than 39-dimensional PNCC features. But, the GBFB features are a rich representation of the speech spectrogram (as an overcomplete basis), and an appropriate dimension reduction/manifold learning technique is the key to generalizing these features for the large vocabulary task. Hence, we propose the use of Laplacian Eigenmaps to obtain a reduced manifold of 13 dimension (from a 564-dimensional GBFB feature set) for the training dataset with a MLP being used to learn the mapping so that the same can be applied to out-of-sample points, i.e., the test dataset. The reduced GBFB features are then concatenated with the 26-dimension PNCC plus acceleration coefficients. This technique should lead to better accuracies as speech lies on a non-linear manifold rather than a linear feature space. [This project was supported in part by DARPA.]

3:45

2pSCa7. Language material for English audiovisual speech recognition system development. Andrzej Czyzewski (Multimedia Systems Dept., Gdansk Univ. of Technol., Narutowicza 11/12, Gdansk 80-233, Poland, ac@pg.gda.pl), Tomasz Ciszewski, Dorota Majewicz (Faculty of Philology, Univ. of Gdansk, Gdansk, Poland), and Bozena Kostek (Multimedia Systems Dept., Gdansk Univ. of Technol., Gdansk, Poland)

The bi-modal speech recognition system requires a 2-sample language input for training and for testing algorithms which precisely depicts natural English speech. For the purposes of the audio-visual recordings, a training data base of 264 sentences (1730 words without repetitions; 5685 sounds) has been created. The language sample reflects vowel and consonant frequencies in natural speech. The recording material reflects both the lexical word frequencies and casual speech sound frequencies in the BNC corpus of approx. 100m words. The semantically and syntactically congruent sentences mirror the 100m-word corpus frequencies. The absolute deviation from source sound frequencies is.09% and individual vowel deviation is reduced to a level between 0.0006% (min.) and 0.009% (max.). The absolute consonant deviation is 0.006% and oscillates between 0.00002% (min.) and 0.012% (max.). Similar convergence is achieved in the language sample for testing algorithms (29 sentences; 599 sounds). The post-recording analysis involves the examination of particular articulatory settings which aid visual recognition as well as co-articulatory processes which may affect the acoustic characteristics of individual sounds. Results of bi-modal speech elements recognition employing the language material are included in the paper.

4:00

2pSCa8. Characteristics of automatic and human speech recognition processes. Mark VanDam (Speech & Hearing Sci., Washington State Univ., PO Box 1495, Spokane, WA 99202, mark.vandam@wsu.edu) and Noah H. Silbert (Commun. Sci. & Disord., Univ. of Cincinnati, Cincinnati, OH)

In a previous report [VanDam and Silbert (2013) *POMA***19**, 060006], we investigated performance of a commercially available automatic speech recognition (ASR) system [LENA Research Foundation, Boulder, CO] on acoustic recordings from family speech in naturalistic environments. We found that the ASR more accurately labeled children over adults and fathers over mothers, and human judge labels included substantial individual variation. The present work extends previous work by investigating the possible sources for both machine- and human labeling decisions. Classification tree models were fit to several acoustic variables for machine- and human labels of *CHILD*, *MOTHER*, and *FATHER*. Results suggest that (a) fundamental frequency (f_0) and duration measures influenced label assignment for both machine and human classifications, (b) the error of the fitted models is lower for the machine labeling procedure than for human judges, (c) machine- and human decision processes use the acoustic criteria (i.e., f_0 and duration) differently, and (d) f_0 is more important than duration for all labelers. Results may have implications for improving implementation and interpretation of ASR techniques, especially as they are useful for understanding child language applications and very large, naturalistic datasets that demand unsupervised ASR techniques.

4:15

2pSCa9. Crying for help: The Frye hearing and forensic acoustic analyses in State of Florida vs George Zimmerman. Al Yonovitz (The Univ. of Montana, Dept. of Communicative Sci. and Disord., Missoula, MT 59812, al.yonovitz@umontana.edu), Herbert Joe (Yonovitz and Joe, LLP, Irvine, CA), and Joshua Yonovitz (Yonovitz and Joe, LLP, Missoula, Montana)

Neighborhood watch volunteer George Zimmerman observed suspicious activity and called police. In minutes, he and Trayvon Martin were in an altercation when Mr. Zimmerman shot and killed Trayvon Martin and claimed self defense. The State's audio experts evaluated the 9-1-1 audio recordings to determine who spoke which background phrases, if any. A Frye hearing is where the court determines the reliability and admissibility of an expert's opinion by determining whether an expert's methodologies are generally accepted within the relevant scientific community. The judge in this case ruled that the State's two audio experts would not be permitted to testify during the trial. While the judge found the "aural perception and spectral analysis... are sufficiently established to have gained general acceptance within the scientific community," she took exception to how they were applied in this case. This case provided a venue and forum for opinions on voice analysis, methodologies, standards, and the quality required of evidentiary audio for determination of speaker identification and elimination. Positions taken by witnesses, a critique of the Frye Hearing and the scientific basis for witness and legal conclusions will be discussed.

4:30

2pSCa10. Intensity slopes as robust measure for distinguishing glottalic vs pulmonic stop initiation. Sven Grawunder (Dept. of Linguist, Max Planck Inst. for Evolutionary Anthropology, Deutscher Platz 6, Leipzig 04103, Germany, grawunder@eva.mpg.de)

A novel cross-linguistically robust measure is introduced for the linguistically relevant distinction of pulmonic vs glottalic (ejective) stops.

We propose to parametrize the abruptness of the following vowel onset by using intensity slope (RMS-trajectory) at the voicing onset. This measure was previously discussed only for voicing distinction of pulmonic stops (Harrington, 2012). The dependencies on vowel quality of the following vowel, voice onset time (VOT), phonetic prominence and speaking rate are investigated on a small-scale sample of two speakers from Avar (Nakh-Dagestanian), Ingush (Nakh-Dagestanian), and Georgian (Kartvelian). The results demonstrate robust significant differences of intensity slopes between pulmonic and glottalic stops, the latter showing the steepest slopes/the most abrupt onsets. In some cases of prevoiced stops, the intensity slopes allow even a tripartite distinction of (pre-)voiced, voiceless (aspirated), and ejective stops. However, in order to countervail the influence of vowel specific onset characteristics (e.g., degree of lip rounding) or the influence of VOT, the sample must be controlled for vowel quality and place of articulation. And, since higher speaking rates and less prominent syllables involve higher rates of laryngealized (creaky) vowel onsets, a breakdown of the abruptness distinction is observed under these conditions.

4:45–5:00 General Discussion

TUESDAY AFTERNOON, 3 DECEMBER 2013 PLAZA A, 1:00 P.M. TO 5:00 P.M.

Session 2pSCb

Speech Communication: Speech Perception II (Poster Session)

Meghan Sumner, Chair
Dept. of Linguist., Stanford Univ., Margaret Jacks Hall, Bldg. 460, Stanford, CA 94305-2150

Contributed Papers

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.

2pSCb1. Perception of Scottish Gaelic alternating (leniting) consonants. Natasha L. Warner (Dept. of Linguist, Univ. of Arizona, Box 210028, Tucson, AZ 85721-0028, nwarner@u.arizona.edu), Ian Clayton (Dept. of English, Boise State Univ., Boise City, ID), Andrew Carnie, Muriel Fisher, Dan Brenner, Michael Hammond, Diana Archangeli, and Adam Ussishkin (Linguist, Univ. of Arizona, Tucson, AZ)

Scottish Gaelic, an endangered Celtic language, demonstrates alternations in word-initial consonants, as in "pòg" [p^hok] 'kisses' vs "phòg" [fok] 'kissed.' This process, called lenition, leads to apparent neutralizations of Gaelic segments, for example of the [f] of " phòg" with [f] of "foghlam" [foɫəm] "education," which is not caused by lenition. A perception experiment can show whether listeners hear any residual difference between lenited segments (e.g., [f]<-[p^h]) and the phonetically similar segments ([f]<-[f]). This project used a gating study to investigate when in the word listeners determine which type of sound they are hearing. Preliminary results from 17 native Gaelic listeners indicate that listeners cannot distinguish lenited from phonetically matched consonants (e.g., the two types of [f]) from cues in the consonant itself, but can distinguish both from the unlenited phonologically matched consonants (e.g., [p^h]) very accurately. Listeners become able to distinguish lenited from phonetically matched segments (the two types of [f]) during either the following vowel or the segment after that, depending on what coarticulatory cues with the latter parts of the word are available. Thus, listeners need enough acoustic information to provide lexical disambiguation in order to determine the morphological source of lenited sounds.

2pSCb2. An online investigation of compensation for coarticulation in Mandarin learners of English. David Li (Linguist, Univ. of Southern California, 2133 Sun Ridge Dr., Chino Hills, CA 91709, li.david.c@gmail.com) and Elsi Kaiser (Linguist., Univ. of Southern California, Los Angeles, CA)

Phonological variation can cause lexical ambiguity: The word "run" may sound like "rum" in sentences like "A quick run picks you up" (Gaskell and Marslen-Wilson, 2001) because /n/ is assimilated to /m/ due to the influence of /p/. Previous studies have shown that listeners compensate for coarticulation, perceiving acoustically identical sounds differently in different contexts (e.g., Mann, 1980). We investigated whether native Mandarin speakers with English as a second language (L2) compensate for assimilation in the same manner/to the same extent as native English speakers (L1). Given that there are no Mandarin words with the nasal coda [m], do Mandarin learners of English compensate for phonological variation not present in their L1? We conducted a visual-world eye-tracking study—using English stimuli—to investigate compensation-for-assimilation in L1 English and L1 Mandarin speakers. Identical acoustic sequences were embedded in carrier sentences with fast vs slow speech rates to test interpretation of potentially ambiguous words (cf. rum/run) by L1 and L2 speakers. Eye-movements reveal real-time differences in how L1 and L2 speakers interpret coarticulated speech, with L1 speakers showing signs of rapid compensatory processes. We also discuss how listeners' interpretation is influenced by other acoustic cues in coarticulated speech.

2pSCb3. Classification of affricate burst place in consonant-vowel contexts in English. Jung-Won Lee, Hong-Goo Kang (Elec. and Electron. Eng., Yonsei Univ., 134 Shinchon-dong, Seodaemun-gu, Seoul 120-749, Korea, Republic of, jaesuk2002@dsp.yonsei.ac.kr), and Jeung-Yoon Choi (Res. Lab. of Electron., Massachusetts Inst. of Technol., Cambridge, MA)

This study investigates characteristics of affricate burst place of articulation compared with the bursts for the three places of articulation for stops (labial, alveolar, and velar) in English. The data comprise consonant-vowel tokens in the TIMIT corpus. To assess which stop place of articulation may be used to simultaneously model affricate bursts, Jensen-Shannon divergence measures are found for probability distributions of acoustic-phonetic features. In addition, we conduct classification experiments using combinations of acoustic-phonetic features and Mel-frequency cepstral coefficients (MFCCs), to see how well affricate burst place is classified using models for the three stop places. The experimental results show that although affricate place is similar to the alveolar place of articulation for stops, a separate post-alveolar place for affricate burst provides a better model. The results suggest that a separate affricate place model will be useful in a feature-based speech recognition system that explicitly detects place of articulation for consonants.

2pSCb4. Effect of consonantal place of articulation on the perception of phonetic voicing in plosives. Viktor Kharlamov and Anna Loukianova (Linguist, Univ. of Arizona, Box 210028, Tucson, AZ 85721, kharlamov@email.arizona.edu)

For plosive consonants, production of phonetic voicing often varies across consonantal places of articulation. Anterior stops (e.g., /b/, /d/) tend to show more glottal pulsing compared to the plosives produced towards the back of the oral cavity (e.g., /g/), which is traditionally attributed to the smaller volume of air passing through the glottis during the closure stage of posterior stops. The current work demonstrates that a similar effect of consonantal place of articulation also exists in perception. Results of a series of identification experiments show that English listeners ($n = 60$) are more sensitive to glottal pulsing in word-final dorsals than labials or coronals. The effect is especially robust when, within each experimental block, listeners are exposed to variability in consonantal place of articulation (i.e., when labial, coronal, and dorsal plosives are blocked together) but not to variability in phonetic vowel duration (i.e., when a given block contains only short or long vowels). This suggests that glottal pulsing is more perceptually salient in posterior plosives, for which voicing is less expected in production, and that the sensitivity to voicing in dorsals is stronger when listeners can do an online comparison of voicing cues across different places of articulation.

2pSCb5. Discrimination between fricatives and affricates pronounced by Japanese native speakers at various speaking rates. Kimiko Yamakawa and Shigeaki Amano (Human Informatics, Aichi Shukutoku Univ., 9 Katahira, Nagakute, Aichi 4801197, Japan, jin@asu.aasa.ac.jp)

Fricatives [s] and affricates [ts] uttered at a normal speaking rate are successfully discriminated with two variables: a rise part duration and a steady + decay part duration [Yamakawa *et al*., Acoust. Sci. Tech. **33**(3), 154–159 (2012)]. This study examined whether [s] and [ts] uttered at various speaking rates are also well discriminated with these variables. Discriminant analyses with the two variables were performed on [s] and [ts] in word-initial position in a carrier sentence pronounced by eight native Japanese speakers at fast, normal, and slow speaking rates. Discriminant error rates were low for the fast (9.8%, $n = 512$), normal (5.7%, $n = 512$), and slow (13.1%, $n = 512$) speaking rates. However, the error rate was high (22.0%, $n = 1536$) when the three speaking rates were analyzed together. It decreased to 15.7% when the two variables were normalized with an averaged mora duration of the carrier sentence. These results suggest that [s] and [ts] can be discriminated at various speaking rates with a normalized rise part duration and normalized steady + decay part duration. [This work was supported by JSPS KAKENHI Grant Numbers 22720173, 24652087, and 25284080, and by Aichi Shukutoku University Special Research Grant 2011-2012 and Corporate Research Grant 2013-2014.]

2pSCb6. Relationship of training of pitch-pattern reconstruction ability to speech perception in normal-hearing young adults. Kristen M. Cortese (Commun. Disord. & Sci., Rush Univ., 1001 West Cypress Dr., Arlington Heights, IL 60005, kristen_cortese@rush.edu), Stanley Sheft, Valeriy Shafiro, and Derek J. Stiles (Commun. Disord. & Sci., Rush Univ., Chicago, IL)

Previous research has indicated that processing of pitch changes is related to speech perception. Current work investigated the relationship of pitch-pattern training to speech perception in young normal-hearing adults. The training protocol was based on an interactive pattern-reconstruction task in which listeners assembled four or five tones (frequency: 400–1750 Hz, duration: 75–600 ms) to match a random target sequence. Control and experimental groups, of 13 subjects each, were tested. The study included a pretest, three training sessions (experimental group only), and a posttest. Training consisted of three 40-min sessions of interactive pitch-pattern reconstruction. In the pre- and posttest, listeners also reconstructed patterns of sinewave speech, and completed measures of speech perception which included sinewave speech and speech-in-noise tasks. Despite training with only tonal patterns, results showed significant improvement in the ability to reconstruct patterns for both tonal and sinewave-speech stimuli. Significant improvements were obtained for intelligibility of sinewave speech between pre- and posttest, which were greater for the experimental than control group. A greater relationship between results from the intelligibility and pattern-reconstruction conditions post training was also found. Overall, results suggest that the training protocol may benefit speech perception, especially in conditions of degraded speech. [Work supported by NIDCD.]

2pSCb7. Comprehending speech at artificially induced rates. Lucia da Silva (Linguist, Univ. of Br. Columbia, 6335 Thunderbird Crescent Box 137, Vancouver, BC V6T2G9, Canada, lucia@alumni.ubc.ca), Adriano V. Barbosa (Dept. of Electronics, Federal Univ. of Minas Gerais, Belo Horizonte, Brazil), and Eric Vatikiotis-Bateson (Linguist., Univ. of Br. Columbia, Vancouver, BC, Canada)

This study confirms and extends Fairbanks, Guttman and Miron's study (1957), where comprehension of time-compressed (2:1), recorded speech was improved when played back twice. Specifically, we examine the perception of pitch corrected and non-pitch corrected compressed speech when presented twice in succession and twice with delays of minutes between stimulus presentations. Additionally, we investigate whether visual speech information aids or hinders comprehension by comparing fast rate effects when presented acoustically and audiovisually. A motivation for this study is the increasing dependency on web-based multimedia recordings for knowledge transfer.

2pSCb8. Perceptual interactions among components of a spectral-domain voice source model. Marc Garellek (Linguist. Dept., UCSD, 9500 Gilman Dr. #0108, La Jolla, CA 92093-0108, mgarellek@ucsd.edu), Robin A. Samlan, Jody Kreiman, and Bruce Gerratt (Dept. of Head and Neck Surgery, UCLA, Los Angeles, CA)

A psychoacoustic model of the source spectrum has been proposed in which source contributions to overall voice quality can be quantified by four spectral slope components: H1-H2 (the amplitude difference between the first and second harmonics), H2-H4, H4-2000 Hz (i.e., the harmonic nearest to 2000 Hz), and 2000–5000 Hz. The natural variability of these components has been described, along with the just noticeable differences (JNDs) for each component. The goals of this study are to identify how perceptual sensitivity to each component slope varies as a function of the adjacent slope(s). The JNDs were obtained for stimuli based on synthetic copies of one female voice. The stimuli were manipulated so that spectral slope varied in 0.5 dB increments (1 dB for 2000–5000 Hz) for a particular component at "high" and "low" values of the adjacent slopes. Thirty-three listeners completed an adaptive up-down paradigm. Preliminary results suggest that sensitivity to a particular spectral component varies as a function of some adjacent components more than the other. Furthermore, sensitivity varies based on whether the adjacent component has a high or low slope. These interactions among spectral components will be interpreted with respect to the variability in spectral configuration observed in 144 voices.

2pSCb9. Lexical segmentation of speech from energy above 5 kHz. A. Davi Vitela (Univ. of Arizona, 1131 E. 2nd St., Tucson, AZ 85721, adv1@email.arizona.edu), Brian B. Monson (DUKE-NUS, Singapore, Singapore), and Andrew J. Lotto (Univ. of Arizona, Tucson, AZ)

Research in the field of speech perception has traditionally focused on the acoustic information or cues present in the frequency region below 5 kHz, thus ignoring high frequency energy (HFE). A recent series of studies, however, demonstrated that listeners could determine the mode of production (speech or singing) and further, could identify what was being spoken or sung from the HFE alone and with a speech-shaped masking noise in the lower frequencies. This begs the question as to what types of information listeners are extracting to guide their perception. The current study examined the ability of listeners to transcribe short semantically-unpredictable but syntactically-well-formed spoken phrases that were high-pass filtered at 5.6 kHz. Of particular interest was the ability of some listeners to correctly determine the placement of word boundaries even without the availability of low-frequency information. These findings add to a growing literature on the linguistically relevant information present in higher frequency regions. Results will be framed within current theories of acoustic signatures to lexical segmentation. [Work supported by NIH-NIDCD.]

2pSCb10. Prosodic disambiguation of Korean relative clause attachments. Younah Chung (Linguist., UCSD, UCSD Linguist Dept., 9500 Gilman Dr., La Jolla, CA 92093-0108, yachung@ucsd.edu) and Chongdok Kim (Inst. of Lang. and Information Studies, Yonsei Univ., Seoul, South Korea)

Korean relative clauses do not have to be adjacent to the substantives they modify; genitive noun phrases can interpose between the relative clause and the substantive. Thus, the syntactic analysis of a given relative clause is inherently ambiguous, such that the relative clause can modify either the head noun of a genitive noun phrase or the complex noun phrase as a whole. However, in spoken language, one rarely finds this type of ambiguity even though the relative clause could modify either of the two nouns. This suggests that speakers use prosody to convey intended meaning, with different prosodic cues associated with different syntactic boundaries, thus, providing information necessary for disambiguation. We examined the clarification of surface structure ambiguity in Korean relative clause attachments via prosody. The survey confirmed that native speakers think relative clauses are surface ambiguous. The acoustical analysis revealed that prosodic disambiguation consists of pause insertion, pitch raising, and final lengthening at the syntactic boundary. Regarding the relative importance of different prosodic measures used by speakers, we discovered that pitch raising was a more powerful measure than final lengthening in prosodic disambiguation, and that pausing played the least important role in the parsing of an ambiguous relative clause.

2pSCb11. The effect of phonetic orthography on the perception of Mandarin syllables. Yu-Jung Lin, Chung-Lin Yang (Linguist., Indiana Univ., 800 N Union St. Apt. 405, Bloomington, IN 47408-2230, lin41@indiana.edu), and Chien-Jer Charles Lin (East Asian Lang. and Cultures, Indiana Univ., Bloomington, IN)

It has been found that vowels have perceptual advantage over tones (e.g., Ye and Connie, 1999) and that phonetic orthographic information can be activated in speech perception (e.g., Frauenfelder, Segui, and Dijkstra, 1990). In the present study, we used a tone-vowel detection task where Zhuyin (Taiwanese) and Pinyin users (Mainland Chinese) were asked to monitor Mandarin syllables containing [i4] to test how tonal and vowel information are processed in syllable monitoring, and to examine if Zhuyin and Pinyin users' responses are affected by their phonetic orthographic systems. The results demonstrated that in both groups, vowel-mismatched syllables (e.g., bu4) were rejected faster than tone-mismatched syllables (e.g., bi2). Moreover, the RT of rejecting vowel-mismatched syllables and double-mismatched syllables (e.g., bu2) were similar, indicating that as long as vowel was mismatched, tonal information caused little difference in RT when rejecting the mismatched syllables. This finding was consistent with Ye and Connine (1999). Furthermore, both groups resort to their phonetic orthography in simple syllables (e.g., bi4, zhi4). However, in complex syllables (e.g., biao4), different patterns were observed between the two groups, possibly due to the factors such as how tones are marked in their phonetic orthography or how they learned the orthography.

2pSCb12. Perceiving politeness from speech acoustics alone: A cross-linguistic study on Korean and English. Bodo Winter (Cognit. and Information Sci., Univ. of California, Merced, 3681 San Jose Ave., Apt. 4, Merced, CA 95348, bodo@bodowinter.com), Lucien Brown, Kaori Idemaru (East Asian Lang. & Literatures, Univ. of Oregon, Eugene, OR), and Sven Grawunder (Linguist., Max Planck Inst. for Evolutionary Anthropology, Leipzig, Germany)

Politeness is a crucial aspect of everyday speech communication; however, there are to date only few acoustic studies on this topic. Winter and Grawunder (2012) showed that for Korean speakers, politeness is reflected in pitch, intensity, voice quality and speaking rate. Here, we extend this production study by investigating whether Korean and English listeners can perceive the intended politeness of short Korean utterances based on speech acoustics alone. In two experiments with a total of 47 English and 30 Korean listeners, we found that both groups can detect the intended politeness purely based on the phonetic qualities of speech. In one experiment, accuracy was low (Korean: 58%, English: 53%) because speakers heard multiple voices in a randomized fashion, not allowing them to familiarize with any particular voice. In a design that was blocked by speaker voice, accuracy was higher (Korean: 70%, English: 58%), showing that vocal politeness can be used as a cue when the voice is known. This shows that politeness is not only expressed by honorific lexical forms commonly employed in Korean, but also by speech acoustics. It is remarkable that English speakers performed above chance at all, pointing to cross-linguistic regularities in how politeness is expressed vocally.

2pSCb13. The role of vowels in determining a male speaker's sexual orientation. Erik C. Tracy (Psych., Univ. of North Carolina Pembroke, PO Box 1510, Pembroke, NC 28372, erik.tracy@uncp.edu)

Prior research (Tracy and Satariano, 2011) found that listeners primarily use the vowel in a monosyllabic word to determine the speaker's sexual orientation. Given this result, if listeners were presented utterances that contained multiple vowels, then the accuracy of their sexual orientation ratings should increase. In Experiment 1, listeners were presented with trisyllabic words ("authentic"). They heard the first syllable (/ə/), the first two syllables (/əθɛn/), and the entire word (/əθɛntɪk/). Sexual orientation ratings were the most accurate for the entire word and the least accurate for the first syllable. These results were replicated in experiment 2, which used only bisyllabic words. An items analysis from both experiments revealed that a syllable that contained more phones resulted in higher accuracy ratings compared with a syllable that contained fewer phones. For example, listeners' sexual orientation ratings were more accurate when they heard the first syllable of "lavender" compared to the first syllable of "authentic." These findings suggest that when the number of vowels is kept constant, listeners' responses are more accurate if the utterance contains additional phones.

2pSCb14. Acoustic features mediating height estimation from human speech. John Morton (Psych., Washington Univ., 2309 Laurenwood Dr., Chesterfield, MO 63017, jrmhvc333@yahoo.com), Mitchell Sommers (Psych., Washington Univ., St. Louis, MO), Steven Lulich (Linguist, Indiana Univ., Bloomington, IN), Abeer Alwan, and Harish Arsikere (Eng., UCLA, Los Angeles, CA)

The current experiment was aimed at providing the first direct evidence regarding the role of subglottal resonances in height discrimination. Past research has investigated whether or not listeners are able to discriminate which of two talkers is taller (Rendall *et al.*, 2007), but has not established what parameters of the acoustic speech signal listeners use to distinguish speaker height. In the current study, we examined the role of subglottal resonances in height discrimination. Subglottal resonances generally refer to resonances of the lower airways starting in the lungs and terminating at the glottis. Subglottal resonances would be a good candidate for use in height estimation because they remain relatively stable across vowels and are relatively unaffected by other factors influencing the supra-laryngeal vocal tract; instead they are influenced nearly exclusively by the lower airway acoustic properties. Listeners participated in two tasks, the first task was a two-alternative forced choice (2AFC) height discrimination test, in which listeners heard sentences produced by two talkers of the same gender and were asked to determine which of the two was the tallest. The second task involved listening to five talkers (all same gender) sentences, and then ranking those individuals from tallest to shortest. Findings indicate that listeners are able to discriminate and rank speakers heights better than chance, and the role of subglottal resonances will be reported.

2pSCb15. Lexically biased perceptual adaptation. Michael McAuliffe (Linguist., Univ. of Br. Columbia, Totem Field Studios, 2613 West Mall, Vancouver, BC V6T 1Z4, Canada, mcauliff@interchange.ubc.ca)

Listeners are able to rapidly shift their acoustic-phonetic categories in the face of speaker-specific variation. However, an open question is how top-down expectations can influence this perceptual adaptation. Under Adaptive Resonance Theory [Grossberg (2003)], if an acoustic token provides initial activation of a linguistic category, top-down expectations about that category can boost the resonance and lead to activation of the category even with an ambiguous token. Ambiguous tokens in positions with stronger expectations are hypothesized to lead to less perceptual adaptation than in positions with lower expectations. To test this hypothesis, two groups of participants will be exposed to 10 ambiguous s-ʃ sounds during a lexical decision exposure task. The groups differ in where in the word these sounds will occur, either in the onset or coda of CVC words. Lexical bias, as a form of top-down expectations, is stronger for ambiguous coda sounds [Pitt and Szostak (2012)]. I predict that in a s-ʃ categorization task following exposure, the coda group will differ less than the onset group from a control group that only completes the categorization task, indicative of less perceptual adaptation.

2pSCb16. Degraded word recognition in isolation vs a carrier phrase. Kathy M. Carbonell and Andrew J. Lotto (Speech Lang. & Hearing Sci., Univ. of Arizona, 1131 E 2nd St., Tucson, AZ 85721, kathyc@email.arizona.edu)

Recognizing a spoken word presented in isolation is a markedly different task from recognizing a word in a carrier phrase. The presence of a carrier phrase provides additional challenges such as lexical segmentation but also provides additional information relevant to word recognition such as speaking rate and talker-specific spectral characteristics. The current set of studies is part of an attempt to determine how target word recognition differs in isolation versus in a carrier phrase. In an initial experiment, a set of 220 spoken CVC words were noise-vocoded (6 channel) and presented to listeners either in isolation or following a noise-vocoded carrier phrase—"The next word on the list is…" The target words were transcribed in each condition and scored for initial consonant accuracy and overall word accuracy. Despite the lack of semantic or syntactic information provided by the carrier phrase accuracy for both word and consonant recognition were much higher in the carrier phrase context. A second experiment, used different talkers for the carrier phrase and target word. A smaller but significant benefit over isolation was present for these mixed talker stimuli suggesting that the benefits of the carrier phrase include, but are not limited to, talker-specific information.

2pSCb17. Differences in the recognition of careful and casual speech. Meghan Sumner, Jeremy Calder, Annette D'Onofrio (Dept. of Linguist, Stanford Univ., Margaret Jacks Hall, Bldg. 460, Stanford, CA 94305, sumner@stanford.edu), Kevin B. McGowan (Dept. of Linguist, Stanford Univ., Stanford, Michigan), and Teresa Pratt (Dept. of Linguist, Stanford Univ., Stanford, CA)

Previous work in spoken word recognition and speech perception has shown two seemingly conflicting patterns. While some studies have shown a processing benefit for more frequent word variants (i.e., in a casual speech mode), others have found a benefit for more canonical word forms (i.e., in a careful speech mode). This study aims to reconcile these findings, proposing that different types of processing apply to each speech mode –top-down processing for casual speech, and bottom-up for careful speech. Listeners in an auditory priming task heard natural (non-spliced) sentences spoken in either a careful or casual speech mode. The final word of the auditory prime was either semantically predictable from the preceding sentence context or unpredictable. After the audio prime, listeners responded in a lexical decision task to a visual probe: either the final word heard in the prime, an unrelated word, or a nonword. Preliminary results suggest that, regardless of speech style, reaction times are faster for related targets in the semantically predictable conditions than for unrelated targets. Crucially, responses to the target word in the careful condition are delayed compared to casual speech for semantically unpredictable sentences. The implications for the apparent paradox in previous results will be discussed.

2pSCb18. Use of acoustic cues in the perception of complex syllable structure. Goun Lee (Linguist., Univ. of Kansas, 1541 Lilac Ln., Blake Hall, Rm. 427, Lawrence, KS 66045-3129, cconni@ku.edu)

This study examines second/foreign language learners' (L2ers') use of acoustic cues in the perception of syllable structure (onset clusters) that does not exist in the native language (L1, here, Korean). It investigates whether the target-like perception of syllable structure is facilitated by VOT ([stɪn] vs [sətɪn]) and partial devoicing of the liquid ([plim] vs [pəlim]) as compared to when either cue is absent ([blint] vs [bəlint]). This study also examined whether production errors can be predicted by perception errors. Twelve Korean-speaking English L2ers and 12 native English speakers completed AXB and production tasks in which they listened to and produced nonce words in the above three conditions. The AXB results showed significant effects of L1 and condition, but no interaction between the two. Participants were most accurate on the condition with VOT (English: 98%, Korean: 85%), followed by partial devoicing of the liquid (English: 96%, Korean: 79%) and without either cue (English: 94%, Korean: 74%). L2ers' perception and production patterned alike, but no correlation was found between the two. These findings suggest that acoustic cues like VOT and partial devoicing of the liquid facilitate L2ers' perception of new syllable structure, but do not play a direct role in its production.

2pSCb19. The perception of English [h] and [ɹ] by Brazilian Portuguese speakers. Denise M. Osborne (Univ. of Arizona, 814 E. 9th St. Apt 7, Tucson, AZ 85719, dmdcame@hotmail.com)

Brazilian Portuguese (BP) learners of English have difficulties in differentiating between initial English [h] and [ɹ] (Zimmer, Silveira, and Alvers, 2009). This study investigates how speakers who have BP as their L1 and English as their L2 perceive the phonetic distance of English [h] and [ɹ], and how they and monolingual BP speakers map these phonemes onto BP sound categories. 32 native BP learners of English participated in three consecutive experiments: AXB Discrimination, Identification, and Assimilation Tests. In addition, 18 monolingual BP speakers participated in the same Assimilation test. Significant effects in ANOVAs on the three experiments support the results. Beginners and intermediates were able to hear the distinction acoustically. Only intermediates, however, used the distinction to identify English words. Both monolingual BP speakers and intermediates assimilate English [h] primary to BP double <rr> (reflecting the dialectal and allophonic variation of the BP rhotic sounds). Beginners showed a failure to assimilate L2 sounds to their L1 BP categories. Comparison across these experiments shows that, at some stages of learning, BP speakers experience difficulty in evaluating these L2 sounds relative to their L1 inventory and in using the L2 distinction at the lexical level.

2pSCb20. Effects of hearing loss and linguistic context for phoneme and word recognition in noise. Adam Svec, Benjamin Munson, and Peggy B. Nelson (Dept. of Speech-Language-Hearing Sci., Univ. of Minnesota, 164 Pillsbury Dr. SE, Minneapolis, MN 55455, svecx002@umn.edu)

This project measured whole-word and phoneme recognition in normally hearing (NH) and hearing-impaired (HI) participants listening in steady-state noise. Stimuli were modeled after those used in Olsen *et al.* *[EarHear 1997]* including 10-word lists of consonant-vowel-consonant (CVC) monosyllables containing 10 vowels and 20 consonants. Words were presented in isolation and in contextually correct sentences. Speech-shaped noise was presented at a range of signal-to-noise ratios, from -9 to $+3$ dB SNR for NH listeners, and from -6 to $+6$ dB SNR for HI listeners. Results suggested that HI listeners needed an increased proportion of components (p_p) to recognize a whole word (p_w) relative to their NH counterparts (e.g., a higher j value) when stimuli were presented at unfavorable SNRs. For -3 and 0 dB SNR conditions, j factors for NH listeners (0 dB SNR: $j = 2.17$) were consistently lower than those for HI listeners (0 dB SNR: $j = 2.52$). However, for the 3 dB SNR conditions, j factors were higher for NH listeners ($j = 3.06$) than those for HI listeners ($j = 2.58$). This suggests an unexpected interaction between hearing acuity, background noise, and the recognition of phonemes versus whole words. [Research supported by DC008306 to Peggy Nelson.]

2pSCb21. Sociolinguistic perceptions of Californian /æ/ backing. Annette D'Onofrio (Linguist., Stanford Univ., 450 Serra Mall, Bldg. 460, Stanford, CA 94305, annetted@stanford.edu)

This paper investigates perceptions of a feature of the "California Vowel Shift" (Eckert, 2004): the backing of the TRAP vowel. The present study probes whether or not Californian and non-Californian listeners exhibit knowledge of TRAP backing's dialectal patterning in perception. An American English speaker's TRAP vowel was manipulated to create a 9-step continuum from /a/ to /æ/ in a sentence frame. 144 native U.S. listeners heard one of these steps, accompanied by orthographic information about the word spoken (e.g., the word was explicitly given as "blocked" or "blacked"). Listeners provided social categorizations of the speaker. Listeners were more likely to categorize the speaker as Californian when presented with TRAP orthography (e.g., "blacked") paired with a token on the continuum that was closer to /a/. 60 listeners participated in a phoneme categorization task using the same manipulated stimuli. When listeners were told that a speaker was from California, they were more likely to place the boundary between "blacked" and "blocked" closer to the /a/ end of the continuum than if told the speaker was from elsewhere. Together, these experiments indicate that listeners exhibit awareness of TRAP-backing's dialectal association with California, the top-down knowledge of which can affect lexical categorization.

2pSCb22. Detecting speaker change in background audio streams. Hilary Toh, Pei Xuan Lee (Victoria Junior College, 23 Li Hwan View, Singapore, Singapore 556913, Singapore, toh.si.yin.hilary.2011@vjc.sg), Boon Pang Lim, and Nancy F. Chen (Human Lang. Technol., Inst. for Infocomm Res., Singapore, Singapore)

The cocktail party effect refers to the ability of humans to selectively focus on one speech stream among multiple human-generated background conversations and noise sources. This ability is influenced by various factors including directionality, speaking rates, different speech accents and speaking styles. It is displayed most clearly as a binaural effect, requiring listeners to be able to identify and differentiate between multiple sound sources simultaneously. Past studies [Cherry, J. Acoust. Soc. Am. **25**(5), 975–979 (1953)], have shown that listeners can detect a change of speaker voice from male to female in unattended audio streams, while focusing their attention on repeating speech in an attended audio stream. Our work further investigates this by focusing on speaker change across the same gender: a more difficult task given a smaller variance in pitch across speakers. Experiments were conducted on 24 listeners and findings suggest that they are able to recognize speaker changes of the same gender. Two factor ANOVA shows an interaction between listener and speaker gender ($F(1, 22) = 7.688$, $p = 0.01$), suggesting listeners perform better when detecting speaker changes within their own gender.

2pSCb23. A word to the eyes: Visual cues benefit lexical segmentation in noise. Kate Helms Tillery, Sarah J. Cook, Rene L. Utianski, Julie M. Liss, and Michael F. Dorman (Dept. of Speech and Hearing Sci., Arizona State Univ., Coor Hall 2211, P.O. Box 870102, Tempe, AZ 85287-0102, kate.helms-tillery@asu.edu)

In English, most word-initial syllables are stressed. Listeners use these syllable strength cues to identify word boundaries in degraded acoustic conditions. This is evident in the types of lexical boundary errors they make when tasked with parsing a continuous stream of degraded speech: word boundaries are more often inserted before strong syllables than before weak syllables. Listeners also use visual cues from the talker's face to glean both phonemic and prosodic speech information in degraded listening conditions. While the benefits of lip-reading have received much attention, the role of visual cues in lexical segmentation remains largely unexplored. The present study examined the effect of auditory-visual cues on lexical boundary decisions. Normal-hearing listeners identified target phrases degraded by multi-talker babble. Responses in auditory-only and auditory-visual conditions were analyzed for percent words correct and lexical boundary error type. Results indicate large inter-individual variability, but overall an increase in word identification accuracy and a decrease in lexical boundary errors in the auditory-visual condition. Further, some listeners made a greater proportion of lexical boundary insertions before strong syllables, suggesting that the addition of visual cues increased their use of syllable strength to identify word boundaries. Implications for clinical populations will be discussed.

2pSCb24. Talker familiarity effects on speech-in-speech perception. Angela Cooper and Ann R. Bradlow (Linguist., Northwestern Univ., 2016 Sheridan Rd., Evanston, IL 60208, akcooper@u.northwestern.edu)

Talker familiarity can facilitate the extraction of linguistic content from speech signals embedded in broadband noise; however, relatively little research has investigated the impact of talker familiarity with competing speech in the background. This study explores the effects of familiarity with the target or competing talker in speech-in-speech perception. Listeners were first familiarized with and trained to identify three female voices. They then completed a sentence recognition task in the presence of 1-talker babble. Familiarity with either the target or background talker was manipulated in separate conditions. Results revealed significantly better sentence recognition for familiar relative to unfamiliar target talkers in the presence of an unfamiliar background talker; however, sentence recognition with an unfamiliar target talker did not differ depending on background talker familiarity. Thus, while listeners were able to capitalize on familiarity with a talker's voice to aid target speech recognition, familiarity with the competing talker was neither facilitative nor inhibitory. This suggests that the influence of talker familiarity is limited to the actively attended stream. This stands in contrast to other aspects of the unattended stream which have been shown to exert an influence on speech-in-speech recognition including language and semantic content of the background speech.

2pSCb25. Investigating the perception of relative speaker size using synthetic talkers. Santiago Barreda (Physiol., Univ. of Arizona, Dept. of Linguist., Edmonton, AB T6G 2E7, Canada, sbarreda@ualberta.ca) and Terrance M. Nearey (Linguist., Univ. of AB, Edmonton, AB, Canada)

Several previous experiments have investigated the perception of speaker size by presenting listeners with acoustic stimuli that differ in average f0 and/or apparent vocal tract length (i.e., higher formant frequencies overall), and asking listeners to make judgments of relative or absolute speaker size. Typically, these experiments use stimuli with a fixed phonetic content so that the acoustic characteristics of the stimuli may be compared directly. In this experiment, listeners were presented with pairs of vowels produced by synthetic speakers with different apparent vocal tract lengths and the same f0, and were asked to make judgments of relative size. However, listeners were presented with either the same, or different vowels produced by the two speakers. In some cases, differences associated with varying vocal tract lengths were in conflict with differences arising from the formant patterns associated with the differing vowel categories (e.g., lower F1 and F2 for /u/ vs /e/). Results suggest that judgments of relative size are affected by both the vowel categories presented and the apparent vocal tract lengths of the synthetic speakers. That is, more than a simple (category-corrected) vocal tract length estimate is involved in making size judgments for an unknown speaker.

2pSCb26. The effects of listener biases on speech intelligibility. Jamie Russell (Linguist., Univ. of Br. Columbia, 9320 Dixon Ave., Richmond, BC V6Y1E7, Canada, jamie.russell@alumni.ubc.ca) and Molly Babel (Linguist., Univ. of Br. Columbia, Vancouver, BC, Canada)

Race, gender, and age have traditionally been considered the three big social categories perceivers attend to. Recent research has highlighted how

perceivers, listeners in this case, use the acoustic-phonetic information from spoken language to evaluate and categorize individuals. For example, studies have shown that different dialects elicit varying judgments of intelligence, friendliness and attractiveness. Such biases have implications regarding our assessment of individuals, be it in the clinic or the courtroom. While we know that our perception of speech can trigger biases, it is important to explore the opposite—whether our biases affect our perception of speech. This study aims to contribute to the knowledge base of the latter by examining the effects of listener biases on speech intelligibility. Listeners will be presented with sentences, independently controlled for clarity, produced by model talkers of different dialects and backgrounds. This auditory stimuli will be accompanied by an image or description of the talker as a method of eliciting or constructing a preconceived linguistic bias. After each sentence, participants will be asked to type out what they heard. Analyses will examine the proportion of correct words, confidence ratings, response time, and other measures exploring participants' ratings of vocal qualities.

2pSCb27. Speaker normalization in noisy environments using subglottal resonances. Harish Arsikere and Abeer Alwan (Elec. Eng. Dept., Univ. of California, Los Angeles, 56-125B Eng. IV Bldg., 420 Westwood Plaza, Los Angeles, CA 90095-1594, hari.arsikere@gmail.com)

This work investigates the use of subglottal resonances (SGRs) for speaker normalization in noisy environments. Based on our previous work, a noise-robust algorithm is developed for estimating the first three SGRs from speech signals; it achieves robustness by factoring the short-term (or local) signal-to-noise ratio into the estimation process. The SGR estimates provided by this algorithm are refined by applying maximum-likelihood (ML) corrections, and are used in a non-linear frequency-warping technique that we recently developed. This SGR-based normalization (SN) scheme is evaluated on the AURORA-4 database in clean and noisy conditions. Using power-normalized cepstral coefficients (PNCCs) as front-end features, SN reduces the average word error rate by 8.7% relative to ML-based vocal-tract length normalization (VTLN). A fast version of SN (without ML corrections of SGR estimates) is also found to outperform VTLN (by 5.9% relative); it is computationally less complex than VTLN and hence a potential alternative for real-time applications.

Session 2pSP

Signal Processing in Acoustics, Underwater Acoustics, Structural Acoustics and Vibration, and Animal Bioacoustics: Defense Applications of Acoustical Signal Processing

R. Lee Culver, Cochair
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Brian G. Ferguson, Cochair
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Chair's Introduction—1:25

Invited Paper

1:30

2pSP1. Defense applications of acoustic signal time delay estimation. Brian G. Ferguson and Kam W. Lo (Maritime Div., DSTO, PO Box 44, Pyrmont, NSW 2009, Australia, Brian.Ferguson@dsto.defence.gov.au)

Measurements of the differences in the times of arrival (or time delays) of a signal at spatially-separated sensors can be used to provide localization information about a source. For continuous broadband signals, the time delay for a given pair of sensors is estimated using the generalized cross-correlation method. The time delay estimate corresponds to the time lag at which the cross-correlation function attains its maximum value. The variance of the time delay estimates depends on the signal-to-noise ratio, signal bandwidth, and integration time. Errors in the source localization parameters depend on the variance of the time delay estimates as well as on the source-sensor geometry, stationarity of the sound propagation medium, uncertainty in the actual sensor positions, and the presence of multipath arrivals. Numerous practical examples of time delay estimation and the instantaneous localization of sources of military interest are presented for acoustic sensors deployed on land and under water. Also, localization performance is observed to degrade when the sound propagation medium becomes nonstationary, the direct path and multipath arrivals are unresolvable, or the differential Doppler effect is significant. Finally, the results of source motion parameter estimation using sequences of time delay estimates from pairs of sensors are presented.

Contributed Papers

1:50

2pSP2. Passive acoustic localization of small aircraft. Alexander Sedunov, Alexander Sutin, Hady Salloum (Stevens Inst. of Technol., 711 Hudson St., Hoboken, NJ 07030, asedunov@stevens.edu), and Nikolay Sedunov (Stevens Inst. of Technol., Jersey City, NJ)

Stevens Institute of Technology has built the Acoustic Aircraft Detection (AAD) system for the detection, tracking and classification of Low Flying Aircraft (LFA). LFA may be of concern as they have been used for illicit operations. The AAD consists of several nodes deployed in a wide area, where each node acquires signals from a cluster of five microphones. The calculation of the cross-correlation function of acoustic signals received by various microphone pairs is applied for finding the direction of the signal arrival. Fusion of time difference of arrival estimates from several pairs of acoustic sensors has resulted in the improvement of angle estimation accuracy. Triangulation of the direction of arrival estimates from two or more nodes was applied for determining the target position and altitude. Kalman filtering was used for smoothing the target tracks and decreasing the localization uncertainties. Software for predicting the performance was developed and was used to inform sensor placement in the field tests. The field tests were conducted with various kinds of LFA—single-engine, helicopters, and ultralights. Comparison of the acoustic tracking results with the GPS ground truth showed that the errors are similar to the theoretical predictions. [This work was funded by DHS S&T.]

2:05

2pSP3. Automated acoustic detection and classification of small aircraft. Yegor Sinelnikov, Alexander Sutin, Alexander Sedunov, and Hady Salloum (Stevens Inst. of Technol., 711 Hudson St., Hoboken, NJ 07030, asedunov@stevens.edu)

Automated acoustic detection and classification of low flying aircraft can be used for the prevention of illicit operations. In this presentation, we review past techniques and a new acoustic classification and detection technique. The sound detection algorithm is based on locking in narrow frequency components in the spectra of the recorded aircraft sound. The classification algorithm uses tonal components in the spectral and cepstral domains using discrete time window peak picking. A set of chosen classifiers include: fundamental (pitch) frequency extracted by the cepstral analysis, frequency of tonal components with the maximal amplitude, power spectral estimate in various frequency bands, and number of peaks in predefined frequency and frequency windows. Various small single engine aircraft, ultralights, and helicopters are acoustically detected and classified as they approach the sensor. The algorithm enables simultaneous target detection, reduces noise sensitivity, and minimizes classifier feature space, while maintaining good classification separation. The acoustic classification algorithm was incorporated into the Acoustic Aircraft Detection system developed by Stevens. This combination allowed automated Doppler correction of the harmonic lines, the optimal part of the aircraft signal selection, and establishing classification merits based on aircraft track. [This work was funded by DHS S&T.]

2:20

2pSP4. Coordinated optimization of stationary and moving sensors with scenario-specific coverage and cost goals. Sergey Vecherin, D. Keith Wilson (Signature Phys. Branch, U. S. Army ERDC-CRREL, 72 Lyme Rd., Hanover, NH 03755, Sergey.N.Vecherin@usace.army.mil), and Chris Pettit (Aerosp. Eng. Dept., U. S. Naval Acad., Annapolis, MD)

The problem of simultaneously optimizing coverage among stationary sensors placed on the ground and moving sensors on aircraft is considered. The formulation extends a previously developed algorithm for optimizing placement of ground sensors subject to signal propagation effects and terrain-dependent coverage preferences. To this end, candidate aircraft routes should be characterized in terms of cost and coverage. Cost can reflect a variety of disincentives, for example, the probability that an aircraft can be heard on the ground. Coverage reflects the area of regions with probability of detection higher than a specified threshold. The extended algorithm can be applied to a variety of planning scenarios, such as determining an optimal combination of routes for multi-aircraft operations, optimization of aircraft routes to supplement ground-sensor coverage, optimization of ground sensors to cover blind spots of aircraft coverage, and simultaneous optimization of ground sensors and aircraft coverage. An illustrative problem of routing an unmanned aircraft system (UAS) to provide surveillance around a roadway while minimizing aircraft audibility at specified locations on the ground is considered in detail.

2:35

2pSP5. Performance comparison of distributed and centralized systems based on information. Tsih C. Yang (College of Marine Sci., Nat. Sun Yet-sen Univ., 70 Lien Hai Rd., Kaohsiung 804, Taiwan, tsihyang@gmail.com)

Distributed systems using many sensors widely distributed over a large area present an alternative way for target detection and environmental sensing. The diversity of opportunities for detection by widely distributed sensors seems attractive but the signals received on distributed sensors are usually not coherent due to the wide separations between the receivers. Hence, conventional approaches based on signal coherence/gain no longer apply. In this paper, information theory will be used to compare their performance. Treating the target radiated signal as a communication signal, transmitting continuous Gaussian-distributed alphabets, the Shannon channel capacity yields the maximum information that the receivers can learn about the (target) transmitted signal. The channel capacity can then be used as a metric to compare the performance of various sensor systems in the ideal case. Matched tracking processing is introduced to motivate a capacity-based detector and detection capacity. Based on that, it is found that the distributed systems can achieve in principle an area of coverage two to three times larger than that of a centralized system under the right conditions, and the area of coverage by the entire system can be significantly larger than the sum of detection areas of individual nodes.

2:50–3:05 Break

3:05

2pSP6. Acoustic signatures of small boats measured in controlled lake conditions. Alexander Sutin, Michael DeLorme, Hady Salloum, Alexander Sedunov, Nikolay Sedunov, Robert Weiss, Mikhail Tsionskiy, and Howard Goheen (Stevens Inst. of Technol., 711 Hudson St., Hoboken, NJ 07030, asutin@stevens.edu)

Measurements were conducted under controlled conditions at Lake Hopatcong, NJ, to compare acoustic signatures of various small boats. The tested boats included: an outboard-driven Panga, a multiple-outboard driven go-fast boat, an electric drive vessel, a personal watercraft (PWC), and two small outboard-driven rigid hull inflatable boats (RIBs). Stevens Passive Acoustic Detection System (SPADES) was used for acoustic measurements and the specially developed "Portable Vessel Data System" (PVDS) was used to record vessel position, speed, shaft RPM, and vessel orientation. Vessel acoustic Source Level measurements were conducted by comparing the recorded vessel acoustic signals with a signal generated by a calibrated emitter from the same point. Dependencies of acoustic signatures on boat speed and loading were also investigated. For several vessels, the Source Level decreased when its speed increased. Analysis of tonal components in the vessels acoustic signatures with Detection of Envelope Modulation on Noise (DEMON) allowed to determine the number of engines per boat and even the gear ratio for transformation of shaft rotation to the propeller rotation. [This work was supported by DHS S&T.]

3:20

2pSP7. Active–passive acoustic system for underwater port protection. Hady Salloum (Stevens Inst. of Technol., Hoboken, NJ), Andrew Meeecham (Sonardyne, Inc., Newport, RI), and Alexander Sutin (Stevens Inst. of Technol., 711 Hudson St., Hoboken, NJ 07030, asutin@stevens.edu)

Active and passive acoustic systems may be used for port protection against threats from divers, free swimmers, Unmanned Underwater Vehicles, etc. Currently, underwater port protection is provided by Diver Detection Sonars (DDS), and Sonardyne has developed the market-leading active system, Sentinel. Stevens' scientific research in passive acoustics led to the development of algorithms for threat detection that were realized in the Stevens Passive Acoustic Detection System (SPADES). Sonardyne and Stevens have formed a partnership to investigate a combined active/passive system that will leverage the functionality of existing sensors to design, develop, and produce a system with superior capabilities and performance as opposed to a single system. The combined system provides detection not only for underwater targets, but also for surface vessels. Possible configurations include: (1) a scenario where acoustic pulses emitted by the Sonardyne sonar and reflected by the target can be received and processed by SPADES; (2) an "acoustic fence," whereby the detection of targets occurs as they cross a line between the transmitter and receiver. This configuration takes advantage of the significant increase in target scattering strength in the forward direction versus that from the backscatter; and (3) a fusion between detections and/or tracks from the passive and active sensors to provide an increased probability of detection and reduced probability of false alarm.

3:35

2pSP8. Practical applications of track segment association algorithms to an active sonar network for underwater port surveillance. Patrick Edson and Peter J. Stein (Scientific Solutions, Inc., 99 Perimeter Rd., Nashua, NH 03063, pedson@scisol.com)

Acoustic detection and tracking of small (swimmer-size) targets while minimizing the incidents of false alerts can be challenging in a shallow, multipath, high-clutter harbor environment. One common problem involves track intermittency where periods or locations of high clutter or multipath interference inhibit detection and cause tracks to fail and restart. This paper describes the results of applying track segment association and related algorithms from airborne and ground radar applications [*e.g.,* Zhang and Bar-Shalom, IEEE Trans. Aerosp. Electron. Syst. **47**(3), 1899–1914 (2011)] to an active sonar network for underwater port surveillance [Edelson *et al*., J. Acoust. Soc. Am. **129**(4), 2598 (2011), Stein *et al*., J. Acoust. Soc. Am. **121**(5), 3084 (2007)] using real-world swimmer and synthetic target data.

3:50

2pSP9. Adaptive processing for mitigation of biologically induced active-sonar clutter and reverberation. Jason E. Summers (Appl. Res. in Acoust. LLC, 1222 4th St. SW, Washington, DC 20024-2302, jason.e.summers@ariacoustics.com)

Scattering from biological entities can be a significant source of reverberation and clutter for active sonar systems operating in deep or shallow water. Resonant scattering from the swim bladders of fish is a dominant source of this biologically induced reverberation and clutter for midfrequency active sonar (1–10 kHz), which can obscure targets, result in false targets, and otherwise overload operators of tactical systems. Though significant questions remain regarding the fisheries oceanography needed to predict clutter and reverberation, the basic physics of scattering from individual fish and groups of fish is well understood. This motivates a data-driven, clutter adaptive approach to mitigation of biologically induced reverberation and clutter that utilizes physical models without requiring unavailable databases of fish-species abundance and local spatiotemporal distributions. In this work, physics-based models relating physical parameters of fish and their spatiotemporal behaviors to spectral properties and spatiotemporal

distributions of scattering form the basis of *in situ* parameter estimation. In this presentation, adaptive processing developed for the early stages of the active-sonar processing chain using these *in situ* estimates is described, including waveform and matched-filter design and within and across-beam normalization. [Work supported by a NAVSEA Phase I SBIR award.]

4:05

2pSP10. Accuracy and resolution of bearing measurements for crossed-dipole receiver. Chunsheng Liu and Mark V. Trevorrow (Defence R&D Canada Atlantic, 9 Grove St., Dartmouth, NS B2Y 3Z7, Canada, chunsheng.liu@drdc-rddc.gc.ca)

A well-known system for estimating the bearing to an underwater acoustic source consists of two crossed dipole sensors and an omni-directional sensor. At DRDC Atlantic, several bearing estimation methods for this three-channel system in the presence of ocean noise have been examined with the goal of minimizing the bearing error. Example algorithms are arctangent and maximum likelihood beamformer. These algorithms have tradeoffs between estimation accuracy and computational efficiency, because any practical bearing-estimation algorithm must remain efficient enough for implementation in a real-time sonar processor. The accuracy of these algorithms can be demonstrated through comparison with the Cramér-Rao lower bound. Previously the calculation of bearing accuracy was based on the assumption of isotropic noise and a single signal source. However, the case of anisotropic noise and multiple-sources is more realistic, especially for bistatic or multistatic active sonar. Adaptive methods such as the Minimum Variance Distortionless Response (MVDR) beamformer will be explored to address this problem. The paper will discuss the potential improvement of bearing estimate accuracy and bearing resolution through numerical models.

4:20

2pSP11. Fast nearfield to farfield conversion for circular synthetic aperture sonar. Daniel Plotnick and Phillip L. Marston (Washington State Univ., 1510 NW Turner Dr., Apt. 4, Pullman, WA 99163, dsplotnick@gmail.com)

Monostatic circular synthetic aperture sonar (CSAS) images are formed by processing azimuthal angle dependent backscattering from a target at a fixed distance from a collocated source/receiver. In the laboratory data is taken by fixing the source location and spinning the target via a rotating mount. Typical CSAS imaging algorithms [Marston *et al*., Proc. IEEE Oceans (2011); Ferguson *et al*., J. Acoust. Soc. Am. **117**, 2915 (2005)] assume the scattering data are taken in the farfield. Experimental constraints may make farfield measurements impractical and thus require a target to be scanned in the nearfield. If left uncorrected, this results in distortions of the target image and possible distortions of the angular dependence of features. A fast approximate Hankel function based algorithm is presented that converts nearfield data to farfield data. Images and spectrograms of an extended target are compared for both cases. Spatial sampling requirements for data collection are also considered. [Work supported by ONR.]

TUESDAY AFTERNOON, 3 DECEMBER 2013 — CONTINENTAL 6, 2:00 P.M. TO 4:25 P.M.

Session 2pUW

Underwater Acoustics: Sound Propagation Through and Scattering by Internal Waves, Spice, and Finestructure in Shallow Water II: Past, Present, and Future

Steven I. Finette, Cochair
Acoust. Div., Naval Res. Lab., 4555 Overlook Ave. SW, Washington, DC 20375-5320

James Lynch, Cochair
Woods Hole Oceanogr., M.S. # 11, Bigelow 203, Woods Hole, MA 02543

Invited Papers

2:00

2pUW1. Predicting internal waves of acoustical significance. Timothy F. Duda (Appl. Ocean Phys. and Eng. Dept., Woods Hole Oceanographic Inst., Woods Hole, MA 02543, tduda@whoi.edu)

The broad dynamic range of internal gravity wave periods and scale lengths, as well as the variable group geometries of these waves, means that they are responsible for a variety of acoustic propagation effects. As a result, their degree of acoustical significance and the specific effects that they cause depend on time and place. A variety of internal-wave (IW) acoustic effects will be reviewed, with analysis of the regimes where each effect may have first-order propagation consequences. Many of the effects are specific to details such as the IW amplitudes, the angle between acoustic and IW propagation paths (and thus IW direction), IW proximity to sound source and/or receiver, and whether the IW field can be considered to be a random medium or, alternatively, a set of identifiable scattering features. The current state of our knowledge motivates a question: To what degree are specific acoustically relevant aspects of the IW field, and thus the acoustic effects, predictable? The answer to this question is being investigated through efforts to build ocean models bridging large-scale ocean dynamics, smaller-scale IW, and submesoscale dynamics, plus two-dimensional and three-dimensional acoustic propagation behavior.

2:20

2pUW2. Anisotropy in horizontal plane for the sound propagation in the ocean in the presence of internal waves. Boris Katsnelson (Marine GeoSci., Univ. of Haifa, 1, Universitetskaya sq, Voronezh 394006, Russian Federation, katz@phys.vsu.ru)

Review of researches dedicated to the sound propagation in the presence of internal waves is presented (starting with Evgeny Malyuzhinets, 1959 and Owen Lee, 1961). Main attention is paid on dependence of properties of the sound field (for example, sound amplitude fluctuations) on direction of propagation in horizontal plane. It was established—three main mechanisms: mode coupling, adiabatic variations, and horizontal refraction can be manifested in dependence on direction of sound propagation relative wave front of internal waves. Theoretical results as well as results of numerical modeling are presented illustrating different properties of acoustical signals for mentioned mechanisms including dependence on mode number and frequency. Experimental observations are also presented confirming real manifestations of mentioned three mechanisms.

Contributed Papers

2:40

2pUW3. Modeling the nonlinear internal wave field for coastal acoustics. Arthur E. Newhall, Ying-Tsong Lin, James F. Lynch, Timothy F. Duda (Appl. Ocean Phys. and Eng., Woods Hole Oceanogr. Inst., 98 Water St., 210 Bigelow Lab., M.S. #11, Woods Hole, MA 02543, anewhall@whoi.edu), and Karl R. Helfrich (Physical Oceanogr. Dept., Woods Hole Oceanographic Inst., Woods, MA)

It is now well known that coastal nonlinear internal waves, a common ocean feature, have large effects on the propagation of sound in the coastal ocean. Modeling such waves for input to acoustic models using regional ocean numerical models is difficult due to the fine spatial and temporal resolution needed. A multi-university program called Integrated Ocean Dynamics and Acoustics (IODA) is addressing this problem using two approaches—one a two-way, direct nested ocean model approach, and the other a blending of mesoscale ocean models and fine scale internal wave models. In this talk, we will discuss the latter approach. Issues of the internal wave source positions, their nonlinear propagation, earth rotation effects, field interpolation, output verification and integration of the ocean model with acoustics codes will be discussed. The results going from an ocean model to fully 3-D acoustic fields will be presented. [Work sponsored by ONR MURI program.]

2:55

2pUW4. Effect of internal waves on the waveguide invariant distribution. Daniel Rouseff (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St,, Seattle, WA 98105, rouseff@apl.washington.edu)

Multipath propagation in the ocean waveguide results in constructive and destructive interference between the paths. The resulting interference pattern, mapped versus range and frequency, often exhibits striations, alternating bands of high and low intensity. In simple waveguides, the trajectory of these striations can often be described by a single scalar parameter, the so-called waveguide invariant. In more complicated waveguides, the concept can be generalized to create the waveguide invariant distribution. In the present work, the effects of shallow water internal waves on the waveguide invariant distribution are examined. It is shown how internal waves can blur otherwise sharp striations and flatten the waveguide invariant distribution. [Work supported by ONR.]

3:10

2pUW5. Modal interference and frequency striations induced by moving nonlinear internal waves. Tsih C. Yang (College of Marine Sci., Nat. Sun Yet-sen Univ., 70 Lien Hai Rd., Kaohsiung 804, Taiwan, tsihyang@gmail.com) and Jin-Yuan Liu (Nat. Taitung Univ., Taitung, Taiwan)

Spectrogram of a broadband signal propagated over distances often displays striated bands of constant acoustic intensity levels plotted against the range and frequency. This phenomenon is due to modal interference and the slope of the striation is related to the source range and frequency by the waveguide invariant parameter "beta." As an application, one can estimate the range to a moving source based on the striation of the spectrogram (as a function of frequency and time), knowing the source velocity and the waveguide invariant parameter. Conversely, no frequency striation is expected in the spectrogram of the received signal when both the source and receiver are fixed (the frequency content remains unchanged with respect to time). However, when nonlinear internal waves, appearing as a train of moving solitons, are present in the propagation path, the spectrogram of the received signal will display frequency striations (with respect to time) even for a fixed source and receiver. This paper applies the mode coupling equation to calculate the modal interference, intensity fluctuation and the striation slope. Broadband spectrograms are simulated to show the frequency striation as the solitons move from the receiver to the source. Results from at-sea data will be presented for comparison.

3:25

2pUW6. Synthetic aperture sonar imaging of breaking internal wave structures. Anthony P. Lyons (Appl. Res. Lab, Penn State Univ., University Park, State College, PA 16803, apl2@psu.edu), Roy E. Hansen, Torstein O. Sæbø (Norwegian Defence Res. Establishment (FFI), Kjeller, Norway), and Hayden J. Callow (Kongsberg Maritime, Horten, Norway)

In October 2012, the Centre for Maritime Research and Experimentation (CMRE) conducted the ARISE'12 trials from the NATO research vessel Alliance, off of Elba island, Italy. During this trial, data were collected by the Norwegian Research Defence Establishment (FFI) using a HUGIN AUV with interferometric synthetic aperture sonar (SAS). Large visible structures in the SAS images and in the SAS bathymetries were caused by features in the water column, called boluses, which formed after breaking internal wave events. Variation in backscattered intensity from the seabed was caused by focusing of sound by the bolus and errors in bathymetry by the changes in phase as the acoustic energy moved through the lower sound speed bolus. A ray model and a 3D parabolic equation (PE) model were used to simulate the effects of the bolus on the acoustic field incident on the bottom and results compared favorably with SAS data. Models for the change in bolus size and speed as it propagated upslope also gave reasonable comparison to estimates obtained from SAS intensity images. [The work performed by APL was supported by the US Office of Naval Research.]

3:40

2pUW7. Modeling sound propagation through internal waves using a spectral element method. Sumedh M. Joshi (Ctr. for Appl. Mathematics, Cornell Univ., 657 Rhodes Hall, Ithaca, NY 14850, sumedh.m.joshi@gmail.com), Megan S. Ballard (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), and Peter J. Diamessis (Dept. of Civil and Environ. Eng., Cornell Univ., Ithaca, NY)

Considered here is the problem of low-frequency sound propagation over shallow, shoaling bathymetry in the presence of perturbations to the background sound velocity profile due to internal waves (IW). The question we attempt to answer is: to what degree can heuristic models of IWs coupled to numerical sound propagation models capture the variability in sound propagation observed in the environment? A high-order finite element model is employed to compute the acoustic field as it propagates through these IWs. The generality of the finite element method allows for spatial and temporal sound speed variations, and its convergence properties yield arbitrarily small error as the grid is rened. Simulations in the presence

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and absence of IWs will demonstrate the degree to which IWs influence sound propagation. Different models of IWs will demonstrate the sensitivity of the sound propagation to the choice of heuristic used for the IWs. Results will be shown for shoaling waveguides of O(100 m) depth and O(10 km) range, and frequencies of O(50 Hz). [Work supported by the NDSEG Fellowship.]

3:55

2pUW8. From Swarm 1995 to Shallow Water Experiment 2006: What have we learned in shallow waveguide acoustics? Mohsen Badiey (College of Earth, Ocean, and Environment, Univ. of Delaware, 261 S. College Ave., Robinson Hall, Newark, DE 19716, badiey@udel.edu)

Last 20 years have witnessed some very interesting scientific discoveries in shallow water waveguide physics. While before1995 SWARM experiment most studies in shallow water waveguides considered two-dimensional (2D) slices of the ocean waveguide to describe the behavior of sound wave propagation, many studies in the recent years have focused on the three dimensionality of the acoustic wave propagation in the presence of water column variability. This paper summarizes the highlights of what has been done with reference to the published literature and some ongoing work. [Work supported by ONR322OA.]

4:10

2pUW9. Estimates of source range using horizontal multipath in continental shelf environments. Megan S. Ballard (Appl. Res. Labs. at the Univ. of Texas at Austin, P.O. Box 8029, Austin, TX 78758, meganb@arlut.utexas.edu)

In a continental shelf environment, energy from an acoustic source propagating obliquely upslope repeatedly reflects from the sea surface and sloping seafloor with a consequent change in direction with each bottom reflection. Measurements of this type of effect can be observed in beamformed data from a horizontal line array. The arrivals from a single source are seen on two beams: a direct path with bearing angle corresponding to the source location and a refracted path with a bearing angle inshore of the source. In this work, the horizontal multipath effects are exploited to estimate the location of an acoustic source. Using the hybrid modeling approach of vertical modes and horizontal rays, rays are traced in the horizontal plane with refraction determined by the modal phase speed. Invoking reciprocity, the rays orientate from the center of the array and have launch angles equal to the estimated bearing angles of the direct and refracted paths. The location of the source in the horizontal plane is estimated from the point where the rays intersect. The technique is applied to data recorded on a horizontal line array located about 12 km east of the southern coast of Florida. [Work supported by ONR.]

TUESDAY AFTERNOON, 3 DECEMBER 2013 UNION SQUARE 14, 1:45 P.M. TO 3:15 P.M.

Meeting of Accredited Standards Committee (ASC) S1 Acoustics

R.J. Peppin, Chair ASC S1
5012 Macon Road, Rockville MD 20852

A. Scharine, Vice Chair ASC S1
U.S. Army Research Laboratory, Human Research & Engineering Directorate
ATTN: RDRL-HRG, Building 459 Mulberry Point Road
Aberdeen Proving Ground MD 21005 5425

Accredited Standards Committee S1 on Acoustics. Working group chairs will report on the status of standards currently under development in the areas of physical acoustics, electroacoustics, sonics, ultrasonics, and underwater sound, etc. Consideration will be given to new standards that might be needed over the next few years. Open discussion of committee reports is encouraged.

People interested in attending the meeting of the TAGs for ISO/TC 43 Acoustics, ISO/TC 43/SC 3, Underwater acoustics, and IEC/TC 29 Electroacoustics, take note - those meetings will be held in conjunction with the Standards Plenary meeting at 9:00 a.m. on Tuesday, 3 December 2013.

Scope of S1: Standards, specifications, methods of measurement and test, and terminology in the field of physical acoustics, including architectural acoustics, electroacoustics, sonics and ultrasonics, and underwater sound, but excluding those aspects which pertain to biological safety, tolerance and comfort.

Meeting of Accredited Standards Committee (ASC) S12 Noise

W.J. Murphy, Chair, ASC S12
NIOSH, 4676 Columbia Parkway, Mail Stop C27, Cincinnati, OH 45226

S.J. Lind, Vice Chair, ASC S12
The Trane Co., 3600 Pammel Creek Road, Bldg. 12-1, La Crosse WI 54601-7599

Accredited Standards Committee S12 on Noise. Working group chairs will report on the status of noise standards currently under development. Consideration will be given to new standards that might be needed over the next few years. Open discussion of committee reports is encouraged.

People interested in attending the meeting of the TAG for ISO/TC 43/SC 1 Noise, take note - that meeting will be held in conjunction with the Standards Plenary meeting at 9:00 a.m. on Tuesday, 3 December 2013.

Scope of S12: Standards, specifications and terminology in the field of acoustical noise pertaining to methods of measurement, evaluation and control, including biological safety, tolerance and comfort, and physical acoustics as related to environmental and occupational noise.

TUESDAY EVENING, 3 DECEMBER 2013 8:00 P.M. TO 10:00 P.M.

OPEN MEETINGS OF TECHNICAL COMMITTEES

The Technical Committees of the Acoustical Society of America will hold open meetings on Tuesday and Thursday evenings beginning at 8:00 p.m. and on Wednesday evening beginning at 7:30 p.m.

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

Committees meeting on Tuesday are as follows:

Acoustical Oceanography	Plaza B
Animal Bioacoustics	Union Square 23/24
Architectural Acoustics	Golden Gate 4/5
Engineering Acoustics	Mason
Noise	Continental 9
Physical Acoustics	Continental 7/8
Psychological and Physiological Acoustics	Continental 2/3