Session 1pAA

Architectural Acoustics and Signal Processing in Acoustics: Acoustics in Coupled Volume Systems

U. Peter Svensson, Cochair
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Invited Papers

1:00

1pAA1. Free path statistics in coupled rooms. Michael Vorlaender (Inst. of Tech. Acoust., RWTH Aachen Univ., Neustr. 50, Aachen 52066, Germany, mvo@akustik.rwth-aachen.de)

Statistics of free paths in room acoustics led to the statistical reverberation theory of diffuse fields in spaces and thus to the well-known reverberation formulas. The specific distribution of free paths, however, can also give a more specific insight into the sound field and its deviations from the expected result using the mean free paths. Recently, Hanyu (JASA **128**, 1140, 2010) extended the statistical reverberation theory toward a separation of absorption effects and mixing effects by scattering. This approach can predict nonlinear decay curves and the transition of energy between specular and scattered processes. In the work presented here, a concept of Vorländer (J. Sound Vib. **232**, 129, 2000) is revisited. Free path distributions are logged in the room simulation program RAVEN and displayed in real time with regard to the geometric configuration of coupled spaces, and to the influence of absorption and scattering at the boundaries. This allows the observation of the decay curves in dependence on the actual free path distribution. From this data, it is tried to further develop simple prediction scheme of decay curves in non-diffuse and coupled spaces.

1:20

1pAA2. Inferring delays between subrooms in systems of coupled rooms. Jason E. Summers (Appl. Res. in Acoust. LLC, 1222 4th St. SW, Washington, DC 20024-2302, jason.e.summers@ariacoustics.com) and Jonathan Botts (Dept. of Media Technol., Aalto Univ., Espoo, Finland)

Delay-differential-equation models of reverberant-energy decay in systems of coupled rooms were recently introduced as a semi-empirical correction to standard statistical-acoustics models in order to better account for the delay of energy transfer between subrooms of a coupled system [Summers, J. Acoust. Soc. Am. **132**, EL129-EL134, 2012]. Here, a Bayesian approach to parameter estimation for these models is developed and evaluated on simulated and measured data. Using these data, the ability to invert for delays from measured decay curves is assessed and an interpretation of the physical meaning of those inferred delays is presented. Finally, the theoretical and mathematical relationships between higher-degree-of-freedom models resulting from the introduction of delay and those resulting from the introduction of additional decay rates to the standard sum-of-decaying-exponential model are considered, and the resulting implications for parameter estimation and model selection are described.

1:40

1pAA3. Room acoustical conditions in coupled video conference rooms. Erlend I. Gundersen (Aker Solutions, Lysaker, Norway) and U. Peter Svensson (Electron. and Telecommunications, Norwegian Univ. of Sci. and Technol., O.S. Bragstads pl. 2B, Trondheim NO-7491, Norway, svensson@iet.ntnu.no)

A video conferencing situation combines the acoustical properties of two rooms. The resulting convolution of the two room impulse responses leads to a total impulse response with a reverberation, which is not a classical exponential decay. As a consequence, relationships between parameters such as clarity and reverberation time will be significantly different from those in single rooms, and this will furthermore affect the combination's suitability for speech communication. In this study, a measurement survey is presented from 11 rooms with video conferencing equipment. Their volumes ranged from 24 to 117 m^3, and their mid-range reverberation times were between 0.29 s and 0.70 s. Median values were 82 m^3 and 0.41 s, respectively. Impulse responses were measured in all rooms and in a subsequent analysis stage, impulse response pairs were convolved, simulating a connection between the corresponding rooms. Those convolved IRs were analyzed in terms of clarity and reverberation time. Recommended parameter values for single rooms were used as guidelines to understand the qualities of convolved rooms.

2:00

1pAA4. Auralization of virtual rooms in real rooms using multichannel loudspeaker reproduction. Soenke Pelzer and Michael Vorlaender (Inst. of Tech. Acoust., RWTH Aachen Univ., Neustr. 50, Aachen 52066, Germany, spe@akustik.rwth-aachen.de)

When playing auralizations including virtual room reverberation through loudspeaker-based reproduction systems, there is usually an interaction between the auralized virtual rooms with the real room acoustics of the listening environment. In case of a listening room which is not perfectly dry, it is investigated which criteria the listening room should fulfill to avoid considerable interference with the auralizations. In a further step, a computer room acoustics simulation is extended to account for the listening space by modifying the resulting room impulse responses, so that the final room-in-room situation matches best to the targeted virtual room acoustics. The presented technique is then applied in a multimodal immersive virtual display (CAVE-like environment) where room acoustics are not matter of choice due to restrictions of projection screen materials and placement.

2:20–2:35 Break

2:35

1pAA5. Energy decay analysis in coupled volumes using an acoustic wave simulator. Anish Chandak, Lakulish Antani (Impulsonic, Inc., 222 Old Fayetteville Rd., C101, Carrboro, NC 27510, achandak@impulsonic.com), and Dinesh Manocha (Dept. of Comput. Sci., Univ. of North Carolina at Chapel Hill, Chapel Hill, NC)

The acoustics in coupled volumes present various challenges. For example, the sound energy decay curve is double-sloped indicating that a single reverberation time cannot be assigned to coupled volumes. Such a behavior depends on various factors like the aperture sizes between the coupled volumes, absorption coefficients of each volume, etc. Recently, many studies using computer simulation have been performed to better understand the acoustic behavior of coupled volumes. None of these studies perform full 3D acoustic wave simulation on coupled volumes as they are regarded as computationally expensive and limited to very small spaces. We perform acoustic wave simulation using a relatively new wave solver called adaptive rectangular decomposition (ARD). ARD is more accurate than geometric acoustics techniques. It has been demonstrated to provide reliable simulation results through comparison with measurement data in indoor and outdoor scenes. Furthermore, it is practical and can handle large acoustic spaces on a single desktop. In this study, we analyze energy decay curves in coupled volumes using ARD. We perform numerical simulation for various coupled volume configurations such as varying aperture sizes, absorption coefficients, etc., and present their decay curves for analysis.

2:55

1pAA6. Multiple-slope sound energy decay investigations in single space enclosures with specific geometrical and material attributes. Zühre Sü Gül (Architecture; R&D, Middle East Tech. Univ.; MEZZO Studyo LTD, METU Technopolis KOSGEB-TEKMER No112, ODTU Cankaya, Ankara 06800, Turkey, zuhre@mezzostudyo.com), Ning Xiang (Graduate Program in Architectural Acoust., School of Architecture, Rensselaer Polytechnic Inst., Troy, NY), and Mehmet Caliskan (Mech. Eng., Middle East Tech. Univ., Ankara, Turkey)

Acoustical coupling until now has basically been studied to define the peculiar sound field within acoustically coupled enclosures in which multiple-slope energy decays can often be observed. The key concern of this study is to reveal the potential of multiple-slope energy decay formation in over-size single space structures with particular geometry and distribution of materials in different acoustical performance characteristics. Specifically, multiple dome superstructures, composed of one central dome supported by semi-domes and transitional elements, are selected for the case studies. The interpretation of the acquired data is carried in order to broaden the definition of "the coupled volume system" with an emphasis on invisible sources and apertures of acoustical coupling in a "single volume system." The methodology of the research involves joint use of *in-situ* acoustical measurements, acoustical modeling/simulation methods, and computational analyses. Bayesian analysis approach is applied in quantifying multiple-slope decay parameters. Initial results for selected cases indicate double and triple slopes for field tests and even more slope natures for simulations at various frequency bands. Future work aims to elaborate the mechanism of multiple-slope decay occurrence by energy feedback and distribution analysis.

Contributed Papers

3:15

1pAA7. Binaural effects in convolved room impulse responses. Ulrich Reiter and U. Peter Svensson (Electron. and Telecommunications, Norwegian Univ. of Sci. and Technol., O.S. Bragstads pl. 2B, Trondheim NO-7491, Norway, svensson@iet.ntnu.no)

In situations like video and teleconferencing, the acoustical properties of two rooms are involved via a convolution of the two room impulse responses. Reverberation in the two rooms will play quite different roles since the sender room reverberation will partially reach the listener, in the receiver room, from the source direction. Therefore, binaural suppression of reverberation will be less efficient for parts of the total impulse response. Here, the situation is analyzed in terms of a simple room impulse response model with a direct sound followed by an ideal exponential decay. Such a model permits parametric studies, even of convolved impulse responses. Listening test results will quantify the importance of these binaural effects for the perceived quality of speech signals.

3:30

1pAA8. Stages with high ceilings, pipe organs, and active acoustics. Roger W. Schwenke and Steve Ellison (Res. & Development, Meyer Sound Labs., 2832 San Pablo Ave., Berkeley, CA 94702, rogers@meyersound.com)

Two case studies are presented of rooms that have a high ceiling over the stage to accommodate a pipe organ. Svetlanov hall is the principal venue of Moscow's International Performing Arts Center. It has two to four rows of chorus seating on two levels at stage left and right. The upstage wall is occupied by the largest pipe organ in Russia. It has a physical reverberation time of 1.7 s, which is within the accepted range for symphonic music, but longer reverberation times would be preferred for some pipe organ repertoire. Christopher Cohan Performing Arts Center at California Polytechnic State University has a proscenium and thrust stage. The Forbes Pipe Organ is housed on the stage right wall of the thrust stage. The symphony usually performs on the thrust stage entirely in front of the proscenium with the solid decorative fire curtain down. Both rooms had poor communication on stage between performers, which led them to implement a solution using active acoustics. In both rooms all of the active acoustic elements overhead of the stage are on motors and can be retracted when not in use.

3:45

1pAA9. Sound propagation to and around the balcony edge in a performance hall. Liu Yee Cheung and Shiu Keung Tang (Dept. of Bldg. Services Eng., The Hong Kong Polytechnic Univ., Hung Hom, Kowloon, Hong Kong, louisa.cheung@connect.polyu.hk)

To investigate the propagation of sound from the source to and around the balcony, over 150 points were measured in a 1420-seat auditorium. Radial points around the balcony edge with distances of 1.5 and 2 m, as well as intermediate points on the direct paths between the omni-directional sound source on the stage and the centers of the radial positions around the balcony edge, were also measured. A 1:20 scaled-model of this hall was done as a complement to the full-scaled measurement done to study the propagation in finer details. Various acoustic parameters were obtained. With the wave file of the impulse response recorded, the propagation pattern could be traced.

MONDAY AFTERNOON, 2 DECEMBER 2013 UNION SQUARE 23/24, 1:00 P.M. TO 5:15 P.M.

Session 1pAB

Animal Bioacoustics: Signal Identification, Processing, and Analysis

Brian K. Branstetter, Chair
National Marine Mammal Foundation, 2240 Shelter Island Dr. #200, San Diego, CA 92106

Contributed Papers

1:00

1pAB1. Acoustic feature extraction and classification in Ishmael. David K. Mellinger (Coop. Inst.for Marine Resources Studies, Oregon State Univ., 2030 SE Marine Sci. Dr., Newport, OR 97365, David.Mellinger@oregonstate.edu)

Ishmael is a user-friendly bioacoustic analysis tool for Windows. It includes displays of sound waveforms and spectrograms, recording capability for real-time input, several methods for acoustic localization, beamforming, several methods for automatic call detection, and a sound annotation facility. Ishmael is intended for users wishing to analyze large volumes of data quickly and easily. Ishmael's capabilities for classification now include a feature-extraction module that implements noise-resistant acoustic characterizations based on Fristrup's AcouStat system (Fristrup, Woods Hole Tech. Rept. WHOI-92-04, 1992). These features can be used as input to classifier(s) connected to Ishmael and implemented in MATLAB. Examples of cetacean (odontocete and mysticete) sound feature extraction and classification will be shown.

1:15

1pAB2. Can wavelets solve the cocktail party? Mark Fischer (Aguasonic Acoust., P. O. Box 308, Rio Vista, CA 94571-0308, info@aguasonic.com)

There are many eco-systems that exist where avian populations are dense. In such environments, it is often the case that many different species may sing contemporaneously with significant overlap in the frequencies used by each. Work will be presented to support the hypothesis that wavelet processing allows an analog solution to this "cocktail party" problem.

1:30

1pAB3. Identification of individual beaked whales in the northern Gulf of Mexico. Juliette W. Ioup, George E. Ioup (Dept. of Phys., Univ. of New Orleans, New Orleans, LA 70148, jioup@uno.edu), Christopher O. Tiemann (Appl. Res. Labs., Univ. of Texas, Austin, TX), Natalia A. Sidorovskaia (Dept. of Phys., Univ. of Louisiana at Lafayette, Lafayette, LA), and Arslan M. Tashmukhambetov (Dept. of Physics, Univ. of New Orleans, New Orleans, LA)

Recent Littoral Acoustic Demonstration Center (LADC) multi-mooring Environmental Acoustic Recording System (EARS) data from the northern Gulf of Mexico are analyzed to deduce identifications of individual beaked whales. Procedures are built on previously applied self-organizing map techniques for clustering sperm whale clicks and beaked whale clicks from workshop data. Associating the clicks from beaked whales is difficult because, compared to sperm whales, beaked whales have lower source level, a narrower beam, and a faster rate of turning. Recordings of individual clicks can be clustered according to their time domain signal, frequency spectrum, or wavelet spectrum. For example, clicks clustered according to the magnitude of their frequency components show similarities for all the clicks in a class (representing an individual), but significant differences from class to class, suggesting that this approach has promise for identifying individuals. Recent work by Baggenstoss (J. Acoust. Soc. Am. **130**, 102–112 (2011); J. Acoust. Soc. Am. **133**, 4065–4076 (2013)), who has used cross correlations of clicks to assist in associating beaked whale clicks into trains, reinforces the idea that single click properties can be associated with individuals. [Research supported by SPAWAR and ONR.]

1:45

1pAB4. Dolphin biosonar target detection in noise, auditory filter shape, and temporal integration. Whitlow W. Au (Hawaii Inst. of Marine Biol., Univ. of Hawaii, 46-007 Lilipuna Rd., Kaneohe, HI 96744, wau@hawaii.edu)

The biosonar target detection in noise capability of the Atlantic bottlenose dolphin (*Tursiops truncatus*) has been studied as early as 1981 by Au and Penner. At that time, they presented results of the performance of two dolphins as a function of the signal energy to the density of the noise intensity (as in human psycho-acoustics) and not the signal energy to the noise energy ratio. Subsequently, two important pieces of information were determined, that is the auditory filter shape of the bottlenose dolphin in 2012 along with the temporal integration time for the reception of broadband biosonar echoes in 1988. In all but one experiment, it has been shown that the auditory system of odontocetes has a constant-Q characteristic. The biosonar detection capability of the bottlenose dolphin is revisited, and the target detection performance is now determined as a function of the received energy in the echo to the received noise energy. This presentation is an example of how some biosonar questions can only be answered after many years of related research and how sequences of research projects should be established so that current results can be applied to past results to gain a deeper understanding of the biosonar process.

2:00

1pAB5. Sperm whale coda repertoires in the western Pacific Ocean. Elizabeth L. Ferguson (Bio-Waves, Inc., 12544 Caminito Mira Del Mar, San Diego, CA 92130, eferguson@bio-waves.net), Thomas F. Norris, Cory A. Hom-Weaver, and Kerry J. Dunleavy (Bio-Waves, Inc., Encinitas, CA)

Sperm whales are social cetaceans that live in matrilineal family units, and inhabit all major ocean basins from the tropics to polar regions. They produce stereotyped patterns of 3 to 40 broadband clicks, termed "codas," that typically occur within a period of less than 3 s. Coda repertoires can be assigned to a "vocal clan," a type of social group used to define sperm whale population structure. Extensive studies of vocal clans have been conducted in the eastern tropical Pacific (ETP); however, little is known about sperm whale coda repertoires in the western Pacific. We reviewed codas recorded from independent sperm whale groups that were acoustically and visually encountered during two marine mammal surveys conducted in the Northern Mariana Islands (10 groups) and Palau region (3 groups), in 2007 and 2012, respectively. Three bioacousticians qualitatively classified codas to type, which indicated the presence of the "+1" and "regular" vocal clan. These data are now being analyzed using multivariate methods described by Rendell and Whitehead (2003) to quantitatively classify codas from each group. The identification of vocal clans within this region has implications for understanding the culturally linked stock distribution of sperm whales across the Pacific Ocean.

2:15

1pAB6. The effects of site and instrument variability on recognizing odontocete species by their echolocation clicks. Johanna Stinner-Sloan, Marie A. Roch (Dept. of Computer Science, San Diego State Univ., 5500 Campanile Dr, San Diego, CA 92182-7720, Johanna.RosalieChristina@gmail.com), and Simone Baumann-Pickering (Scripps Inst. of Oceanogr., Univ. of California, San Diego, San Diego, CA)

This work conducts a systematic study of the impact of variable site and instrument conditions on the performance of a species recognition task. Echolocation clicks were collected from six different sites in the Southern California Bight from multiple deployments of instruments with nine different preamplifiers. The classification performance of a Gaussian mixture model using cepstral features is examined on Risso's and Pacific White-Sided dolphins. One hundred three-fold Monte Carlo experiments are conducted. When grouped so that each acoustic encounter is either in the training or test set, a mean error rate of $1.9\%\pm4.4\sigma$ is obtained. In spite of correction for preamplifier response curves, grouping by preamplifier and site increases error dramatically to $20.9\%\pm18.1\sigma$ and $25.9\%\pm28.1\sigma$, respectively. We introduce noise compensation techniques that reduce error rates as follows: by encounter, $0.5\%\pm0.3\sigma$, by preamplifier, $1.7\%\pm2.3\sigma$, and by site, $9.4\%\pm16.7\sigma$.

2:30

1pAB7. Using inter-click intervals to separate multiple odontocete click trains. Eva-Marie Nosal (Ocean and Resources Eng., Univ. of Hawaii at Manoa, 2540 Dole St., Holmes Hall 405, Honolulu, HI 96822, nosal@hawaii.edu), Anders Høst-Madsen, and Jeremy Young (Elec. Eng., Univ. of Hawaii at Manoa, Honolulu, HI)

We developed a method to separate click trains from multiple odontocetes that relies on click timing only. The method assumes that inter-click intervals (ICIs) are slowly varying and that their distribution is known, though possibly with unknown parameters. ICIs are modeled as a renewal process and click trains are separated by maximizing the likelihood of click assignments based on ICI distributions. We present results from application of this timing-based separation method to simulated and real datasets. [Work supported by the National Science Foundation and the Office of Naval Research.]

2:45

1pAB8. Comparisons of beaked whale signals recorded in eastern Atlantic to known beaked whales signals. Odile Gerard (DGA, Avenue de la Tour Royale, Toulon 83000, France, odigea@gmail.com)

Beaked whales are a group of more than 20 genetically confirmed species; they are very elusive and were among the least known species until a few years ago. Because of their sensitivity to sonar, an increased research effort dedicated to these species and in particular to their signals started 10 years ago. Despite this effort, the signals of many beaked whale species remain unknown. Signals of Blainville's (*Mesoplodon densirostris*), Cuvier's (*Ziphius cavirostris*), Gervais' (*Mesoplodon europaeus*), and Northern Bottlenose Whales (*Hyperoodon ampullatus*) have been studied and described in different articles. The signals of three unknown species (different from the above mentioned species signals) have been reported. All seven species produce upsweep frequency modulated signals which seem species specific. In 2010, NATO Undersea Research Centre (NURC) conducted a sea-trial in Eastern Atlantic Ocean, Southwest of Portugal. Three different types of beaked whale signals were recorded. The characteristics of these signals will be presented and compared to the known signals of beaked whales. The conclusions of this analysis will be presented.

3:00–3:15 Break

3:15

1pAB9. Whistle classification by spectrogram correlation. Yang Lu and David Mellinger (Oregon State Univ., 2030 SE Marine Sci. Dr., Newport, OR 97365, lu.yang@noaa.gov)

The spectral properties of whistles are investigated for sounds collected from six species of odontocetes in the North Pacific: bottlenose dolphins (*Tursiops truncatus*), melon-headed whales (*Peponocephala electra*), short- and long-beaked common dolphins (*Delphinus delphis* and *D. capensis*), and Hawaiian spinner dolphins (*Stenella longirostris longirostris*). The proposed method first uses the k-means algorithm to cluster whistles of each species into groups which represent their spectral contours. The features used for clustering are based on a method described by Mellinger *et al.* (J. Acoust. Soc. Am. **107**, 3518-3529). Short spectrogram kernels of different slopes are cross-correlated with training data to obtain histograms of slopes and other acoustic properties. The features are used in a random forest classification algorithm. The classified whistles which belong to different clusters but the same species are merged together, resulting in classification to species. The performance of these methods, as evaluated via confusion matrices, is compared in terms of accuracy and complexity with other extant algorithms.

3:30

1pAB10. Robust recognition of whistle-like frequency contours by a bottlenose dolphin (*Tursiops truncatus*). Brian K. Branstetter (National Marine Mammal Foundation, 2240 Shelter Island Dr. #200, San Diego, CA 92106, brian.branstetter@nmmpfoundation.org), Amy Black, Kimberly Bakhtiari, and Jennifer Trickey (G2 Software Systems Inc., San Diego, CA)

Whistle use by bottlenose dolphins serves several functions including individual identification, maintaining group cohesion, long-range communication, recruitment during feeding, and advertising emotional state. Whistles can vary considerably in amplitude, duration, and frequency. Despite this variability, dolphins must learn to associate specific whistle features (e.g., frequency contour) with meaningful events or objects, requiring plasticity in the dolphin's recognition abilities. To test robust recognition, a bottlenose dolphin learned to associate whistle-like frequency contours with arbitrary objects. For example, whistle-A would cue the dolphin to touch object-A, and whistle-B would cue her to touch object-B. The whistles were then altered by varying amplitude, duration, and transposing frequency. Changes in amplitude and duration had little effect on recognition, however frequency transposition as small as 1/3 octave resulted in poor performance. The results are discussed from a comparative cognition perspective.

3:45

1pAB11. Sparse representation classification of dolphin whistles using local binary patterns. Mahdi Esfahanian, Hanqi Zhuang, and Nurgun Erdol (Elec. Eng., Florida Atlantic Univ., 777 Glades Rd., EE-96 Bldg., Boca Raton, FL 33431, mesfahan@fau.edu)

A sparse representation classifier (SRC) has been adapted and applied to spectrograms to identify bottlenose dolphin whistles by their types. The classifier that relies on near completeness of the training features renders their choice no longer crucial as long as criteria are met to assure signal sparsity. Signal sparsity is ensured via the employment of a robust, effective, and computationally simple local binary patterns (LBP) operator that eliminates the need for costly denoising and contour tracking operations. The performance of the proposed method is compared to classifier-feature combinations of the K-nearest neighbor (KNN) and support vector machine (SVM) classifiers, and feature vectors of time-frequency contour parameters, Fourier descriptors, and raw data. The experimental results demonstrate superior accuracy and robustness of the proposed method to classify dolphin whistles into distinct call types. The method can be generalized to all narrowband signals with time varying spectra.

4:00

1pAB12. Discrimination of baleen whales frequency-modulated downsweep calls with overlapping frequencies. Hui Ou, Whitlow Au (Hawaii Inst. of Marine Biol., Univ. of Hawaii, Kaneohe, HI, wau@hawaii.edu), Sofie V. Parijs (Northeast Fisheries Sci. Ctr., National Marine Fisheries Sci., Woods Hole, MA), Erin M. Oleson (Pacific Fisheries Sci. Ctr., National Marine Fisheries Sci., Honolulu, HI), and Shannon Rankin (Southwest Fisheries Sci. Ctr., National Marine Fisheries Sci., La Jolla, CA)

Spectrograms generated with the pseudo Wigner-Ville distribution (PWVD) provide much higher simultaneous time-frequency (TF) resolution compared with the traditional method using the short time Fourier transform (STFT). The WV-type spectrogram allows bioacousticians to study the fine TF structures of the sound, such as the instantaneous frequency, instantaneous bandwidth, contour slope, etc. These features set the foundation of identifying sounds that are usually considered difficult to discriminate using the traditional method. However, the PWVD requires much higher computational effort than the STFT method. In this research, the advantage of the WV spectrogram analysis was demonstrated by a case study on frequency-modulated, downsweep sounds from fin whales, sei whales, and blue whales D-calls. These calls overlapped in frequency range and have similar time duration. Automatic detection of fin, sei or blue whales FM downsweeps using the traditional spectrogram methodology tend to be ineffective because of the large temporal ambiguities needed to achieve the necessary frequency resolution. However, their WV spectrograms showed distinguishable characteristics, for example, the TF contour of fin and sei whales exhibited concave and convex shapes respectively. A support vector machine (SVM) classifier was trained and tested based on the parameters extracted from the WV spectrograms.

4:15

1pAB13. Fin whales (*Balaenoptera physalus*) in British Columbia sing a consistent song. Barbara Koot (Dept. of Zoology, Univ. of Br. Columbia, Rm. 247, 2202 Main Mall, Vancouver, BC V6T 1Z4, Canada, b.koot@fisheries.ubc.ca), John K. Ford (Fisheries and Oceans Canada, Nanaimo, BC, Canada), David Hannay (JASCO Appl. Sci., Victoria, BC, Canada), and Andrew W. Trites (Dept. of Zoology, Univ. of Br. Columbia, Vancouver, BC, Canada)

Geographic differences in fin whale song that may be related to population structure have been documented in the Atlantic and Southern Oceans. However, information on the songs and population structure of fin whales in the North Pacific is limited. We analyzed fin whale songs recorded over 9 months by an Autonomous Underwater Recorder for Acoustic Listening device (AURAL, Multi-Electronique Inc.) deployed west of Vancouver Island, British Columbia (July 2010 to March 2011). Our analysis focused on internote intervals—the song characteristic that others have shown to display the most geographic variation. We found that beginning in mid-August and continuing to the end of our study, fin whales produced one stereotyped song consisting of alternating classic (C) and backbeat (B) notes. Internote interval evolved slightly over this time period, and small differences in interval length occurred among individual songs. However, all songs shared the same note arrangement (i.e., the interval between C and B notes was always 30% longer than the interval between B and C notes). All whales recorded in this study produced a similar song, suggesting that they may belong to the same acoustic population. Future studies will help to determine the spatio-temporal boundaries of this acoustic population.

4:30

1pAB14. Vocal repertoire of Southeast Alaskan humpback whales (*Megaptera novaeangliae*). Michelle Fournet (College of Earth Ocean and Atmospheric Sci., Oregon State Univ., 425 SE Bridgeway Ave., Corvallis, OR 97333, mbellalady@gmail.com) and Andy Szabo (Alaska Whale Foundation, Seattle, WA)

Humpback whales (*Megaptera novaeangliae*) are vocal baleen whales that exhibit complex social interactions across broad spatial and temporal scales. On low latitude breeding grounds, humpback whales produce complex and highly stereotyped "songs" as well as a range of "social sounds" associated with breeding behaviors. While on their Southeast Alaskan foraging grounds, humpback whales produce vocalizations during cooperative foraging events as well as a range of unclassified vocalizations for which the social context remains unknown. This study investigates the vocal repertoire of Southeast Alaskan humpback whales from a sample of 366 vocalizations collected over a three-month period on foraging grounds in Frederick Sound, Southeast Alaska. We used a two-part classification system, which included aural-spectrogram and statistical cluster analyses, to describe and classify vocalizations. Vocalizations were classified into 19 individual call types nested within four call classes. The vocal repertoire of Southeast Alaskan humpbacks shows moderate overlap with vocalizations recorded in Atlantic foraging grounds and along the Australian migratory corridor.

4:45

1pAB15. The acoustic signature of the male northern elephant seal: Individual variation supports recognition during competitive interactions. Caroline Casey (Ecology and Evolutionary Biol., Univ. of California Santa Cruz, 100 Shaffer Rd., Santa Cruz, CA 95060, cbcasey@ucsc.edu), Colleen Reichmuth (Inst. of Marine Sci., UC Santa Cruz, Santa Cruz, CA), Selene Fregosi (Cooperative Inst. for Marine Resources Studies, Oregon State Univ. and NOAA Pacific Marine Environ. Lab., Newport, OR), Isabelle Charrier (Equipe Communications Acoustique, Université Paris Sud, Orsay, France), and Nicolas Mathevon (Laboratoire de Biologie Animale, Université Jean Monnet, Saint-Etienne, France)

Northern elephant seals (*Mirounga angustirostris*) have a polygynous breeding system in which adult males establish dominance hierarchies that determine access to females. Acoustic signaling plays an important role in settling fights between males, as stereotyped displays elicit appropriate behavioral responses from individuals without contact during an energetically demanding breeding season. To determine whether reliable differences exist in the acoustic displays of individuals and whether these differences function to convey identity, we behaviorally and acoustically sampled male seals during the breeding season. Vocalizations were recorded during competitive interactions and analyzed for spectral, temporal, and amplitude characteristics. A cross-validated discriminant function analysis revealed small differences within—and significant differences between—the calls produced by 17 adult males of known dominance status. To determine whether acoustic displays serve as individual signatures that males learn to recognize during the breeding season, we conducted playback experiments to test if having prior experience with a particular caller would influence the approach or avoidance response of the listener. Our findings reveal that these unique acoustic signals serve as individual vocal signatures, and males likely remember the identity of their rivals based on call features that have been associated with the outcome of previous competitive interactions.

5:00

1pAB16. Detection of complex sounds in quiet and masked conditions by a California sea lion (*Zalophus californianus*) and a harbor seal (*Phoca vitulina*). Kane A. Cunningham (Ocean Sci., Univ. of California at Santa Cruz, 100 Shaffer Rd., Santa Cruz, CA 95060, kacunningham413@yahoo.com), Brandon Southall (Southall Environ. Assoc., Aptos, CA), and Colleen Reichmuth (Inst. of Marine Sci., Long Marine Lab., Univ. of California at Santa Cruz, Santa Cruz, CA)

Standard audiometric data, such as absolute detection thresholds and critical ratios, are often used to inform noise-exposure limits for marine mammals. However, these data are traditionally generated using simple stimuli, such as pure-tones and flat-spectrum noise, while natural sounds tend to have more complex structure. In this experiment, detection thresholds for complex stimuli were obtained in (a) quiet and (b) masked conditions for one California sea lion and one harbor seal. For part (a), three stimuli types were synthesized, each isolating a common feature of marine mammal vocalizations: amplitude modulation (AM), frequency modulation (FM), and harmonic structure. Detection thresholds in quiet conditions were then obtained for these stimuli at frequencies spanning the functional hearing range. For part (b), the same complex signals were combined with flat-spectrum noise or shipping noise. To test how well standard hearing data predict detection of complex sounds, the results of parts (a) and (b) were compared to *a priori* predictions based on previously obtained audiogram and critical ratio data. Preliminary results indicate that absolute detection thresholds for AM and FM stimuli are reliably predicted by audiogram data, but that thresholds for harmonic stimuli are lower than predicted, in some cases by more than 10 dB.

MONDAY AFTERNOON, 2 DECEMBER 2013 PLAZA A, 1:30 P.M. TO 5:00 P.M.

Session 1pAO

Acoustical Oceanography: Contributed Papers in Acoustical Oceanography (Poster Session)

Timothy K. Stanton, Chair
Woods Hole Oceanogr. Inst., M.S. #11, Woods Hole, MA 02543-1053

Contributed Papers

1pAO1. Observations of region-specific fish behavior using long- and short-range broadband (1.5—6 + kHz) active acoustic systems. Timothy K. Stanton (Dept. Appl. Ocean Phys. and Eng., Woods Hole Oceanogr. Inst., M.S. #11, Woods Hole, MA 02543, tstanton@whoi.edu), J. Michael Jech (Northeast Fisheries Sci. Ctr., NOAA, Woods Hole, MA), Roger C. Gauss (Acoust. Div., Code 7164, Naval Res. Lab., Washington, DC), Benjamin A. Jones (Oceanogr. Dept., Naval Postgrad. School, Monterey, CA), Cynthia J. Sellers (Dept. Appl. Ocean Phys. and Eng., Woods Hole Oceanogr. Inst., Woods Hole, MA), and Joseph M. Fialkowski (Acoust. Div., Code 7164, Naval Res. Lab., Washington, DC)

Two broadband active acoustic systems, in concert with traditional narrowband systems and nets, were used to study distributions of fish in three regions within the Gulf of Maine. The long-range multi-beam broadband system detected fish out to 15 km range and the downward-looking short-range broadband system detected fish throughout the water column close behind the ship. The multi-year (2007–2011) study revealed distinct spatial patterns of fish and corresponding echo statistics in each region—diffusely distributed, sparsely distributed compact patches, and long (continuous) shoals. The broadband capabilities of the sonar systems (each spanning 1.5—6 + kHz) uniquely allow observations of resonance phenomena of the local swimbladder-bearing fish. The observed resonances were consistent with the fish species, sizes, and depths that were concurrently sampled in each area from a second research vessel. Spectral peak analysis also interestingly revealed the presence of distinct modes, which may be useful indicators of mixed-species and/or mixed-sized (e.g., juvenile and adult) assemblages of fish. [Work supported by Office of Naval Research.]

1pAO2. The history detectives: Establishing the parameters of the 1960 Perth-Bermuda antipodal acoustic propagation experiment. Brian D. Dushaw (Appl. Phys. Lab., Univ. of Washington, 1013 N.E. 40th St., Seattle, WA 98105-6698, dushaw@apl.washington.edu)

In 1960 three 300-lb explosive shots were detonated off Perth, Australia at 3 am, 22 March (local) by HMAS Diamantina to determine if those sound signals could propagate the antipodal distance to the Bermuda SOFAR station. These data offer a rare measure of the ocean temperature a half century ago, averaged across large stretches of the Southern, South Atlantic, and North Atlantic Oceans. The accuracy of these data are determined by the accuracy of the essential parameters of the experiment, e.g., the time and position of the shots. The narrative of HMAS Diamantina the night of 21 March 1960 was reconstructed from the ship's log, the captain's Report of Monthly Proceedings, and other information. The experiment was conducted with care to obtain a precise measurement, subject to the resources available to the ship at the time. The largest uncertainty is in the position of the shots, determined by triangulation from shore landmarks in the evening, celestial navigation at dawn, and dead reckoning in between. In addition, the depth was measured at the time of the shots. The 1960 position was measured to an equivalent travel-time accuracy of about 3 s, biased toward closing the range to Bermuda.

1pAO3. Fluctuations of the sound field in the presence of internal Kelvin waves in a stratified lake. Boris Katsnelson (Marine Geosci., Univ. of Haifa, 1, Universitetskaya sq, Voronezh 394006, Russian Federation, katz@phys.vsu.ru), Andrey Lunkov (Wave Res. Ctr., General Phys. Inst., Moscow, Russian Federation), and Ilia Ostrovsky (Kinneret Limnological Lab., Oceanogr. Limnological Res., Haifa, Israel)

In stratified lakes internal waves has great ecological significance since they affect mixing, resuspension, material transport, chemical regime and ecosystem productivity. Reconstruction of spatio-temporal heterogeneity of the basin scale internal waves and their accurate parameterization are important tasks. The effect of internal Kelvin waves (IKWs) on spatiotemporal variability of the mid-frequency (1 kHz) sound field in a deep lake using geoacoustic modeling is studied. It is demonstrated that IKWs cause significant fluctuations of the sound field, such as horizontal shift of interference structure. This shift can be easily measured in situ and used for practical reconstruction of IKW parameters. Overall, it is suggested implementing the low-cost geoacoustic methodology for accurate parameterization of the basin scale internal waves and studying their dynamics.

1pAO4. Measurements of diel variation of acoustic backscatter power from phytoplankton. Hansoo Kim, Tae-Hoon Bok, Juho Kim, Dong-Guk Paeng (Dept. of Ocean Syst. Eng., Jeju National Univ.,102 Jejudaehakno, Jeju-si, Jeju Special Self-Governing Province 690-756, South Korea, hansoo5714@naver.com), Md Mahfuzur Rahman Shah, and Joon-Baek Lee (Dept. of Earth and Marine Sci., Jeju National Univ., Jeju, South Korea)

Phytoplankton is a primary producer in the ocean, and its vegetative activity plays an important role in controlling the global environment. In this study, the high-frequency acoustic signals were collected to evaluate the photosynthetic activity of phytoplankton during a day. The integrated backscatter power (IBP) from a dinoflagellate *Cochlodinium polykrikoides* was measured by a 5 MHz acoustic system during a 5-day cultivation period with a 14 h:10 h light:dark cycle. IBP increased by 0.6 dB in five days, but varied by 0.83 ± 0.1 dB during an irradiance cycle. The daily increase in IBP was a result of an increase in the number of cells during cultivation, while the diel variation was partly resulted from the variation of volume of cells by photosynthesis. In addition, cell division and separation might affect IBP. IBP of another dinoflagellate *Amphidinium carterae* Hulburt was also measured by a 10MHz transducer during a 3-day cultivation period while cell volume and photosynthetic capacity were measured four times a day (07:00, 12:00, 19:00, 24:00). This study suggests that high-frequency acoustics may be a meaningful tool to investigate the photosynthetic metabolism of a phytoplankton cell. [This work was supported by Defense Acquisition Program Administration and Agency for Defense Development under the contract UD130007DD.]

1pAO5. Broadband normal mode energy metrics in the presence of internal waves. Georges Dossot, Steven Crocker (NUWC, 1176 Howell Str., Bldg. 1320, Code 1524, Rm. 260, Newport, RI 02841, georges.dossot@navy.mil), James H. Miller, Gopu R. Potty (Ocean Engineering, Univ. of Rhode Island, Narragansett, RI), and Mohsen Badiey (College of Marine Studies, Univ. of Delaware, Newark, DE)

During the Shallow Water 2006 experiment a J-15 acoustic source deployed from the Research Vessel Sharp transmitted broadband (100–500 Hz) chirp signals 15 km away from a vertical line array. The array was intentionally positioned near the shelf-break front and in an area where internal waves are known to occur. Normal mode decomposition helps provide clues regarding the physics behind signal fluctuations due to internal waves, but often analyses are accomplished at single frequencies. A broadband modal beamformer approach is offered that extracts separate modal arrivals. A method for a mode-dependent matched filter is suggested which helps extract the separate arrivals in a low-signal environment. These methods can be used to compute mode-independent energy statistics which help explain signal fluctuations as an internal wave traverses the source-receiver path. [Work sponsored by the Office of Naval Research.]

1pAO6. Acoustic mapping of ocean currents using networked distributed sensors. Chen-Fen Huang (Inst. of Oceanogr., National Taiwan Univ., No. 1, Sec. 4, Roosevelt Rd., Taipei 10617, Taiwan, chenfen@ntu.edu.tw), TsihC Yang, and Jin-Yuan Liu (Inst. of Appl. Marine Phys. and Undersea Technol., National Sun Yat-sen Univ., Kaohsiung, Taiwan)

Distributed underwater sensors are expected to provide oceanographic monitoring over large areas. As fabrication technology advances, low cost sensors will be available for many uses. The sensors communicate to each other and are networked using acoustic communications. This paper first studies the performance of such systems for current measurements using tomographic inversion approaches to compare with that of a conventional system which distributes the sensors on the periphery of the area of interest. It then proposes two simple signal processing methods for ocean current mapping (using distributed networked sensors) aimed at realtime in-buoy processing, Tomographic inversion generally requires solving a challenging high dimensional inverse problem, involving substantial computations. Given distributed sensors, currents can be constructed locally based on data from neighboring sensors. It is shown using simulated data that results obtained using distributed processing are similar to those obtained from conventional tomographic approaches. The advantage for distributed systems is that by increasing the number of nodes, one gains a much more improved performance. Furthermore, distributed systems use much less energy than conventional tomographic system for the same area coverage. Experimental data from an acoustic communication and networking experiment are used to demonstrate the feasibility of acoustic current mapping.

1pAO7. Low frequency scattering from fish schools: A comparison of two models. Maria P. Raveau (Hydraulic and Environ. Eng., Pontificia Universidad Catolica de Chile, Vicuña Mackenna 4860, Macul, Santiago 7820436, Chile, mpraveau@uc.cl) and Christopher Feuillade (Phys. Dept., Pontificia Universidad Catolica de Chile, Santiago, Chile)

A theoretical comparison between two scattering models for fish schools was performed. The effective medium approach, based on the work of Foldy [Phys. Rev. **67**, 107–109 (1945)], which has previously been used to describe scattering from bubble clouds, was used to calculate the variations of the scattering length of a fish school with frequency, and with azimuth. Calculations were also performed with a model used previously to study collective back scattering from schools of swim bladder fish, which incorporates both multiple scattering effects between fish, and coherent interactions of their individual scattered fields [J. Acoust. Soc. Am., **99**(1), 196–208 (1996)]. Two different packing algorithms were used, in order to investigate the influence of the spatial distribution of fish on the scattering response of the school. Comparison of the two models shows good agreement in the forward scattering direction, where no frequency dependent interference effects are observed. The models indicate divergent results in the back scattering direction, where the arrangement of fish in the school strongly affects the scattering amplitude. The upper frequency limit of the effective medium approach is also discussed, and the effect of the depth of the school in the water column. [Research supported by ONR.]

1pAO8. Sound absorption by fish schools: Forward scattering theory and data analysis. Christopher Feuillade (Facultad de Física, Pontificia Universidad Católica de Chile, Av. Vicuña Mackenna 4860, Santiago 8920007, Chile, chris.feuillade@gmail.com) and María P. Raveau (Facultad de Ingeniería, Pontificia Universidad Católica de Chile, Santiago, Chile)

A model used previously to study collective back scattering from fish schools [J. Acoust. Soc. Am., **99**(1), 196–208 (1996)] is used to analyze the forward scattering properties of these objects. There is an essential physical difference between back and forward scattering from fish schools. Strong frequency dependent interference effects, which affect the back scattered field amplitude, are absent in the forward scattering case. This is critically important when analyzing data from fish schools to study their size, the species and abundance of fish, and fish behavior. Transmission data can be processed to determine the extinction of the field by a school. The extinction of sound depends on the forward scattering characteristics of the school, and inversion of absorption data to provide information about the fish should be based upon a forward scattering paradigm. Results are presented of an analysis of transmission data obtained during the Modal Lion experiment in September 1995, and reported by Diachok [J. Acoust. Soc. Am., **105**(4), 2107–2128 (1999)]. The analysis shows that using forward scattering typically leads to significantly larger estimates of fish abundance than previous analysis based on back scattering approaches. [Research supported by ONR.]

1pAO9. Synthetic aperture geoacoustic inversion in the presence of radial acceleration dynamics. Bien Aik Tan, Peter Gerstoft, Caglar Yardim, and William Hodgkiss (Univ. of California San Diego, 9500 Gilman Dr., La Jolla, CA 92093-0238, btan@ucsd.edu)

Traditionally matched-field geoacoustic inversion experiments sampled the acoustic field on long arrays and require powerful transmissions in order to reduce parameter uncertainty. However, single-hydrophone based geoacoustic inversion methods are attractive compared to the ones using long arrays. A low signal to noise ratio (SNR), single-receiver, broadband, and frequency coherent matched-field inversion that exploits coherently repeated transmissions to improve estimation of the geoacoustic parameters have been previously proposed. The long observation time creates a synthetic aperture due to relative source-receiver motion. This paper extends broadband synthetic aperture geoacoustic inversion to cases where the velocity of the source/receiver changes. In addition, a pulse-by-pulse coherent processing using the Bayesian approach is proposed. The method is

demonstrated with low SNR, 100–900 Hz LFM data from the Shallow Water 2006 experiment.

1pAO10. Intensity statistics for dual-band, 500 km, ocean acoustic transmissions in the Philippine Sea 2010 experiment. Andrew A. Ganse, Rex K. Andrew, James A. Mercer (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, aganse@apl.washington.edu), Peter F. Worcester, and Matthew A. Dzieciuch (Scripps Inst. of Oceanogr., Univ. of California, San Diego, La Jolla, CA)

The May 2010 component of the North Pacific Acoustic Laboratory's Philippine Sea experiment included low-frequency transmissions over a 500 km path from the APL-UW ship-suspended multi-port source to a distributed vertical line array (DVLA) with 149 working hydrophones spaced over most of the water column. The multi-port source has two acoustic resonances at approximately 200 and 300 Hz. To accommodate those resonances in this experiment, two M-sequence signals at those center frequencies were transmitted simultaneously. Data recorded over a 60 h data window in the experiment show deep fades in both frequency bands, i.e., frequency-dependent variations of 10–20 dB in acoustic arrival intensities on the DVLA. The measurements are processed with Wiener filtering based on source-monitoring receptions of the transmission on a hydrophone 20 m from the source, to compensate for effects of the power amplifier and the transducer. Statistics of the received acoustic intensities are reported for both frequency bands. Correlations in the intensity between these broadly separated frequency bands allow one to explore questions of the role in arrival-splitting (micro-multipathing) in the deep fades observed in this experiment. [Work supported by ONR.]

MONDAY AFTERNOON, 2 DECEMBER 2013 GOLDEN GATE 2/3, 1:00 P.M. TO 3:45 P.M.

Session 1pBA

Biomedical Acoustics: Bubble Detection in Diagnostic and Therapeutic Applications II

Charles C. Church, Chair

National Ctr. for Physical Acoust., Univ. of Mississippi, 1 Coliseum Dr., University, MS 38677

Contributed Papers

1:00

1pBA1. Nonlinear dynamics and control of ultrasound contrast agent microbubbles. James M. Carroll, Leal K. Lauderbaugh, and Michael L. Calvisi (Department of Mechanical and Aerospace Engineering, Univ. of Colorado, Colorado Springs, 1420 Austin Bluffs Parkway, Colorado Springs, CO 80918, mcalvisi@uccs.edu)

The nonlinear response of ultrasound contrast agent microbubbles is well-known and is important for both diagnostic and therapeutic purposes. A model of thin-shelled, spherical contrast agents subject to ultrasound is presented based on a modified form of the Rayleigh-Plesset equation combined with terms to account for the shell influence. Techniques from dynamical systems theory are employed to analyze the model and elucidate the complex behavior of contrast agents forced by ultrasound. It is shown that the contrast agent can undergo a sequence of bifurcations as the acoustic pressure amplitude is increased, leading to a transition from periodic to chaotic oscillations at sufficiently large forcing. Yet even at high forcing, regions of periodic behavior exist. The response of the contrast agent, however, is strongly dependent on the shell properties and the applied acoustic forcing. Furthermore, a nonlinear, sliding mode controller is developed and applied to the spherical contrast agent model to demonstrate the feasibility of using the nonlinear controller to modulate the contrast agent response through the incident ultrasound. Applications of the nonlinear control system to contrast agents include radius stabilization in the presence of an acoustic wave, excitation of radial growth and subsequent collapse, and generation of periodic radial oscillations while a contrast agent is within an acoustic forcing regime known to cause a chaotic response.

1:15

1pBA2. A numerical model for large-amplitude spherical bubble dynamics in tissue. Matthew T. Warnez and Eric Johnsen (Mech. Eng., Univ. of Michigan, 405 N Thayer, Ann Arbor, MI 48104, mwarnez@umich.edu)

In a variety of therapeutic and diagnostic ultrasound procedures (e.g., histotripsy, lithotripsy, and contrast-enhanced ultrasound), cavitation occurs in soft tissue, which behaves in a viscoelastic fashion. While stable bubble oscillations may occur in ultrasound, the most dramatic outcomes (tissue ablation, bleeding, etc.) are usually produced by inertial cavitation. Historically, Rayleigh-Plesset equations have been used to investigate the dynamics of spherical bubbles, including in biomedical applications. For large-amplitude bubble oscillations in tissue, it is clear that compressibility, heat transfer and nonlinear viscoelasticity play important roles. However, no existing model includes all of these effects. To address this need, we use a compressible Rayleigh-Plesset equation (Keller-Miksis) adjoined with heat conduction in conjunction with an upper-convected Zener viscoelastic model, which accounts for relaxation, elasticity, and viscosity. The partial differential equations describing the stress tensor components in the surrounding medium are solved using a spectral collocation method. The method proves to be robust even for strong bubble collapse. Numerical comparisons with previous models are made, comparisons to experiments are included, and the dependence of bubble dynamics on viscoelastic parameters is explored. This model is used to revisit the inertial cavitation threshold in biomedical settings.

1:30

1pBA3. Delay differential equations for single-bubble dynamics in a compressible liquid. Derek C. Thomas (Dept. of Phys. and Astronomy, Brigham Young Univ., N223 ESC, Provo, UT 84602, derekcthomas@gmail.com), Yurii A. Ilinskii (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), and Mark F. Hamilton (Dept. of Mech. Eng., The Univ. of Texas at Austin, Austin, TX)

Most common methods to include the effects of liquid compressibility in models for single-bubble dynamics rely on series expansions to some order in the inverse of the sound speed in the liquid. It has been shown that the Keller-Miksis model for single-bubble dynamics can be obtained from a series expansion of a delay differential equation related to the Rayleigh-Plesset equation for a single bubble. The iterative approach used to obtain the series expansion of the delay-based model becomes unworkable for more complicated models of bubble dynamics. Therefore, to provide an alternative, simpler method to model the effects of liquid compressibility, the

delay differential equation model proposed by Ilinskii and Zabolotskaya [J. Acoust. Soc. Am. **92**, 2837 (1992)] is analyzed directly. The results of the delay-based formulations are compared to those produced by models based on common series expansions. Alternative formulations of the delay differential equation are also considered and compared.

1:45

1pBA4. Shell material parameter measurements of polymer ultrasound contrast agents. Pavlos Anastasiadis, John S. Allen (Mech. Eng., Univ. of Hawaii, 2540 Dole St., Holmes Hall 302, Honolulu, HI 96822, alleniii@hawaii.edu), Parag V. Chitnis, and Jeffrey A. Ketterling (Riverside Res., New York, NY)

Polymer shelled ultrasound contrast agents have been used in ultrasound imaging and tissue perfusion studies. The destruction of the agent produced by the rupture of the shell often through complicated buckling has been quantified with overpressure experiments and optical visualization. Approximate material parameters have been correspondingly estimated with this methodology but additional steps are needed to translate for a viable high throughput technique. Ultra-high frequency acoustic microscopy (1 GHz) provides a non-invasive method to image and measure the shell's elastic properties. Scanning acoustic microscopy at 1 GHz is used to determine the shell density and elastic modulus for polymer shelled agents of three different shell thicknesses. The effect bending thickness is estimated for comparison with overpressure experiments.

2:00

1pBA5. Subharmonic response and threshold of polymer ultrasound contrast agents. RIntaro Hayashi, John S. Allen (Mech. Eng., Univ. of Hawaii, 2540 Dole St., Holmes Hall 302, Honolulu, HI 96822, alleniii@hawaii.edu), Jonathan Mamou, Parag V. Chitnis, and Jeffrey A. Ketterling (Riverside Res., New York, NY)

Analytical expressions for subharmonic threshold previously developed for free gas bubbles often been generalized and used for ultrasound contrast agents. However, many of those formulations were developed under the assumptions of mono-frequency steady state, continuous acoustic forcing which are typically not applicable to diagnostic and therapeutic ultrasound. Nonstationary subharmonic responses are investigated by analytical and numerical methods. The subharmonic threshold is investigated as means to differentiate between the proposed constitutive formulations of polymer shells in terms the shear modulus parameter. The potential influence of other agents in close proximity on the subharmonic response and threshold is highlighted with respect to the phase interaction between coupled agents.

2:15

1pBA6. *In vitro* acoustic characterization of echogenic liposomes with a polymerized lipid bilayer (Pol-ELIPs). Shirshendu Paul (Mech. Eng., Univ. of Delaware, 130 Acad.. St., Newark, DE 19716, spaul@udel.edu), Rahul Nahira, Sanku Mallik (Pharmacy, North Dakota State Univ., Fargo, ND), and Kausik Sarkar (Mech. and Aerosp. Eng., George Washington Univ., Washington, Virginia)

Liposomes are typically lipid bilayer vesicles with an aqueous interior. A modified preparation protocol can make them echogenic, i.e., capable of scattering incident acoustic pulses, by incorporating gas pockets in their structure. Here we study echogenic liposomes with a polymerized lipid bilayer that can potentially improve their stability and nonlinear behavior. The Pol-ELIPs prepared were found to have a very polydisperse size distribution with an average size of 3μm. The frequency dependent attenuation experiment did not show any distinct peak due to the high polydispersity. They showed echogenicity in our *in vitro* scattering experiments, even without the presence of bovine serum albumin in the reconstituting media. Scattered response measured from Pol-ELIP suspension showed non-linear behaviors with distinct second-harmonic and subharmonic peaks, in contrast to non-polymerize ELIPs that did not show any subharmonic response. Scattered fundamental, second-harmonic and subharmonic responses at a lipid concentration of 1μg/ml showed up to 35, 30, and 35 dB enhancement, respectively. The subharmonic response showed all its characteristic features—its appearance only above a threshold excitation level (150 kPa) and then a sharp rise with a subsequent saturation. The study proves the echogenicity of the novel Pol-ELIPs with interesting nonlinear properties. [Work partially supported by NSF.]

2:30

1pBA7. Effects of ambient pressure variation on the subharmonic response from contrast microbubbles: Effects of encapsulation. Nima Mobaderasny (Mech. and Aerosp. Eng., George Washington Univ., 802 22nd St. NW, Phillips Hall 738, Washington, DC, sany@gwu.edu) and Kausik Sarkar (Mech. and Aerosp. Eng., George Washington Univ., Washington, Virginia)

We investigate the ambient pressure dependent subharmonic response from encapsulated contrast microbubbles for non-invasive estimation of local blood pressure. We have previously found that the subharmonic response from a free microbubble can either increase or decrease or vary nonmonotonically [Katiyar *et al*., J. Acoust. Soc. Am. **129**, 2325–2335] depending on the ratio of the excitation frequency to the resonance frequency (f/f0). Here, we extend this work to encapsulated microbubbles assuming various interfacial rheological models for the encapsulation. With an exponential elasticity model, the general trend remains similar to that of the free bubble. However, one also obtains chaotic oscillations at very low excitation frequencies and smaller damping coefficients. The specific trends are also disrupted when excitation pressure is increased or bubble size is changed. In the talk we will discuss the effects of variations in radius and different parameters of the encapsulation. [Work partially supported by NSF.]

2:45

1pBA8. Probing the bioeffects of cavitation at the single-cell level. Fang Yuan, Georgy Sankin, Chen Yang, and Pei Zhong (Mech. Eng. & Mater. Sci., Duke Univ., Science Dr., Durham, NC 27713, fang.yuan@duke.edu)

Cavitation induced bioeffects has not been resolved satisfactorily due to the randomness in the inception and bubble dynamics produced by ultrasound. We have developed a microfluidic system to observe consistently the interaction of laser-generated tandem bubbles (50 μm in diameter) with resultant jet formation, cell deformation, and localized membrane rupture with progressive diffusion of propodium iodide (PI) into individual HeLa cells placed nearby. We observe a clear stand-off distance (SD) dependence in the bioeffects produced by the tandem bubbles. At SD of 10 μm, all cells underwent necrosis with high, unsaturated level of PI uptake. At SD of 20~30 μm, 58 to 80% of the cells showed repairable membrane poration with low to medium but saturated level of PI uptake. Within this range, the sub-population of cells that survived without apoptosis increased from ~9% at SD of 20 μm to ~70% at SD of 30 μm. The maximum PI uptake, pore size, and estimated membrane strain, however, could vary by more than an order of magnitude at each SD. At SD of 40 μm, no detectable PI uptake was observed. This experimental system provides a unique tool to probe the bioeffects of cavitation at the single cell level.

3:00

1pBA9. Magnetic targeting of microbubbles at physiologically relevant flow rates. Joshua W. Owen, Paul Rademeyer, and Eleanor Stride (Dept. of Eng. Sci., Univ. of Oxford, Inst. of Biomedical Eng., Old Road Campus Res. Bldg., Oxford OX3 7DQ, United Kingdom, joshua.owen@eng.ox.ac.uk)

The localization of microbubbles to a target site has been shown to be essential to their effectiveness in ultrasound mediated drug delivery and gene therapy. The incorporation of super paramagnetic nanoparticles into the microbubble coating enables them to be manipulated using an externally applied magnetic field. Magnetic microbubbles have been shown to be effective in therapeutic delivery both *in vitro* and *in vivo* in a mouse model. The aim of this experiment was to determine under what conditions in the human body magnetic microbubbles can be successfully imaged and targeted. Different flow rates and shear rates were generated in a tissue mimicking phantom and targeting was observed using a 9.4 MHz ultrasound imaging probe. For the highest shear rates, targeting was also observed optically. Results indicate that magnetic microbubbles can be successfully targeted at shear rates found in the human capillary system (>1000/s) and at flow rates found in the veins and smaller arteries (~200 ml/s). Successful retention was also demonstrated in a perfused porcine liver model simulating conditions *in vivo*. This study provides further evidence for the potential of magnetic microbubbles for targeted therapeutic delivery.

3:15

1pBA10. Device for detection of cavitation in by spectral analysis in megasonic cleaning. Claudio I. Zanelli, Samuel M. Howard, Dushyanth Giridhar, and Petrie Yam (Onda Corp., 1290 Hammerwood Dr., Sunnyvale, CA 94089, cz@ondacorp.com)

The authors present a device and a method to detect cavitation in megasonic cleaning environments. The device is small enough to fit between wafers in the semiconductor industry, allowing the monitoring of the cleaning process.

3:30

1pBA11. A meshless bubble filter for an extracorporeal circulation using acoustic radiation force. Koji Mino, Manami Kataoka, Kenji Yoshida (Faculty of Life and Medical Sci., Doshisha Univ., 1-3 Tataramiyakodani, Kyotanabe, Kyoto 610-0321, Japan, dmm1017@mail4.doshisha.ac.jp), Daisuke Koyama (Faculty of Sci. and Eng., Doshisha Univ., Kyotanabe, Kyoto, Japan), Kentaro Nakamura (Precision and Intelligence Lab., Tokyo Inst. of Technol., Yokohama, Kanagawa, Japan), Masayoshi Omori, Shigeki Kawarabata, Masafumi Sato (Central Res. Lab. JMS Co., Ltd., Hiroshima, Hiroshima, Japan), and Yoshiaki Watanabe (Faculty of Life and Medical Sci., Doshisha Univ., Kyotanabe, Kyoto, Japan)

Arterial filters are employed in extracorporeal circulations to remove microbubbles and thrombus from the blood flow and prevent the emboli. Filters with mesh structure have a risk to generate the thrombi when the blood flows through it. In this report, a meshless filter using ultrasound is discussed. The filter consists of an aluminum cylinder (length: 130 mm; inner diameter: 30 mm) and two annular ultrasound PZT transducers. The filter has one inlet at the center of the side and two outlets at both ends. By exciting the transducer, the acoustic traveling wave can be generated in the liquid inside the filter. Air bubbles flowing from the inlet can be led toward the outlet by acoustic radiation force. The characteristics of the filter were investigated through a circulation system using distilled water at the driving frequencies of 200 kHz and 1 MHz. Flow and injected air were set as 5.0 l/min and 10 ml/min, respectively. The microbubbles were filtered by using ultrasound and the amount of filtered bubbles was increased with the input voltage to the transducer: 50.6 and 53.7% of microbubbles were filtered at 200 kHz and 1 MHz, respectively, when the input voltage was 100 Vpp.

MONDAY AFTERNOON, 2 DECEMBER 2013 — CONTINENTAL 1, 1:00 P.M. TO 4:15 P.M.

Session 1pMU

Musical Acoustics: General Musical Acoustics

Randy Worland, Chair
Physics, Univ. of Puget Sound, 1500 N. Warner, Tacoma, WA 98416

Contributed Papers

1:00

1pMU1. Digital fabrication of vocal tract models from magnetic resonance imaging during expert pitch bending on the harmonica. John Granzow (CCRMA, Stanford Univ., 660 Lomita Dr., Stanford, CA 94305, johknee5@gmail.com), Peter Egbert (Ophthalmology, Stanford Univ., Palo Alto, CA), David Barrett (CCRMA, Stanford Univ., San Jose, CA), Thomas Rossing (CCRMA, Stanford Univ., Palo Alto, CA), and Lewis Shin (Radiology, Stanford Univ., Palo Alto, CA)

Expressive pitch bending on the harmonica requires the acoustic coupling of contiguous free reeds, an effect known to arise from highly constrained vocal postures. These techniques have been difficult to demonstrate and instructors often rely on associated vowels to help novices achieve the required constriction of the vocal tract. Magnetic Resonance Imaging (MRI) was recently used to expose the precise vocal contours involved in such expert pitch bends (Egbert, *et al*., J. Acoust. Soc. Am. **133**, 3590 (2013). To further this investigation, we process the MRI data using 3D slicing software, generating digital models of the vocal tract during sustained bends. In addition to providing volumetric data, the models are also fabricated with fusion deposition modeling and tested on real harmonicas. These tests reveal error tolerances in the conversion from MRI slices to 3d printed models when working with geometries that are highly constrained by a desired acoustic output. Furthermore, comparisons between human performance and simulated output provide clues to the contribution of factors not reproduced in the plastic models such as pharynx dilation.

1:15

1pMU2. Attack transients in free reed instruments. Jennifer Biernat (Mansfield Univ. of Pennsylvania, 172 South Broad St., Nazareth, PA 18064, jenbiernat@yahoo.com) and James P. Cottingham (Physics, Coe College, Cedar Rapids, IA)

Attack transients of harmonium-type reeds from American reed organs have been studied in a laboratory setting with the reeds mounted on a wind chamber. Several methods were used to initiate the attack transients of the reeds, and the resulting displacement and velocity waveforms were recorded using a laser vibrometer system and electronic proximity sensors. The most realistic procedure had a pallet valve mechanism simulating the initiation of an attack transient that depressing an organ key would provide. Growth rates in vibrational amplitude were then measured over a range of blowing pressures. Although the fundamental transverse mode is dominant in free reed oscillation, the possibility of higher transverse modes and torsional modes being present in transient oscillation was also explored. The reeds studied are designed with a spoon-shaped curvature and a slight twist at the free end of the reed tongue, intended to provide a more prompt response, especially for larger, lower-pitched reeds for which a slow attack can be a problem. The effectiveness of this design has been explored by comparing these reeds with equivalent reeds without this feature. [Work supported by National Science Foundation REU Grant PHY-1004860.]

1:30

1pMU3. A study of oboe reed construction. Julia Gjebic, Karen Gipson (Physics and Music, Grand Valley State Univ., 1 Campus Dr., 118 Padnos Hall, Allendale, MI 49401, gjebicj@mail.gvsu.edu), and Marlen Vavrikova (Music, Grand Valley State Univ., Allendale, MI)

The construction of reeds is of much interest in the oboe community, because professional oboists spend as much time making reeds as they do practicing. Each oboist uses an individual methodology resulting from different training and personal physiology. To investigate how different reed construction affects the resulting sound, 22 professional oboists were recruited to make three reeds apiece for this study. First, a controlled batch of reed cane (internodes of the grass *Arundo Donax*) was selected based on microscopic inspection of cellular composition as well as macroscopic physical attributes. For most of the participants, the cane was then processed identically to the stage known as a *blank*, after which the participants finished their reeds according to their usual methods. (The few participants who made their own blanks still used the controlled cane and also a controlled *staple*, the metal cylinder that attaches the reed to the oboe.) The sound spectra of recordings of each participant playing on his/her respective reeds were analyzed, as was a spectrum of the *crow* (sound without the oboe attached) of each reed in an anechoic chamber. These spectra were correlated to measured physical characteristics of the reeds.

1:45

1pMU4. Spectral character of the resonator guitar. Daniel Ludwigsen (Kettering Univ., 1700 University Ave., Flint, MI 49504, dludwigs@kettering.edu)

The resonator guitar was invented in the 1920s, with one or more metal cone resonators set into the body. These additions were originally meant to amplify the sound of the acoustic guitar for performance in a band. The distinct timbre of the resonator ensured that the design survived even after electrification, especially in blues and bluegrass genres. A study of the sound radiated from different models of resonator guitars, as well as a similar standard acoustic guitar, compares spectral features to understand the unique sound of the resonator guitar.

2:00

1pMU5. A method for obtaining high-resolution directivities from the live performance of musical instruments. Nicholas J. Eyring (Dept. of Phys. and Astronomy, Brigham Young Univ., 485 N 450 East, Orem, UT 84097, eyringj@gmail.com), Timothy W. Leishman, and William J. Strong (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT)

Directivity measurements for live performance of musical instruments present several experimental challenges, including the need for musicians to play consistently and reproducibly. Some researchers have chosen to implement fixed, limited-element microphone arrays surrounding instruments for rough directivity assessments. Unfortunately, with practical numbers of microphones and data acquisition channels, this approach limits spatial resolution and field decomposition bandwidth. Higher-resolution data may be obtained with a given microphone and channel count by rotating a musician in sequential azimuthal angle increments under a fixed semicircular microphone array. The musician plays a selected note sequence with each increment, but corrections must be made for playing variability. This paper explores the development of this method, which also uses rotating reference frame microphones and frequency response function measurements. The initial developments involve a loudspeaker, with known directivity, to simulate a live musician. It radiates both idealized signals and anechoic recordings of musical instruments with random variations in amplitude. The presentation will discuss how one can reconstruct correct source directivities from such signals and the importance of reference microphone placement when using frequency response functions. It will also introduce the concept of coherence maps as tools to establish directivity confidence.

2:15

1pMU6. Difference thresholds for melodic pitch intervals. Carolyn M. McClaskey (Cognit. Sci., Univ. of California, Irvine, 4308 Palo Verde Rd., Irvine, CA 92617-4321, carolyn.mcclaskey@gmail.com)

Pitch-interval processing is an important aspect of both speech and music perception. The current study investigated the extent to which relative pitch processing differs between intervals of the western musical system and whether these differences can be accounted for by the simplicity of an interval's integer-ratio. Pitch-interval discrimination thresholds were measured using adaptive psychophysics for sequentially presented pure-tone intervals with standard distances of 1 semitone (minor second, 16:15), 6 semitones (the tritone, 45:32), and 7 semitones (perfect fifth, 3:2) at both high (1500–5000 Hz) and low (100–500 Hz) frequency regions. Results show similar thresholds across all three interval distances with no significant difference between low and high frequency regions. Consistent with previous studies, thresholds obtained from musicians were considerably lower than those from non-musicians. Data support enhanced pitch-interval perception by musicians but argue against an effect of frequency-ratio simplicity in the case of pure-tone melodic intervals.

2:30

1pMU7. Analyzing the time variance of orchestral instrument directivities. Adam T. Buck and Lily M. Wang (Durham School of Architectural Eng. and Construction, Univ. of Nebraska-Lincoln, 1110 S. 67th St., Omaha, NE 68182-0816, atbuck@unomaha.edu)

Thirteen-channel anechoic recordings of 20-s musical excerpts for violin, flute, and bass trombone were analyzed to determine how significantly an instrument's directivity changes as it is played. A clearer understanding of the time variance of instrument directivities may help in the advancement of room acoustical computer modeling, as the source properties are crucial to the simulation's accuracy. Previous research has accurately documented the static spatial characteristics of instrument directivities. A multi-channel auralization technique incorporating time-varying source behavior has been developed, but the time variance of the directivities has yet to be explored. In this project, a time-windowing technique was utilized to calculate the directivity index throughout each channel recording for each instrument. The results were analyzed in terms of maximum directivity index and a sampling of complete directivity patterns, and finally used to explore quantification methods. The time variations displayed by each instrument's directivity were unique in terms of magnitude, direction, and frequency. Calculating the average change in directivity index for each channel at each frequency band was found to be a suitable method for summarizing the results. [Work supported by a UNL Undergraduate Creative Activities and Research Experience Grant.]

2:45

1pMU8. Testing a variety of features for music mood recognition. Bozena Kostek and Magdalena Plewa (Gdansk Univ. of Technol., Narutowicza 11/12, Gdansk 80-233, Poland, bokostek@audioacoustics.org)

Music collections are organized in a very different way depending on a target, number of songs or a distribution method, etc. One of the high-level feature, which can be useful and intuitive for listeners, is "mood." Even if it seems to be the easiest way to describe music for people who are non-experts, it is very difficult to find the exact correlation between physical features and perceived impressions. The paper presents experiments aimed at testing a variety of low-level features dedicated to music mood recognition. Musical excerpts to be tested comprise individual (solo) tracks and mixes of these tracks. First FFT- and wavelet-based analyses, performed on musical excerpts, are shown. A set of "energy-based" parameters is then proposed. These are mainly rms coefficients normalized over the total energy derived from wavelet- based decomposed subbands, variance and some statistical moments. They are then incorporated into the feature vector describing music mood. Further part of experiments consists in testing to what extent these features are correlated to the given music mood. Results of the experiments are shown as well as the correlation analysis between two main mood dimensions—Valence and Arousal assigned to music excerpts during the subjective tests.

3:00–3:15 Break

3:15

1pMU9. Individuals with congenital amusia respond to fractal music differently from normal listeners. 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, fangliufangliu@gmail.com), Sherri L. Livengood (The Roxelyn and Richard Pepper Dept. of Communication Sci. & Disord., Northwestern Univ., Evanston, IL), Cunmei Jiang (Music College, Shanghai Normal Univ., Shanghai, China), 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 neurogenetic disorder predominately defined by impaired perception of musical tonal relationships. This study examined amusics' responses to a gradient of pitch interval complexity in fractal music. Eighteen Mandarin-speaking amusics and 18 controls rated random tone sequences for perceptual (complexity, melodicity) and affective (interest, ease, mood) attributes, and performed a recognition memory task. Sequences were created using fractal model ($1/f^{ß}$) with ß-values ranging from 0.0 (most complex) to 2.6 (least complex). As predicted, both groups rated complexity based on the ß-values, demonstrating that amusics perceived the gradient of pitch interval complexity. However, amusics' ratings deviated from controls in measures of melodicity, affect, and memory performance. For controls, moderately complex sequences (fractal ß-values = 1.4–1.6) were rated the most melodious, drove the highest emotional responses, and were the easiest to remember, whereas amusics' ratings did not respond to this range, but rather followed a more linear trend. These findings suggest that amusics not only have problems with perception of pitch interval relationships (not complexity), but also lack heightened sensitivity to the moderate range in fractal music. This deficit is reflected in broader musical processing including the perception of melody, affective response, and memory for musical sequences. [This work was supported by National Science Foundation Grant BCS-0719666 to P.C.M.W. and Shanghai Normal University funding to C.J.]

3:30

1pMU10. Marching band hearing responses: Indoor rehearsal vs outdoor performance configurations. Glenn E. Sweitzer (Sweitzer LLP, 4504 N Hereford Dr., Muncie, IN 47304, glenn.sweitzer@gmail.com)

Marching band performer responses are gathered to compare how well each performer hears: (1) oneself playing; (2) others playing the same part and instrument; and (3) others playing different parts, by instrument. The measurement protocol is repeated in an indoor purpose-built band rehearsal venue with adjustable sound diffusive vs absorptive wall treatments and, at an open-air outdoor performance venue (grass playing field). All scaled responses are gathered using a personal response system, providing immediate, simultaneous, and anonymous responses that can be compared online. Prior to each set of responses, all performers play together a well-rehearsed score. Responses vary indoors by absorptive treatment and band configuration and, outdoors, by band configuration. Discussion will focus on how marching bands, indoors or outdoors, might be reconfigured to improve hearing for performers, directors, and audiences.

3:45

1pMU11. Representation of musical performance "grammar" using probabilistic graphical models. Gang Ren, Zhe Wen, Xuchen Yang, Cheng Shu, Fangyu Ke, and Mark Bocko (Dept. of Elec. and Comput. Eng., Univ. of Rochester, Rochester, NY 14627, g.ren@rochester.edu)

As a versatile data modeling tool, probabilistic graphical model can be applied to model the complex dependency structures encoded in the contextual "grammar" of music performance. The musical performance grammar here refers to the relational structures of the sonic features extracted from music performances. In the existing literature, the data structure of musical expressive grammar is usually modeled as rule list, following the grammatical format of natural language processing applications. In this work, we apply the representation format of probabilistic graphical model to musical performance features to extend the conventional rule-list format. We choose probabilistic graphical model as an "upgraded" representation for two reasons. First, probabilistic graphical model provides enhanced representation capability of relational structures. This feature enables us to model the complex dependency structure that the conventionally rule list cannot handle. Second, the graphical format of probabilistic graphical model provides an intuitive human-data interface and allows in-depth data visualization, analysis, and interaction. We include the representation and analysis examples of musical performance grammar obtained from both manual analysis and automatic induction. We also implemented interpretation tools that interface the rule-list format and the probabilistic graphical model format to enable detailed comparison with existing results of musical performance analysis.

4:00

1pMU12. Vibrato lab: A signal processing toolbox for woodwind sound analysis. Zhe Wen, Xuchen Yang, Cheng Shu, Fangyu Ke, Gang Ren, and Mark Bocko (Dept. of Elec. and Comput. Eng., Univ. of Rochester, Rochester, NY 14627, g.ren@rochester.edu)

Vibrato is an important performance technique for woodwind instrument that produces amplitude and frequency modulation inside a musical note. We present a MATLAB-based toolbox for detailed analysis of recorded vibrato notes. The harmonic structure of a music note is identified from its spectrographic analysis. Then we separate the harmonic structure into sonic partials using band-passed filters. The sound analysis algorithms, which provide the signal features of the vibrato note, are then performed over these separated sonic partials. In our implementation, each individual sonic partial is modeled as a quasi-monochromatic component, which is a sinusoidal signal with narrow-band amplitude modulation and frequency modulation. Based on this signal modeling technique, the modulating components are extracted from a separated sonic partial using modulation detection algorithms including short-time analysis and Hilbert transform. This toolbox provides comprehensive visualization tools to allow the users to interact and experiment with these signal features. We also implemented an auralization module that allows the users to experiment with the signal parameters and synthesize artificial vibrato sound. The analysis and visualization functionalities are all integrated in a compact graphical user interface that allows the users to intuitively implement complex sound analysis functionalities with simple analysis procedures.

Session 1pPA

Physical Acoustics, Animal Bioacoustics, and Signal Processing in Acoustics: Nonlinear Sound and Ultrasound Field Reconstruction and Related Applications II

Yong-Joe Kim, Cochair
Texas A&M Univ., 3123 TAMU, College Station, TX 77843

Je-Heon Han, Cochair
Mech. Eng., Texas A&M Univ., 3123 TAMU, College Station, TX 77840

Invited Papers

1:00

1pPA1. Nonlinear and transient acoustic holography for characterization of medical ultrasound sources and their fields. Oleg A. Sapozhnikov (Dept. of Acoust., Phys. Faculty, Moscow St. Univ., and Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., Univ. of Washington, Leninskie Gory, Moscow 119991, Russian Federation, oleg@acs366.phys.msu.ru), Wayne Kreider (Ctr. for Industrial and Medical Ultrasound, Univ. of Washington, Seattle, WA), and Sergey A. Tsysar (Dept. of Acoust., Phys. Faculty, Moscow St. Univ., Moscow, Russian Federation)

Holography is based on the possibility of reproducing a 3D wave field from a 2D distribution of the wave amplitude and phase measured along some surface transverse to the wave propagation. Such a measured distribution thus can be considered as a hologram. It provides a boundary condition for the wave equation, and as such is an important characteristic of any ultrasound source. In our previous work we have implemented various holographic approaches for medical ultrasound sources, including transient and nonlinear versions. Here we illustrate the approach with several experimental examples. Transient holography was performed to characterize the surface vibration of a circular, single-element, flat diagnostic probe with a 2.5 cm diameter and a 1 MHz resonance frequency after excitation by a submicrosecond pulse. Nonlinear holography was applied to a single-element focused source with a diameter and focal length of 100 mm. Forward- and backward-propagation algorithms were based on the Westervelt wave equation. A single-element focused source was excited at a sufficiently high power level so that harmonics were developed during nonlinear propagation in water; the nonlinear hologram was recorded as a set of 2D distributions of magnitude and phase for several harmonics. [Work supported by RFBR and NIH EB007643.]

1:20

1pPA2. Crack detection in long rod by impact wave modulation method. Alexander Sutin (Stevens Inst. of Technol., 711 Hudson St., Hoboken, NJ 07030, asutin@stevens.edu), Richard Haskins, and James Evans (Engineer Res. and Development Ctr., U.S. Army Corp of Engineers, Vicksburg, MS)

Engineering Research and Development Center (ERDC) in Vicksburg, MS, is interested in methods of microcracks detection in extended length and conducted test of applications of nonlinear wave modulation spectroscopy (NWMS). This is one of the simplest methods of nonlinear acoustic NDE. It is based on measurements of the modulation of a high frequency wave by a low frequency vibration. NWMS can detect the crack presence but cannot localize cracks. We present modification of NMWS method based on the modulation of by ultrasound by short pulse produced by impact. This method allows crack and damage localization using time delay of the impact produced pulse and modulated part of high frequency wave. The feasibility test was conducted for 60 ft long steel trunnion rod with an imitated crack. The continuous wave ultrasound with the frequency about 50 kHz was modulated by the longitudinal impulse produced by the hammer impact. Time delay between the impact and the received modulated ultrasonic wave allowed finding of the distance to the crack. Propagation speed of the modulated wave was lower than the speed of the hammer impulse as it is followed from the theory describing frequency dispersion of the waves in rods. [The project is funded by the Navigation Research Program at ERDC.]

1:40

1pPA3. Transient nonlinear acoustical holography. Yun Jing (Mech. Eng., North Carolina State Univ., 911 Oval Dr., EB III, Campus Box 7910, Raleigh, NC 27695, yjing2@ncsu.edu) and Jonathan Cannata (HistoSonics Inc., Ann Arbor, MI)

This paper presents our recent work on transient nonlinear acoustical holography. A higher order stepping algorithm is first introduced, which is shown to be significantly more accurate and efficient than the original one (Evaluation of a wave-vector-frequency-domain method for nonlinear wave propagation, Jing *et al*., J. Acoust. Soc. Am. **129**, 32) through systematic numerical study. Underwater experimental results from a highly focused transducer will be presented here to show the validity of the model. Both linear and nonlinear, forward and backward projection of the acoustic field are conducted. While linear acoustical holography is shown to produce erroneous results, good agreement is found between our nonlinear model and the experiment.

2:00

1pPA4. Possibilities of tomography system prototype using third-order acoustic nonlinear effects. Valentin A. Burov, Roman V. Kryukov, Andrey A. Shmelev, Dmitry I. Zotov, and Olga D. Rumyantseva (Dept. of Acoust., Moscow State Univ., Faculty of Phys., Leninskie Gory, GSP-1, Moscow, Russia, Moscow 119991, Russian Federation, blackrainbow13@mail.ru)

Third-order nonlinear acoustical tomography is very important for medical diagnostics, because it will provide information on the new and unexplored quantity—third-order nonlinear acoustical parameter. A prototype of the tomography system for reconstructing the distributions of the acoustic nonlinear parameters is developed in our group on the basis of effect of nonlinear noncollinear interaction of three primary waves. Application of coded primary signals with further correlation processing of a detected signal at combination frequencies makes it possible to reconstruct the complete image of an object using three transmitters and one receiver. Third-order effects are born from two competing processes—the pure third order interaction (informational for diagnostic purposes) and the twofold interaction of the second order (interfering). To explore the application boundaries of the third-order nonlinear acoustical tomography, these two competing processes are considered in details. Two cases are compared. In the first case, the interaction takes place between two broadband coded signals and one monochromatic signal. In the second case, all three primary signals are broadband coded; then, amplitude of the interfering part of the signal is smaller. Moreover, in this case, the possibility of reconstructing the spatial distribution of the second and third-order nonlinear acoustical parameters arises due to the reciprocity principle generalized on the nonlinear scattering processes.

Contributed Papers

2:20

1pPA5. Second-harmonic generation in shear wave beams with different polarizations. Kyle S. Spratt, Yurii A. Ilinskii, Evgenia A. Zabolotskaya, and Mark F. Hamilton (Appl. Res. Labs., The Univ. of Texas at Austin, 4307 Caswell Ave. Apt. E, Austin, TX 78751, sprattkyle@gmail.com)

A parabolic equation describing the propagation of collimated shear wave beams in isotropic elastic solids was derived by Zabolotskaya [Sov. Phys. Acoust. **32**, 296–299 (1986)], and was seen to contain both cubic and quadratic nonlinear terms at leading order. While second-order nonlinear effects vanish for the quasi-planar case of linearly-polarized shear wave beams, the importance of quadratic nonlinearity for more complicated polarizations is not yet well understood. The current work investigates the significance of quadratic nonlinearity by considering second-harmonic generation in shear wave beams generated by a certain class of source polarizations that includes such cases as radial and torsional polarization, among others. Corresponding to such beams with Gaussian amplitude shading, analytic solutions are derived for the propagated beam at the source frequency and the second harmonic. Diffraction characteristics are discussed, and special attention is paid to the relationship between the source polarization of the beam and the polarization of the subsequently generated second harmonic. Finally, suggestions are made for possible experiments that could be performed in tissue phantoms, exploiting the theoretical results of this work. [Work supported by the ARL:UT McKinney Fellowship in Acoustics.]

2:35

1pPA6. Average radiation force at high intensity: Measured data. Nick V. Solokhin (Ultrasonic S-Lab, 3628 Clayton Rd. # 102, Concord, CA 94521, solokhin@comcast.net)

Measurements were done in water at room conditions with HIFU transducer (at frequencies 3.1 and 4.3 MHz, max. pressure amplitude was 6—8 MPa). Average radiation force (ARF) was measured with flat reflecting target at normal incidence. The target was moved along the acoustic axis of the transducer: distance varied from 0.7 F to 1.4 F (F is focal distance). Measured ARF was growing with the distance (~25%) and it got max at distance 1.2 F. This effect meets with growing of nonlinear distortions at growing of intensity and length of passed way. It was measured dependence of ARF upon the angle of incidence and with same target. Measurements were done at incident angles 0, 30, and 60 degree. ARF changed from max value (at 0 angle) and reduced to 0.5 max value at 60 degree.

MONDAY AFTERNOON, 2 DECEMBER 2013 CONTINENTAL 9, 1:30 P.M. TO 3:00 P.M.

Session 1pSPa

Signal Processing in Acoustics: Acoustical Imaging and Sound Field Reconstruction

David Chambers, Chair
Lawrence Livermore National Lab., PO Box 808, L-154, Livermore, CA 94551

Contributed Papers

1:30

1pSPa1. Dynamic acoustical imaging systems with reconfigurable transceiver arrays and probing waveforms. Michael Lee and Hua Lee (Elec. and Comput. Eng., Univ. of California, 3121 Frank Hall, Santa Barbara, CA 93106, hualee@ece.ucsb.edu)

The data-acquisition format of traditional acoustical imaging systems has been operating with structured transceiver arrays and predetermined illumination waveforms. The fixed physical array configurations have been largely linear or planar, for computation simplicity. The conventional systems started with the coherent mode with narrow-band illumination. The coherent illumination waveforms were then replaced by wideband signals to operate in the pulse-echo format for the improvement of range resolution. This paper presents the functionalities of reconfigurable acoustical imaging systems with FMCW probing waveforms, and the equivalence to the conventional modalities. The reconfigurable configuration is applied to both the transceiver aperture arrays and the probing signals for the enhancement of the resolving capability. The reconfigurable array allows us to actively optimize the aperture coverage for superior resolution. To achieve reconfigurable illumination waveforms, programmable stepped-frequency FMCW signaling modality is employed. The presentation of this paper includes the algorithm structure, range estimation, and frequency editing, and of special interest and importance, the detection and imaging of time-varying targets.

1:45

1pSPa2. Reconstruction of arbitrary sound fields with a rigid-sphere microphone array. Efren Fernandez-Grande (Acoust. Technol., DTU, Tech. Univ. of Denmark, Ørsteds Plads, B. 352, DTU, Kgs. Lyngby DK-2800, Denmark, efg@elektro.dtu.dk)

Over the last few years, several studies have examined the potential of using rigid-sphere microphone arrays for reconstructing sound fields with near-field acoustic holography (NAH). The existing methods provide a reconstruction of the sound field based on a spherical harmonic expansion. However, because of the basis functions used, the reconstruction can only be performed in spherical surfaces concentric to the array and inside the source-free region, which imposes a severe limitation on the applicability of the technique due to geometrical constrains. In this paper, a method based on an equivalent source model is proposed, where a combination of point sources is used to describe the incident sound field on the array. This method makes it possible to reconstruct the entire sound field at any point of the source-free domain without being restricted to a spherical surface. Additionally, this approach adds versatility (the reconstruction can be based on the microphones that are closer to the source, for a better conditioning of the problem, or also use non-uniform sampling). The method is presented, examined numerically and experimentally, and compared to the existing methods based on a spherical harmonic expansion.

2:00

1pSPa3. Three-dimensional reconstruction of sound fields based on the acousto-optic effect. Efren Fernandez-Grande (Acoust. Technol., DTU, Tech. Univ. of Denmark, Ørsteds Plads, B. 352, DTU, Kgs. Lyngby DK-2800, Denmark, efg@elektro.dtu.dk) and Antoni Torras-Rosell (DFM, Danish National Metrol.Inst., Kgs. Lyngby, Denmark)

The acousto-optic effect can be used to measure the pressure fluctuations in air created by acoustic disturbances (the propagation of light is affected by changes in the medium due to the presence of sound waves). This makes it possible to measure an arbitrary sound field using acousto-optic tomography via scanning the field with a laser Doppler vibrometer. Consequently, the spatial characteristics of the sound field are captured in the measurement, implicitly bearing the potential for a full holographic reconstruction in a three-dimensional space. Recent studies have examined the reconstruction of sound pressure fields from acousto-optic measurements in the audible frequency range, based on Fourier transforms and elementary wave expansion methods. The present study examines the complete reconstruction of the sound field from acousto-optic measurements, recovering all acoustic quantities, and compares the results to the ones obtained from conventional microphone array measurements.

2:15

1pSPa4. Put your sound where it belongs: Numerical optimization of sound systems. Stefan Feistel (Ahnert Feistel Media Group, Berlin, Germany), Bruce C. Olson (Ahnert Feistel Media Group, Brooklyn Park, Minnesota), and Ana M. Jaramillo (Ahnert Feistel Media Group, 3711 Lake Dr., Robbinsdale, Minnesota 55422, anaja@vt.edu)

A new technology based on FIR filters in combination with room acoustic modeling allows optimizing steerable columns and line arrays to each specific venue in a matter of seconds. Both maximum SPL and maximum sound field uniformity can be prioritized to obtain an ideal sound distribution without losing sound pressure in the wrong directions thus avoiding unwanted reflections. Areas can also be excluded to minimize sound (i.e., stage area). Real-life test results support the theory.

2:30

1pSPa5. Multipole, spherical harmonics and integral equation for sound field reproduction. Jung-Woo Choi (Mech. Eng., KAIST, YuseongGu GuseongDong 373-1, Daejeon 373-1, South Korea, khepera@kaist.ac.kr)

The quality of sound field reproduction depends on the way we use to represent a sound field. For example, the spherical harmonics expansion being used for the higher-order-Ambisonics attempts to represent a desired sound field as a sum of many spherical harmonics, and the method incorporating integral equations, e.g., wave field synthesis, identifies the sound field as a superposition of single or double layer potentials distributed on a boundary surface. In contrast, the multipole expansion converts the desired sound field into equivalent multipole distributions. In this work, we investigate the fundamental differences in these three representations when they are applied for the reproduction of sound fields. In particular, their advantages and disadvantages in representing the directivity of a virtual sound source, translation of a desired sound field, and their benefit in deriving time domain formula for the real-time application will be discussed.

2:45

1pSPa6. The use of interpolated time-domain equivalent source method for reconstruction of semi-free transient sound field. Siwei Pan and Weikang Jiang (State Key Lab. of Mech. Syst. and Vib., Shanghai Jiao Tong Univ., 800 Dongchuan Rd., Shanghai 200240, China, swpan@sjtu.edu.cn)

A semi-free transient sound field is reconstructed by extending the interpolated time-domain equivalent source method (ITDESM) in the free field to the semi-free field. In this approach, the time-domain equivalent sources are placed not only near the actual sound sources, but also in the vicinity of their mirrored sources with respect to the reflecting plane surface. Suppose that the number of equivalent sources distributed around the mirrored sources (virtual equivalent sources) is the same as that used near the actual sources (actual equivalent sources), with their locations symmetrical about the reflecting surface. The reflecting surface considered here can be perfectly rigid as well as impedance-effected. Furthermore, by reformulating the strengths of the virtual equivalent sources at each time instant with those of the corresponding actual equivalent sources, the computation load of solving the equivalent source strengths can be reduced by 50%. Numerical examples of reconstructing the semi-free transient sound field radiated from three monopoles under different reflection conditions demonstrate the feasibility of the proposed method.

Session 1pSPb

Signal Processing in Acoustics: Filtering, Estimation, and Modeling in Acoustical Signal Processing

Edmund J. Sullivan, Chair
Prometheus Inc., 46 Lawton Rook Lane, Portsmouth, RI 02871

Contributed Papers

3:15

1pSPb1. A new look at the matched filter. Edmund J. Sullivan and James G. Kelly (Prometheus Inc., 46 Lawton Rook Lane, Portsmouth, RI 02871, ed@prometheus-us.com)

The matched filter is well known as the optimal linear detector. It is generally used in threshold tests, which depend only on its maximum value. However, it is usually used as an estimator, an example being the case of range determination in the active sonar problem. But even in this case, only its peak value is used. Here, we look at what we refer to as the "complete" matched filter. By examining the full output of the matched filter we show that it is closely related to the deconvolution problem, which is not a detector but an estimator. Further, we show that for the case of a linear chirp (LFM) signal, the full matched filter and the deconvolution problem are essentially identical, the difference being in the power spectrum of the signal. The more the signal power spectrum deviates from whiteness, the greater the difference between the two, even if the time-bandwidth product of the signal remains the same. Moreover, we show that the LFM signal is an optimal signal for the deconvolution problem since it minimizes the variance of the estimate.

3:30

1pSPb2. The spectral profile estimation algorithm: a non-linear, non-*a priori* noise normalization algorithm. Jeffrey A. Ballard (Appl. Res. Labs., The Univ. of Texas, P.O. Box 8029, Austin, TX 78713-8029, ballard@arlut.utexas.edu)

Time-frequency analysis of acoustic signals often involves a background noise spectral estimation step to estimate the signal-to-noise ratio of the signals of interest. This step is typically accomplished with the application of a split mean normalizer (SPN) [Struzinski and Lowe, J. Acoust. Soc. Am. **76** 1738–1742 (1984)]. These normalizers work well against tones and can be tuned with a priori information to normalize wider energy signals but can still suffer performance loss in these cases. For signal dense spectra that contain both tonal and broadband-like energy the SPN has difficulty separating noise from signal, and the signal energy unduly influences the background estimate. This work introduces a new normalization scheme called the Spectral Profile Estimation (SPE) algorithm, which operates with no a priori information to estimate the background noise of these signal dense spectra. The SPE algorithm assumes that the background noise has an f^{-m}, $m>0$, shape, and that signals are superimposed on the noise and are associated with spectra maxima. The SPE algorithm then finds and connects local minima to estimate the background. The SPE algorithm is first explained and then applied to experimental data. Finally, SPE performance is compared to the performance of SPN.

3:45

1pSPb3. Artificial reverberation using multi-portacoustic elements. Warren L. Koontz (Rochester Inst. of Technol., 159 Coco Palm Dr., Venice, Florida 34292, profwub@gmail.com)

Multi-port acoustic elements, including simple two-port elements, can be used both to model acoustic systems and to create acoustic signal processing structures. This paper focuses on the latter, specifically on networks of multi-port acoustic elements that create an artificial reverberation effect. We will introduce some basic two-port and multi-port building blocks, each characterized by a scattering matrix and demonstrate frequency domain and time domain analyses of networks of these elements. We will then propose and investigate networks that create artificial reverberation. We will implement these structures using MATLAB and evaluate and compare them with some existing approaches including the Schroeder reverberator and feedback delay networks. Comparisons will be based on the computed impulse response as well as sound files.

4:00

1pSPb4. An investigation of the subjective quality of non-linear loudspeaker models employing Volterra series approximations. Ian Richter (Peabody Inst. of the Johns Hopkins Univ., 606 St. Paul St, Box #14, Baltimore, MD 21202, ian.richter@gmail.com)

This investigation is an extension of the work of Angelo Farina on the use of Volterra-series approximations to model acoustic systems. Farina's method uses a logarithmically swept sine chirp to extract the transfer function of a system, including its harmonic nonlinearities. I have used this approach to construct computer models of several loudspeakers. I then evaluated my models using a blind listening test to compare their outputs against recordings of the actual loudspeakers. The results of the blind test suggest that, while this algorithm is not sufficient to convincingly model the loudspeaker transfer function, it might be useful as part of a multi-stage algorithm.

4:15

1pSPb5. Online sound restoration system for digital library applications. Andrzej Czyzewski, Janusz Cichowski, Adam Kupryjanow, and Bozena Kostek (Gdansk Univ. of Technol., Narutowicza 11/12, Gdansk 80-233, Poland, ac@pg.gda.pl)

Audio signal processing algorithms were introduced to the new online non-commercial service for audio restoration intended to enhance the content of digitized audio repositories. Missing or distorted audio samples are predicted using neural networks and a specific implementation of the Janssen interpolation method based on the autoregressive model (AR) combined with the iterative restoring of missing signal samples. Since the distortion prediction and compensations algorithms are computationally complex, an implementation which uses parallel computing has been proposed. Many archival recordings are at the same time clipped and affected by wideband noise. To restore those recordings, the algorithm based on the concatenation of signal clipping reduction and spectral expansion was proposed. The clipping reduction algorithm uses an intelligent interpolation to replace distorted samples with the predicted ones based on learning algorithms. Next, spectral expansion is performed in order to reduce the overall level of noise. The online service has been extended with some copyright protection mechanisms. Immunity of watermarks to the sound restoration is discussed with regards to low-level music feature vectors embedded as watermarks. Then, algorithmic issues pertaining watermarking techniques are briefly recalled. The architecture of the designed system together with the employed workflow for embedding and extracting the watermark are described. The implementation phase is presented and the experimental results are reported.

Session 1pUW

Underwater Acoustics and Acoustical Oceanography: Deep Water Acoustics II

John Colosi, Cochair
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Invited Papers

2:00

1pUW1. A comparison of measured and predicted broadband acoustic arrivals in Fram Strait. Hanne Sagen (Nansen Environ. and Remote Sensing Ctr., Bergen, Norway), Peter F. Worcester (Scripps Inst. of Oceanogr., Univ. of California, San Diego, 9500 Gilman Dr., 0225, La Jolla, CA 92093-0225, pworcester@ucsd.edu), Stein Sandven (Nansen Environ. and Remote Sensing Ctr., Bergen, Norway), Agnieszka Beszczynska-Moeller (Inst. of Oceanol. PAS, Sopot, Poland), Matthew A. Dzieciuch (Scripps Inst. of Oceanogr., Univ. of California, San Diego, La Jolla, CA), Florian Geyer (Nansen Environ. and Remote Sensing Ctr., Bergen, Norway), Brian D. Dushaw (Appl. Phys. Lab., Univ. of Washington, Seattle, WA), and Mohamed Babiker (Nansen Environ. and Remote Sensing Ctr., Bergen, Norway)

Fram Strait is the only deep-water connection between the Arctic and the world oceans. On the eastern side, the northbound West Spitsbergen Current transports warm Atlantic water into the Arctic, while on the western side the southbound East Greenland Current transports sea ice and polar water from the Arctic to the Nordic Seas and Atlantic Ocean. Significant recirculation and intense small-scale mesoscale variability in the center of the Strait make it difficult to accurately measure ocean transports through the Strait. An acoustic system for tomography, glider navigation, and passive listening was installed in the central, deep-water part of the Strait during 2010–2012. The integral measurements of temperature provided by tomography and the spatial resolution of the glider data are complementary to the data from the long-term array of oceanographic moorings at 780 50'N. The oceanographic conditions and highly variable sea ice in Fram Strait provide an acoustic environment that differs from both the high Arctic and the temperate oceans and that results in complex acoustic propagation. Improved understanding of the measured acoustic arrivals through comparison with predictions based on available environmental data is important for development of tomographic inversion and assimilation techniques, for glider navigation, and for acoustic communications.

2:20

1pUW2. Observations at Ascension Island of T-phases from earthquakes in the Fiji-Tonga region. Mark K. Prior, Mario Zampolli (PTS, Comprehensive Nuclear-Test-Ban Treaty Organisation, PO Box 1200, Vienna 1400, Austria, mark.prior@ctbto.org), and Kevin Heaney (OASIS, Lexington, MA)

Ascension Island in the Atlantic Ocean is the site of one of the hydrophone stations that make up part of the hydroacoustic network operated by the Comprehensive Nuclear-Test-Ban Treaty Organization. Hydrophones are deployed in two groups of three, known as triads; one to the north of the island and one to the south. Correlation processing across hydrophones allows signal azimuths to be determined and both triads show large numbers of signals arriving from the south-west. These signals are generated by earthquakes in the region between Fiji and Tonga in the Pacific Ocean and travel through Drake Passage between Antarctica and South America. Signal azimuths are studied and it is shown that some signals originate from earthquakes to which two-dimensional propagation modeling would suggest there is no direct path. The mechanisms by which sound from these "blocked" regions might reach Ascension Island are discussed.

2:40

1pUW3. Observation and modeling of three-dimensional basin scale acoustics. Kevin D. Heaney, Richard L. Campbell (OASIS Inc., 11006 Clara Barton Dr., Fairfax Station, VA 22039, oceansound04@yahoo.com), Mark Prior, and Mario Zampolli (CTBTO, Vienna, Austria)

In this paper, an overview of some of the basin-scale recordings of the International Monitoring System of the Comprehensive Test Ban Treaty will be presenting, including observations of distant earthquakes, under-sea volcanoes and cracking ice-sheets. Long range experiments were conducted in 1960 when nearly antipodal receptions were made at Bermuda from SuS charges deployed off the coast of Australia. Three-dimensional propagation effects was an important part of the propagation. For propagation ranges of thousands of

kilometers, interaction with islands, ridges and seamounts is expected to influence propagation. Previous modeling approaches have been either a fully 3D ray approach, a hybrid adiabatic mode-ray approach and a hybrid adiabatic mode—PE approach. In this paper, we present the a global scale three-dimensional split-step Pade Parabolic Equation. Model results are compared with several direct observations of arrivals that are clearly in the 2D propagation shadow. The impact of 3D propagation on the coverage maps of the CTBTO are significant, relative to current 2D predictions.

Contributed Papers

3:00

1pUW4. New formulae for horizontal coherence from path integral theory. John Colosi (Naval Postgrad. School, 833 Dyer Rd., Monterey, CA 93943, jacolosi@nps.edu)

Previously published path integral results for the horizontal coherence length utilized an empirical relation for the phase structure function density that scaled as lag to the three-halves power. Here, a Taylor series expansion is carried out such that the phase structure function density scales instead as the second power of lag, consistent with other path integral coherence scales such as depth and time. The resulting integral equations are solved analytically. The new result shows the expected one over square-root range and one over frequency scalings, and it demonstrates more clearly how transverse coherence is sensitive to the space-time scales of ocean sound-speed perturbations.

3:15

1pUW5. An alternative method for the estimation of underwater acoustic signal coherence. Matthew Dzieciuch and Peter Worcester (SIO/UCSD, 9500 Gilman Dr., IGPP-0225, La Jolla, CA 92093-0225, mad@ucsd.edu)

The estimator-correlator is the optimal method for measuring the travel-time of a scattered underwater tomographic acoustic signal. The method increases the signal-to-noise ratio at some cost in resolution when the signal and noise covariances are known. The noise covariance is easily measured when the signal is not present. The signal covariance may not be known but can be estimated as well. The procedure is to parameterize the signal covariance and find the maximum output signal-to-noise ratio of the estimator-correlator while varying the signal covariance parameters. Following Flatte *et al.*, the path integral model of signal covariance is $R = \exp\{ -(\Delta t/T_c)^2 + (\Delta f/B_c)^2 + (\Delta z/D_c)^2\}$. Thus, a simple search over coherence time, T_c, coherence bandwidth, B_c, and coherence depth, D_c, produces estimates of those parameters. Application of this technique with data from three recent tomographic experiments will demonstrate its efficacy. Time coherence was measured during the 2009 Philippine Sea Experiment at 250 Hz and 190 km range. Vertical coherence was measured during the 2010 Philippine Sea experiment at 250 Hz and at ranges from 120 to 450 km. Horizontal coherence was measured in the North Pacific at 75 Hz and 3500 km range.

3:30

1pUW6. The mode view of long-range propagation through oceanic internal waves. Frank S. Henyey (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, frank@apl.uw.edu)

For long-range acoustic propagation in the ocean, calculation of internal wave effects by ray tracing is far from accurate. Full wave methods must be used. An unperturbed normal mode simulation has been developed for this case. Simulations have been carried out with a single-mode starting field, with a sound speed profile measured in the 2010 Philippine Sea experiment, and internal waves consistent with measurements at the same region and same time. Modes were chosen that turn around half way between the sound axis and the surface. The frequency is 100 Hz, appropriate for long-range propagation experiments. Ensemble averaged mode intensities are consistent with transport theory results, but the individual mode intensities as a function of range are very inconsistent with a random walk ("diffusion") in mode number space.

3:45

1pUW7. Deep fades in intensity: Exploration of measurement-Monte Carlo parabolic equation mismatch in the Philippine Sea. Andrew W. White (Earth and Space Sci., Univ. of Washington, 433 31st Ave. E, Seattle, WA 98112, andrew8@apl.washington.edu), Rex K. Andrew, James A. Mercer (Appl. Phys. Lab., Univ. of Washington, Seattle, WA), Peter F. Worcester, Matthew A. Dzieciuch (Scripps Inst. of Oceanogr., Univ. of California at San Diego, La Jolla, CA), John A. Colosi (Oceanography, Naval Postgrad. School, Monterey, CA), Lora J. Van Uffelen, and Bruce M. Howe (School of Ocean and Earth Sci. and Technol., Univ. of Hawaii at Manoa, Honolulu, HI)

The oceanography of the Philippine Sea is partially characterized by energetic mesoscale and strong locally generated internal tides. Despite the simplification of range-independence and the exclusion of internal tides from Monte Carlo parabolic equation (MCPE) simulations, predictions of scintillation index, variance of log-intensity, and the distribution of intensity for acoustic paths with upper-turning-points (UTP) below the extreme upper ocean generally agree with measurements made during an experiment in 2009. These measures of the fluctuations did not appear to be strongly influenced by the number of UTPs in the path, though a compensating effect due to differences in UTP position cannot be ruled out. Enhanced variability in the form of deep fades is observed for paths turning in the extreme upper ocean; this enhanced variability is not predicted by the MCPE model employed. Seaglider-based observations of mixed-layer depth (from 2010 to 2011) and moored measurements of internal-tide-related sound-speed perturbations are presented. A plane-wave internal-tide model and results from acoustic mode propagation through range-independent profiles measured *in situ* are compared with the observed character of the intensity fades.

Payment of separate registration fee required. See page XXX

MONDAY AFTERNOON, 2 DECEMBER 2013 CONTINENTAL 7/8, 7:00 P.M. TO 9:00 P.M.

Session 1eID

Interdisciplinary: Tutorial Lecture on Time Frequency Analysis: Theory and Applications

R. Lee Culver, Chair
Pennsylvania State University, State College, PA 16804

Chair's Introduction—7:00

Invited Paper

7:05

leID1. Time-frequency analysis: Theory and applications. Leon Cohen (Dept. of Phys., City Univ. of New York, 695 Park Ave., New York, NY 10065-5024, leon.cohen@hunter.cuny.edu) and Patrick Loughlin (Dept. of Bioeng., Univ. of Pittsburgh, Pittsburgh, PA)

Time-varying spectra are one of the most primitive sensations we experience, since we are surrounded by light of changing color, by sounds of varying pitch, and by many other phenomena whose periodicities change. The development of the physical and mathematical ideas needed to explain and understand time-varying spectra has evolved into the field now called "time-frequency analysis." Among the many signals whose frequency content has been shown to vary in time are speech and other animal sounds, biomedical signals (e.g., heart sounds, heart rate, the electroencephalogram (EEG), the electromyogram (EMG), and others), music, radar and sonar signals, and machine vibrations, among others. In this tutorial, we give an overview of time-frequency analysis, with a focus on its applications. We describe how these methods impinge on and clarify issues in biomedical and biological signal analysis, wave propagation, random systems, non-linear systems, and other areas. Of particular interest is the application of time-frequency analysis to pulse propagation in dispersive media. We show that time-frequency considerations lead to new approximation methods for dynamic systems and wave propagation. We describe how to transform wave equations into phase-space, where the resulting equation is often more revealing than the original wave equation. We also discuss the applications to random systems and in particular to the propagation of noise fields.