Session 5aAA

Architectural Acoustics, Speech Communication and Committee on Standards: Speech in Architectural Spaces — Both Intelligibility and Privacy II

Kenneth P. Roy, Cochair
Armstrong World Industries, Innovation Center, 2500 Columbia Avenue, Lancaster, Pennsylvania 17604

Peter A. Mapp, Cochair
Mapp Associates, 5 Worthington Way, Colchester C03 4JZ United Kingdom

Chair's Introduction—8:45

Invited Papers

8:50

5aAA1. Design and acoustics in classrooms—a Philadelphia story. Kenneth P. Roy and Kenneth W. Good, Jr. (Armstrong Innovation Ctr., 2500 Columbia Ave., Lancaster, PA 17604)

The Lamberton School in Philadelphia was built in 1949 and may have met all of the acoustical design requirements of the day . . . but! An acoustic and A/V evaluation of a typical classroom was made both before and after an architectural intervention to address the ANSI S12.60 Classroom Standard. See the numbers and hear the difference, both for yourself, and through the opinions of the teacher who has experienced the change. What was done and the effects thereof will be discussed.

9:10

5aAA2. Recent experience with the voice intelligibility recommendations for fire alarm systems in large athletic spaces. Matthew Moore (Cavanaugh Tocci Assoc., 327 F Boston Post Rd., Sudbury, MA 01776, mmoore@cavtocci.com)

Beginning with the NFPA 72 National Fire Alarm Code, 1999 edition, there has been a recommendation for voice intelligibility with fire alarm systems. The code appendix references a Common Intelligibility Score (CIS) of 0.70 or greater. This paper discusses how the design and construction process for some recent sports facility projects has been affected, and some of the difficulties with meeting this score. Example projects and measurements will be used to illustrate the challenges in these unique environments.

9:30

5aAA3. Computer model studies to predict qualitative and quantitative measures of speech intelligibility in class rooms. Hyeong-seok Kim, Gary Siebein (School of Architecture, Univ. of Florida, P.O. Box 115702, Gainesville, FL 32611-5702), Brian Kreisman, and Carl Crandell (Univ. of Florida, Dauer Hall, Gainesville, FL 32611)

Computer model studies were used to predict qualitative and quantitative measures of speech intelligibility in classrooms under realistic conditions of background noise and reverberation. Fifteen different acoustical measurements related to speech intelligibility were made at multiple locations in three actual classrooms and in computer models of the classrooms. Speech intelligibility (MRT) tests were given to human subjects in each of the actual classrooms at five signal-to-noise ratios. Speech intelligibility tests were also prepared from aural simulations obtained by convolving anechoic speech tracts with impulse responses obtained in the computer models. Correlations ($R2$) between acoustical measures made in the full size classrooms and the computer models of the classrooms of 0.92 to 0.99 with standard errors of 0.033 to 7.311 were found. The scores on the speech intelligibility tests given in the actual rooms in the five noise conditions were closely duplicated in the equivalent tests conducted in a sound booth using the simulated speech signals obtained in the computer models. Both quantitative and qualitative measures of speech intelligibility in the actual rooms were accurately predicted in the computer models.

9:50

5aAA4. A verification of computer modeling predictions of speech intelligibility. Erica E. Bowden, Jonathan Rathsam, and Lily M. Wang (Architectural Eng. Program, Univ. of Nebraska Lincoln, 245 PKI, 1110 S. 67th St., Omaha, NE 68182-0681, ebowden@mail.unomaha.edu)

Computer modeling has become a popular tool in the field of architectural acoustics to predict the acoustical characteristics of a built environment. One characteristic commonly investigated in computer models is speech intelligibility. This project compares speech intelligibility measures of the speech transmission index (STI), the rapid speech transmission index (RaSTI), and the articulation loss of consonants (ALCons) predicted by computer models to measurements taken in existing spaces. Three spaces in the Omaha, NE, area of varying size and function are analyzed: a theater, church, and classroom. Research methodology and conclusions are presented. The results aid in understanding the reliability of 3D computer modeling in speech intelligibility predictions. [Work partially supported by the Univ. of Nebraska Center for Building Integration.]

10:10–10:25 Break

10:25

5aAA5. Some common error mechanisms in making STI intelligibility measurements. Peter A. Mapp (Peter Mapp Assoc., Copford, Colchester, Essex, UK)

STI and its derivatives (RaSTI and STIPa) have become the internationally accepted methods for acoustically measuring the potential intelligibility performance of both sound systems and rooms. Their use has unquestionably brought about a significant improvement in public address/voice alarm system quality and intelligibility. However, in practice, many of the measurements made on site may be unwittingly based on flawed techniques. The paper examines a number of common problems found to affect measurement accuracy. Significant differences between different measurement platforms and techniques are reported. Furthermore, different stimulus formats and even their recording and playback medium are also shown to potentially affect the final result. It is shown that neither STI nor STIPa, in their current formats, accurately predicts the intelligibility of sound systems with irregular frequency responses, particularly when these are operating in reverberant, high signal-to-noise environments, a common effect found in real-world voice alarm and emergency sound systems.

10:45

5aAA6. Prediction of speech intelligibility in rooms—a comparison of five methods. Murray Hodgson (School of Occ. & Env. Hygiene, Univ. of BC, 3rd Fl., 2206 East Mall, Vancouver, BC V6T 1Z3, Canada, hodgson@mech.ubc.ca)

In this paper, five methods for calculating speech intelligibility at listener positions in rooms, from measured impulse responses and speech and noise levels, are presented and compared. All methods involve combining a measure of room reverberation with the received signal-to-noise level differences in the occupied room. The first method involves measured octave-band C50s and signal-to-noise level differences to determine octave-band U50s. The remaining four methods comprise detailed (involving octave-band values) and simplified (involving single- and/or combined-frequency values) versions of two approaches. The first approach (second and third methods) involves measured background-noise levels and values of the Transmission Index measured for infinite signal-to-noise level difference. The second approach (fourth and fifth methods) involves 1000-Hz octave-band early-decay times and A-weighted signal-to-noise level differences. The five methods are briefly presented. Then, predictions by the methods for a number of classrooms are presented and compared, and the differences, the merits, and the disadvantages of the various methods are discussed.

11:05

5aAA7. Comparison of three methods for determining speech transmission index (STI). Robert C. Coffeen (School of Architecture and Urban Design, Marvin Hall, The Univ. of Kansas, Lawrence, KS 66045)

Speech transmission index (STI) is a commonly used descriptor whose goal is to quantify speech intelligibility. STI can be calculated from computer modeling of a particular architectural space, calculated from an impulse response of an existing space, and measured by introducing modulated noise in an existing space. This paper will present calculated and measured STI ratings for a particular and existing architectural space so that these three methods for determining STI can be compared.

11:25

5aAA8. A comparison of computational models for predicting speech intelligibility and speech privacy. Ralph T. Muehleisen (Civil and Architectural Eng., Illinois Inst. of Technol., Chicago, IL, 60616, muehleisen@iit.edu) and C. Walter BeamerIV (Univ. of Colorado, Boulder, CO 80309)

Speech intelligibility (SI) and speech privacy (SP) are very important considerations in the design of classrooms, auditoria, conference rooms, and offices. Predictions of SI and SP are usually done with simple analytic models using Sabine/Eyring theory or CAD programs such as CATT, EASE or ODEON. In this paper we will compare the predictions of speech intelligibility index (SII), speech transmission index (STI), and privacy index (PI) using analytic theories of Sabine/Eyring and a simplified acoustic radiosity theory, a simple computational model using acoustic radiosity and CAD prediction using CATT acoustics. Predictions will be compared with the results of a similar study by Bistafa and Bradley [J. Acoust. Soc. Am. **109**(4), 1474–1482 (2001)].

Contributed Paper

11:45

5aAA9. Acoustical retrofit in residential buildings. Ballard W. George (Environtech Consultants, 1367 Bobolink Circle, Sunnyvale, CA 94087, kingstaco@yahoo.com)

This paper deals with situations where for some reasons problems arise, and one has to apply noise control measures after or near the end of construction. Three cases are discussed. In two of these, recommendations for window acoustical performance were not implemented. In one case, window improvement was needed to achieve consistency with documentation given to the city and to meet design goals. The owner elected to use the option of improved windows combined with sound absorption in terms of a high-performing carpet. In another case, again window recommendations were not followed, and a curtain wall with Lexan lights also contributed to excessive exterior-to-interior noise levels up to approximately 10 dB. A spaced-out gypsum board inner wall layer arrangement, with ventilation openings, was developed by the architect. The proposed mitigation arrangement involved trade-offs regarding acoustical separation and privacy between units. The third case involved a new hotel, where kitchen-serving equipment, not shown on plans provided, created rumble that resulted in complaints regarding one guest unit. A menu of corrective measures was provided, which could be implemented in sequential fashion.

Session 5aAB

Animal Bioacoustics: Animal Bioacoustics General Topics

Ann E. Bowles, Chair
Hubbs Sea World Research Institute, 2595 Ingraham Street, San Diego, California 92109

Chair's Introduction—8:25

Contributed Papers

8:30

5aAB1. An investigation of hearing threshold shift in bats due to blast wave pressure. Larry Pater (ERDC/CERL, 2902 Farber Dr., Champaign, IL 61821, larry.l.pater@erdc.usace.army.mil), Elizabeth Brittan-Powell, Cynthia Moss, Robert Dooling (Univ. of Maryland, College Park, MD 20742), Alexandra Loubeau, and Victor Sparrow (The Penn State Univ., University Park, PA 16802)

A project is described to determine damage to chiropteran (bat) hearing by shock waves such as those emitted by weapons and explosions. Experimental and analytical means will be used to accurately define the stimulus, particularly as regards wave form and spectra in the frequency range 10–100 kHz in which bat hearing is sensitive. Initial results regarding the stimulus as well as hearing thresholds determined via the auditory brainstem response are presented. [Work supported by US Army Engineer Research and Development Center CERL.]

8:45

5aAB2. Acoustic adaptation hypothesis in macro- and micro environments: An analysis of frog calls. Yuan Yao (Dept. of Organistic Biol., Ecology and Evolution, UCLA, Los Angeles, CA 90095) and Ying Lin (UCLA, Los Angeles, CA 90095)

The acoustic adaptation hypothesis states that acoustic signals are structured so as to maximize their performance under the constraints of the environmental acoustics that characterize their native habitats. However, less attention has been paid to the channels in which acoustic communication takes place and the specific constraints that different channels put on acoustic signals. For frog calls, the channel characteristics depend on two factors: the macro environment—such as the habitat; and the micro environment—such as the sites of the sender and receiver. The data in our analysis came from recordings of 81 species of frogs in French Guyana. These species span a variety of macro environments, such as forest, coastal marsh, and savanna, and a variety of micro environments, such as on vegetation, on ground, and under litter. Two methods—spectrogram correlation and hidden markov models—are used to derive a similarity matrix between these calls. The results indicate that calls show a higher similarity within the same macro environment than between different macro environments. On the other hand, measurements of spectral and temporal features suggest that the micro environment also plays a role, while its effect is more salient for frequency components than for time components.

9:00

5aAB3. Statistical automatic species identification of microchiroptera from echolocation calls: Lessons learned from human automatic speech recognition. Mark D. Skowronski and John G. Harris (Computational NeuroEng. Lab, Elec. and Computer Eng., Univ. Florida, Gainesville, FL, markskow@cnel.ufl.edu)

Current automatic species identification of microchiroptera through echolocation search calls utilizes holistic features, such as call duration, bandwidth, and frequency extrema, which stem from expert knowledge of sonograms of individual calls. This approach parallels the early acoustic-phonetic paradigm of human automatic speech recognition (ASR), which relied heavily on expert knowledge to account for variations in canonical linguistic units of interest. Acoustic-phonetic ASR gave way to the statistical paradigm of ASR primarily because of the superior ability of machine learning to account for variations in the acoustic signal (noise, speaker characteristics) [Juang and Furui, Proc. IEEE **88** (8), 1142–1165 (2000)]. In the current work, machine learning methods from human ASR, hidden Markov models (HMM) and dynamic time warping (DTW), are considered for the problem of automatic species identification of microchiroptera from echolocation calls. The extensive history of ASR provides valuable lessons, which are highlighted and applied to species recognition from field-recorded acoustic data. Frame-based features of the acoustic signal (fundamental frequency, log energy, temporal derivatives) are incorporated into HMM and DTW classifiers, and experimental results indicate the superior approach of statistical automatic species identification compared to current techniques.

9:15

5aAB4. Automatic song-type classification and individual identification of the ortolan bunting (*Emberiza hortulana L*) bird vocalizations. Kuntoro Adi (Speech and Signal Processing Lab, Marquette Univ., Milwaukee, WI 53233), Tomasz S. Osiejuk (Adam Mickiewicz Univ. PL-61-485 Poznan, Poland), and Michael T. Johnson (Marquette Univ., Milwaukee, WI 53233)

This paper presents a method for automatic song-type classification and individual identification of the ortolan bunting (*emberiza hortulana L*). This method is based on hidden markov models (HMMs) commonly used in the signal processing and automatic speech recognition research communities. The features used for classification and identification include both fundamental frequency and spectral characteristics. Fundamental frequency features consist of center frequency $f0$, relative $f0$ to moving-average baseline, peak strength, and peak bandwidth. Spectral features are derived from frequency-weighted cepstral coefficients. Using these features one HMM is trained for each type of vocalization both for each individual bird and across the entire population. Preliminary results indicate accuracies of above 90% for both song-type classification and individual identification tasks.

9:30

5aAB5. Comparison and evaluation of animal vocalization enhancement techniques. Jidong Tao and Michael T. Johnson (Dept. of Elec. and Comput. Eng., Marquette Univ., Haggerty Hall, Milwaukee, WI 53201)

Animal vocalization recordings are often corrupted by wideband background noise and interfering signals; however, signal processing methods for enhancement of these waveforms has not received as much attention in the literature as has human speech enhancement. In order to improve vocalization intelligibility and quality, a variety of enhancement methods taken from the field of speech processing are investigated here. These techniques range from traditional approaches such as spectral subtraction to more advanced ones such as Ephraim Malah log-spectral estimation and

wavelet denoising. Results, measured by improvement in signal-to-noise ratio (SNR) and subjective perceptual tests, are given for several noise environments. Signal enhancement is demonstrated across a variety of species, including African elephant, Beluga whale, and ortolan bunting vocalizations. [Work supported by NSF.]

9:45

5aAB6. Classification of natural landmark with biosonar. Maosen Wang and Andreas Zell (RA Dept., WSI, Univ. of Tuebingen, Sand 1, Tuebingen, Germany)

Echolocating bats can make nocturnal flights in acoustically cluttered environments with the use of echolocation. Their marvelous ability to evaluate natural targets in complete darkness provides us an opportunity to learn target detection, classification, and identification with similar biomimetic platforms. In this work, natural landmark classification with a binaural system, a sequential sensing strategy, and a frequency after reconstruction algorithm are adopted to provide sequential acoustic images for target classification. Experimental results suggest that considerable improvements in classification accuracy can be achieved by the use of this sequential classification method.

10:00

5aAB7. Vocalizations of *Equus caballus*: Frequency analysis of horse whinnies. David G. Browning (Dept. of Phys., Univ. of Rhode Island, Kingston, RI 02881, decibeldb@aol.com) and Peter M. Scheifele (Univ. of Connecticut, Storrs, CT 06269)

There are six recognized horse sounds: scream, squeal, nicker, whinny, snort, and blow. Of these whinnies are the most interesting with a high enough source level for communication, perhaps being the horse equivalent of a bark, and a characteristic change in frequency content during the vocalization. This frequency variation is not observed in many farm animal vocalizations, cows, goats, and sheep, for example. An analysis is made of the variation in typical horse whinnies in order to identify any general characteristics as well as any specific variations that can be identified with specific behavior.

10:15

5aAB8. Diel periodicity of fish sound production in Charlotte Harbor, Florida. James Locascio (Univ. of South Florida College of Marine Sci., 140 Seventh Ave. South, St. Petersburg, FL 33701) and David Mann (Univ. of South Florida College of Marine Sci., St. Petersburg, FL 33701)

Diel periodicity in sound production of spawning aggregations of fishes was documented in Charlotte Harbor, Florida from 7 May 2003–10 June 2003. The Long Term Acoustic Recording System (LARS) recorded 10 s of sound every 10 min within the frequency range of 0–1250 Hz. Field data collected in this study revealed diel patterns in fish sound production in great detail. Autocorrelation results demonstrated pronounced diel periodicity in fish sound production with significant lags occurring at 24-h periods. Chorusing events dominated by sand seatrout, *Cynoscion arenarius*, lasted for several hours each night and were not highly variable in duration, maximum recorded SPL, or start and end times. Mean daily (24-h period) maximum sound-pressure level was 119 decibels, mean daily chorus start time was 17:26 h Eastern Standard Time, mean daily chorus end time was 02:10 h Eastern Standard Time, and the average nightly chorus duration lasted 8 h and 43 min.

FRIDAY MORNING, 19 NOVEMBER 2004 ROYAL PALM SALONS 1 & 2, 9:00 TO 11:55 A.M.

Session 5aNS

Noise: Soundscapes and Sound Quality

Brigitte Schulte-Fortkamp, Cochair
Institute of Technical Acoustics, Technical University Berlin, Einsteinufer 25, 10587 Berlin, Germany

Patricia Davies, Cochair
School of Mechanical Engineering, Purdue University, Ray W. Herrick Laboratories, 140 South Intramural Drive, West Lafayette, Indiana 47907-2031

Chair's Introduction—9:00

Invited Papers

9:05

5aNS1. Perception of noise from diesel-engine powered vehicles. Aaron Hastings (Ray W. Herrick Lab., School of Mech. Eng., Purdue Univ., 140 S. Intramural Dr., West Lafayette, IN 47907-2031), Patricia Davies, and Aimee M. Surprenant (Purdue Univ., West Lafayette, IN 47907)

A program was developed to simulate diesel engine sounds to have control over timing and amplitude variations in combustion events. The simulation includes this combustion-related noise, together with a noise floor and families of tones from auxilliary components. Additional shaping filters to control the spectral balance of the sounds are also included. Sound and vibration measurements from ten vehicles powered by diesel engines were taken and combustion event timing and amplitudes were extracted to determine what ranges and types of variations typically occurred. It was found that there were random, deterministic and individual cylinder variations present in one or more of the vehicles. From this analysis, the simulation program was used to generate 199 sounds for a listening test in which 40 subjects participated. Spectral balance, tonalness, and, fluctuations all affected the subjects' responses and, above certain levels, as they increased so did subjects' annoyance ratings. Based on the results of this test, modifications to the psychoacoustic annoyance model [Zwicker and Fastl, *Psychoacoustics: Facts and Models*, 2nd ed. (Springer, New York 1998)] are proposed. These modifications include the incorporation of a term that quantifies how the tonal character of the sound affects annoyance. [Work supported by Isuzu Motor Company.]

9:25

5aNS2. Characterizing short-time transient events in the information technology soundscape. Wade R. Bray (HEAD acoustics, Inc., 6964 Kensington Rd., Brighton, MI 48116)

Information technology (IT) products form an increasing part of modern life in residential, recreational, and work contexts. Although average levels and sound power are generally low and have been improving for years, many IT products and components produce transient sounds which draw attention and elicit customer complaints. Standardized characterization by average sound power, with reference to tonal and impulsive characteristics, may not adequately capture subjectively important time-varying behavior. Time/frequency, time/magnitude, impulsiveness analysis methods, and an adaptive relative pattern-recognition technique will be applied to the short-time transient noises produced by a variety of information technology devices.

9:45

5aNS3. Sound quality in environment: "Psychoacoustic mapping." Klaus Genuit (HEAD acoustics GmbH, Ebertstrasse 30a, 52134 Herzogenrath, Germany, klaus.genuit@head-acoustics.de)

The annoyance due to noise in the environment is often predicted by analysis software which calculates the A-weighted sound pressure level distribution depending on the sound sources and on the sound propagation caused by them. The actual subjective perceived noise annoyance of the environment does not only depend on the A-weighted sound pressure level, but also on the so-called sound quality along with other parameters like the aspects of a soundscape. The question regarding noise annoyance cannot be predicted by the A-weighted sound pressure level alone. The sound quality perceived by the human hearing depends on, among other things, loudness, roughness and sharpness. The previously known methods for the prediction of the spatial A-weighted sound pressure level distribution based on the propagation are not suitable to predict psycho-acoustic parameters in an adequate way. Especially the roughness caused by modulation or the sharpness generated by the contribution of high frequency sound energy at various distances cannot readily be predicted. In the following it will be shown what the requirements are and especially what the challenges are for making such a psychoacoustic mapping.

10:05–10:20 Break

10:20

5aNS4. Community reaction to noise from a replaced bridge span and associated approach. Paul R. Donavan (Illingworth & Rodkin, Inc., 505 Petaluma Blvd. South, Petaluma, CA 94952, pdonavan@illingworthrodkin.com) and Bruce Rymer (California Dept. of Transportation, Sacramento, CA 95814)

A newly constructed replacement of the westbound span of the Carquinez Bridge on Interstate I-80 in the San Francisco Bay Area was recently opened for service. Although the traffic volume and mix remained the same as it was before the project, there was a very strong reaction to the noise produced by the new construction from nearby residences. For these residences, the predicted noise levels were only expected to increase by less than 2 dB due to the bridge and its approaches being somewhat closer to the community. The complaints had two aspects: generally higher traffic noise levels and impulsive slaps from the expansion joints in the viaduct connecting the bridge deck to the at-grade roadway. Although clearly audible in the community, the joint slaps did not contribute to any of the common community traffic noise metrics. For noise mitigation, the viaduct surface was ground to reduce overall tire/pavement noise, however, there was concern that by reducing the masking effect of the pavement texture generated noise, the slaps from joints would become even more noticeable and objectionable. The before and after noise measurements and community response are presented and discussed in this presentation.

10:40

5aNS5. Explorative sound evaluation. Stephan Paul, Brigitte Schulte-Fortkamp (ITA/TU-Berlin, Einsteinufer 25 D-10587, Berlin, Germany), and Klaus Genuit (HEAD-acoustics, Herzogenrath, Germany)

Exposure group description of sound based on AISP (exploration of associated imaginations on sound perception) is a powerful tool in sound quality evaluation. Used in a public research project it was stated that AISP needs further adaptation for automotive use. The present study was intended as a first approach refining the method for industrial application regarding the evaluation of an appropriate test design. The study was carried out using a driving simulator and a car, both with changeable acoustic properties. Verbal evaluations by the driving test individuals and the sound were recorded. Later on an open interview with the driver was carried out based upon given commentaries. The verbal data were analyzed through a qualitative approach. By means of this data the car with changeable acoustic properties driven on road was identified as most appropriate for sound evaluation of car interior sounds. The verbal data analysis and the physical analysis of the sounds gave valuable hints for troubleshooting and target sound definition. As the study showed the strong dependency of sound evaluation on the test design a representative evaluation should be done in an adequate design (car on road). Furthermore, initial usable tools and guidelines for industrial application were developed.

11:00

5aNS6. Ambient noise in Johns Hopkins Hospital. Ilene J. Busch-Vishniac, James E. West, Colin Barnhill, and Tyrone Hunter (Dept. of Mech. Eng. and Dept. of Elec. and Comput. Eng., Johns Hopkins Univ., 3400 N. Charles St., Baltimore, MD 21218)

Although noise in hospitals is widely recognized as a problem, there has been little work to characterize hospital noise reported in the literature. From the literature that does exist, it is clear that noise in hospitals poses problems for patient safety, increases staff stress, and is a leading source of complaints by patients and visitors. It is also clear that few hospitals currently meet the recommended noise limits of the World Health Organization or the US EPA. In this paper we present the first round of results of a noise study at Johns Hopkins Hospital in Baltimore, MD. Some interesting characteristics of the noise include its high level, its unusually flat spectrum through the speech range, and its relative invariability with location.

11:20

5aNS7. Natural and urban soundscapes: The need for multi-disciplined research. Robert Kull (Parsons, 5800 Lake Wright Dr., Ste. 101, Norfolk, VA 23502), Brigitte Schulte-Fortkamp, Sebastian Rossberg (Tech. Univ. Berlin, Berlin Germany) and Tim Lavallee (P.E. LPES, Inc., Smithfield, VA 23430)

Characterizing soundscapes represents a broad variety of approaches (ecological, sociological, phenomenological) in a continuum from completely natural to highly urban environments. There is an urgent need for the use of innovative designs which integrate the different levels of current analyses (qualitative and quantitative; individual and aggregate levels). The criteria (beyond sound level) of a good soundscape (What is a sensitive soundscape? What are soundscape requirements for a resort area or natural quiet?), that should be protected are central to the origin of the soundscape idea. Other classical acoustics questions, such as the use of audibility intrusiveness under critical conditions and under condition of mixed sources and time pattern, should be asked differently to deepen insight and understanding. A first attempt was made to monitor acoustical changes in an Italian mountain resort: Classical indicators (L95, L5, Leq) were used to locate geographical areas of increases in intrusiveness of the soundscape and linked to the activity pattern of the area. Several projects were conducted to define and localize quiet areas for a nature conservation program. Descriptive and analytical techniques, such as participatory sound and listening walks, cognitive maps, acoustical spectrographic maps, and soundscapegraphy are new techniques to consider for multi-disciplined approaches in different environments.

Contributed Paper

11:40

5aNS8. Queuing for quiet—The natural soundscape microstructure from a visitor perspective. Richard D. Horonjeff (Richard D. Horonjeff Consultant in Acoust. and Noise Control, 81 Liberty Square Rd. #20-B, Boxborough, MA 01719) and Grant S. Anderson (Harris Miller Miller & Hanson Inc., Bultington, MA 01803)

The passage of PL100-91 The National Parks Overflights Act of 1987 and PL106-181 The National Parks Air Tour Management Act chartered the National Park Service (NPS) and Federal Aviation Administration (FAA) to restore natural quiet to park settings, especially those with lengthy occurrences of nonindigenous sounds. Dose–response studies sponsored by NPS and FAA showed that among several dependent variables tested, nonindigenous time audible was the best predictor of visitor-reported annoyance and interference with natural quiet. Other than for Grand Canyon NP, PL100-91 lacked specificity for depicting nonindigenous times audible. Since 1991, several field investigations have produced over 400 h of source identification logs maintained by audiometrically screened observers. At over 25 sites in 7 parks, the data acquisition protocol used a forced-choice, hierarchical menu structure with audible source state changes timed to the nearest second. The identical protocol across all studies provided the basis for examining the amount of time a visitor would have to wait (T) to experience a contiguous block of indigenous-only sounds of duration (D), given a random arrival time. The paper reports means and distributions in T for various values of D, observed data clusterings across site groups, and the effects of mitigating variables.

Session 5aSC

Speech Communication: Boundaries, Rhythm and Timing in Spoken Language (Poster Session)

Amalia Arvaniti, Chair
Linguistics Department, University of California, San Diego, 9500 Gilman Drive, La Jolla, California 92093-0108

Contributed Papers

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

5aSC1. Relating the performance and perception of phrasal boundaries. Jelena Krivokapic (Dept. of Linguist., USC, 3601 Watt Way, GFS 301, Los Angeles, CA 90089-1693, krivokap@usc.edu)

This study examines the correspondence between the production of prosodic structure and perceptual judgments regarding prosodic boundary strength. In the production component of the study, three speakers read sentences in which one specific juncture is manipulated (varying in syntax, position in sentence, and phrase length) to elicit phrasal boundaries of differing strengths. Lengthening and pause duration at the boundary were measured. The same sentences, in written form, were presented to a second different group of three speakers who provided estimates of the strength of the target boundary on a scale of eight degrees. The results of the production and the estimation portions of the experiment demonstrate significant correlations between the production boundary strength as reflected in durational properties at the juncture and the boundary strength as estimated in the judging task. These correlations are roughly linear. Further, in both the production and perception domains, a range of boundary strength is exhibited rather than a small discrete set of boundary types. We also examine whether speaker's own boundary strength estimates agree with their productions to a greater extent than estimates of other speakers. [Work supported by NIH.]

5aSC2. On the temporal scope of boundary effects in articulation. Dani Byrd, Jelena Krivokapic (Dept. of Linguist., USC, 3601 Watt Way, GFS 301, Los Angeles, CA 90089-1693, dbyrd@usc.edu), and Sungbok Lee (USC, Los Angeles, CA 90089)

Boundary-adjacent acoustic lengthening is well explored, and the articulatory bases for this lengthening are becoming better understood. However, the temporal scope of boundary effects has not been examined in the articulatory domain. The few acoustic studies examining the distribution of lengthening indicate that boundary effects extend from one to three syllables leftward from the boundary, and that effects diminish as distance from the boundary decreases (Cambier-Langeveld, 1997; Shattuck-Hufnagel and Turk, 1998; Turk, 1999; Berkovits 1993a,b, 1994). This diminishment is predicted by the pi-gesture model of prosodic influence (Byrd and Saltzman, 2003). We present an experiment testing the leftward and rightward scope of articulatory lengthening. One condition (CV#C1VC2VC3V) tested the scope of effects after the boundary, and another (C3VC2VC1V#CV) the scope preceding the boundary (where # indicates an intonational phrase boundary) as compared to a no-boundary control. Movement-tracking (EMA) data allowed the evaluation of constriction formation and release duration, acceleration, and spatial magnitude. Further, FDA (functional data analysis) was used to examine continuous temporal warping properties. Results of both analyses indicate an asymmetrical distribution of boundary effects, in that leftward effects are shorter in scope than rightward effects, though effects in both directions exist around a phrase boundary. [Work supported by NIH.]

5aSC3. Interacting effects of phrasal and syllable position on consonant production. Sungbok Lee (Dept. of Linguist., USC, and USC Viterbi School of Eng., 3601 Watt Way, GFS 301, Los Angeles, CA 90089-1693, sungbokl@usc.edu), Dani Byrd (USC, Los Angeles, CA 90089), Jason Adams (USC, Los Angeles, CA 90089), and Daylen Riggs (USC, Los Angeles, CA 90089)

The complexities of how prosodic structure, both at the phrasal and syllable levels, shapes speech production have begun to be illuminated through studies of articulatory behavior. Here, we pursue the goal of understanding prosodic signatures on articulation by examining the effects of phrasal and syllable position on the constriction formation and release of consonants. Articulatory kinematic data were collected for five subjects using electromagnetic articulography (EMA) to record target consonants (labial, labiodental, and tongue tip), located in (1) either syllable final or initial position and (2) either at a phrase edge or phrase-medially: #C, ##C, C#, C##. The duration, displacement, and time-to-peak velocity of constriction formation and release were determined for the target consonants based on kinematic landmarks in the articulator velocity profiles (zero crossings and extrema). ANOVA determines that syllable and phrasal position consistently affect the movement duration; however, effects on displacement were more variable. For the majority of subjects, the boundary-adjacent portion of the movement (release for a preboundary coda and constriction formation for a postboundary onset) is not differentially affected in terms of phrasal lengthening; both lengthen equivalently. However, two subjects show an interaction such that the codas are lengthened more than the onsets. [Work supported by NIH.]

5aSC4. Towards standard measures of articulatory timing. Leonardo Oliveira, Mariko Yanagawa, Louis Goldstein (Yale Univ. & Haskins Labs., 270 Crown St., New Haven, CT 06511), and Ioana Chitoran (Dartmouth College, Hanover, NH 03755)

Many studies have investigated the relative timing (or overlap or latency) between articulatory events to make inferences about coordination in speech production or about phonetic structure. A major problem is that these studies employ different measures of relative timing. One criterion for choosing a standard measure could be the variability exhibited across repetitions and talkers for a given set of gestures in a given phonological context. If a single measure appears to be the least variable across a variety of experiments, then it is a good index of the stable, phonetically relevant properties of coordination. In this study, 33 measures were compared using coefficient of variation. In four corpora of articulatory data on consonant clusters (EMMA and x-ray microbeam), three articulatory events were defined: gesture onset, target, and release. Based on these events, coefficients of variation for the 33 measures were calculated across repetitions and speakers under a number of different linguistic conditions, i.e., stress, syllable, rate, cluster type and language (English and Georgian). Despite the very different properties of the corpora, the least-variable measures were always very similar and involve normalizing the latency between the two gestures by the constriction formation time (target to onset). [Work supported by NIH.]

5aSC5. Phrase-final frication lengthening in a variety of English spoken in the Iron Range of Northern Minnesota. Matt Bauer (Dept. of Linguist., Georgetown Univ., 37th and O St., Washington, DC 20057)

This study examines acoustic and perceptual correlates of final-devoicing in a variety of English spoken throughout the Iron Range of Northern Minnesota. A pilot study of one speaker indicates devoicing in the dialect is present phrase-finally but not phrase-medially $x2(1) = 0.89$, $p < 0.05$. Measurements of voicing and frication in words like tens and tense reveal devoicing is not due to shortening of the duration of vocal fold vibration (which is what a devoicing process predicts). Instead, frication of phrase-final fricatives are lengthened compared to frication of phrase-medial fricatives $F(2,27) = 17.36$, $p < 0.05$, but vocal fold vibration phrase-finally is unaffected. Thus, the percept of devoicing results from a lengthening process that does not alter voicing duration. The present study will determine the extent to which final-lengthening is present in the speech of other speakers. The project is interesting because (a) Iron Range English was previously thought to exhibit devoicing, not final-lengthening [G. Underwood. Pub. Am. Dialect Society, 67 (1982)] and (b) results address how representations of sound are transferred into speech unfolding in time. Intuitions offered by articulatory phonology indicate timing of phrase-final frication lengthening in Iron Range English is not a reflex of a phonological rule, but instead of differentiated timing of independent gestures.

5aSC6. Final lowering: Fact, artifact or dialectal variation? Amalia Arvaniti (Dept. of Linguist., UCSD, 9500 Gilman Dr. #0108, La Jolla, CA 92093 amalia@ling.ucsd.edu)

The existence of final lowering has been disputed by Grabe (1998), who suggested that Liberman and Pierrehumbert's 1984 finding was due to declination, as there was an extra syllable, ''and,'' between the last two accents in their materials (e.g. ''RASPberries, MULberries and BRAMBLEberries''). Arvaniti and Godjevac (2003), however, replicated the results of L&P with and without the extra syllable (c.f. ''LIma beans, NAvy beans and SOY beans'' vs. ''LIma beans, GREEN beans and SOY beans''). Here, the materials used in A&G were elicited from speakers of Standard British English (SBE). The results confirmed that the disagreement between L&P and Grabe is due to dialect: the SBE speakers did not exhibit final lowering under either condition. They showed less steep $F0$ slopes, so all accents after the first had similar scaling, while the last accent showed a rise (rather than a fall, as in American English), enhancing the lack of final lowering. These differences show that final lowering is better seen as a phonological device that a linguistic variety may employ in a given melody. Also, the cross-varietal comparison supports the view that $F0$ downtrends consist of distinct components (declination, final lowering), each of which may be utilized independently of the others.

5aSC7. Optical phonetics and visual perception of lexical and phrasal boundaries in English. Edward T. Auer, Jr. (House Ear Inst., 2100 West Third St., Los Angeles, CA 90057), Sahyang Kim, Patricia Keating, Rebecca A. Scarborough, Abeer Alwaan (UCLA, Los Angeles, CA 90095-1543), and Lynne E. Bernstein (House Ear Inst., Los Angeles, CA 90057)

Detection of lexical and phrasal boundaries in the speech signal is crucial to successful comprehension. Several suprasegmental acoustic cues have been identified for boundary detection in speech (e.g., stress pattern, duration, and pitch). However, the corresponding optical cues have not been described in detail, and it is not known to which optical cues to boundaries speechreaders are attuned. In a production study, three male American English talkers spoke two repetitions each of eight pairs of sentences in two boundary conditions (one-word versus two-word sequences, one-phrase versus two-phrase sequences). Sentence pairs were constructed such that they differed minimally in the presence of a boundary. Audio, video, and 3D-movements of the face were recorded. Sentence pairs in both boundary conditions differed on both optical and acoustic duration measurements. A subset of the sentence pairs was also presented in a visual perception study. Pairs were chosen *a priori* because they differed or did not differ on the production measures. Fourteen perceivers detected boundary presence from video-alone in a forced-choice paradigm. Both lexical and phrase boundaries were reliably detectable by perceivers. The relation of visual intelligibility to the optical characteristics of production will be discussed. [Research supported by NSF IIS 9996088 and NIH DC04856.]

5aSC8. Pitch/amplitude mismatches in Mandarin. Deborah S. Davison (Res. Compliance, Stanford Univ., 1215A Welch Rd., MC 5401, Stanford, CA 94305)

Pitch and amplitude contours covary in Beijing Mandarin [Sagart (1986)]. Tianjin Mandarin amplitude range of stressed low (L) phonological tones is equal to that of stressed high (H) tones for $>50\%$ of L tokens. $F0$ range for 75–85 dB lexical tonemes T1L and T3LH versus T4HL was 100–158 Hz versus 135–180 Hz, respectively. Dimensions of $f0$/amplitude range for T1L 75–150 Hz/70–85 dB versus T4's H 110–180 Hz/70–85 dB were virtually identical: 75 Hz/15 dB versus 70 Hz/15 dB, respectively. Stressed T1L and T3LH low $f0$'s relatively higher amplitude may compensate for reduced perceptual saliency of low $f0$. T2''H'''s [Yip (2002)] pattern resembles T4H assuming T2's single H tonal target aligns to right rather than left syllable edge (cf. Beijing lexical T2 ''mid-to-high rising tone'') when fully stressed, hence bimoraic. Tonally unspecified left edge T2 $f0$ does not correlate with amplitude. Both monomoraic ''toneless-stressless'' neutral tone T0 syllables preceding stressed T1L and T3LH and lexical T1L and T3LH ''sandhi-changed'' to LH and H preceding stressed T1L and T3LH, respectively, have high $f0$ but lower amplitude than the following stressed low tone. Stress-conditioned $F0$/dB mismatches support the conclusion that high $f0$/dB attracts pitch accent [Davison (2003)], contrary to the assumption that lexical tone languages such as Chinese do not permit phrasal pitch accents internal to intonational phrases [Ladd (1996)].

5aSC9. Form and function of intonational phrase heads. Cynthia Girand (Dept. of Linguist., UCB 295, Univ. of Colorado, Boulder, CO 80309)

This study examines the phonetic and phonological features essential to the investigation of the internal structure of the intonational phrase. This is crucial for identifying and explaining the forms and functions of basic intonational units. There are two primary intonational contour modeling theories [Ladd (1983) and references therein]. One suggests that phrasal contours are the basic units of intonation. Within this theory, contour shapes are associated with particular functions or meanings. In contrast, a more recent theory claims that individual tones (i.e., abstract phonological units) are the basic units of intonation, and intonational contours result from the concatenation of adjacent tones in a phrase. Using 756 utterances

from the Switchboard and Buckeye corpora, the present study takes a closer look at the basic units that compose the intonational contour. While the nucleus has long been identified as a functionally important part of an intonational phrase, the head of an intonational phrase has not been considered, in kind. This work examines global and local phonetic and phonological features (e.g., intensity, pitch range, pitch height, downstepped tones) of intonational phrase heads in an attempt to better understand their forms and functions.

5aSC10. Italian Raddoppiamento: Prosodic effects on length. Rebeka Campos-Astorkiza (Univ. of Southern California, University Park GFS 301, Los Angeles, CA 90089, rebekaca@usc.edu)

Traditionally, Italian Raddoppiamento refers to a lengthening process that targets word-initial consonants after a final stressed vowel. Unfortunately, most theoretical accounts of Raddoppiamento lack a solid phonetic foundation. We report an acoustic study, based on the data obtained from four Tuscan Italian native speakers, investigating both word-initial consonants and vowels in the Raddoppiamento environment. Further, we consider two different prosodic positions, C Raddoppiamento phrase-internally versus across an intonational phrase boundary, which, according to previous analyses (Nespor and Vogel, 1986), should prevent the lengthening from occurring. Finally, stressed and unstressed environments are tested. Thus, the quality and magnitude of Italian Raddoppiamento lengthening as a function of segmental identity, prosodic context, and stress is reported. The results show that consonantal lengthening takes place as expected in the traditional Raddoppiamento environment. On the other hand, word-initial vowels do not lengthen. Stress has an effect on word-final vowel length that is incompatible with previous theoretical accounts of Raddoppiamento. Finally, the presence of an intonational phrase boundary does not, as previously predicted, categorically block the process. Overall, this empirical evidence challenges previous accounts of Raddoppiamento and provides a systematic phonetic documentation of the phenomenon. [Work supported by NIH.]

5aSC11. Children's use of prosody to identify ambiguous sets of compound nouns. Michiko Yoshida and William F. Katz (Univ. Texas—Dallas, Callier Ctr. for Commun. Disord., 1966 Inwood Rd., Dallas, TX, 75235, michiko@utdallas.edu)

Research has shown that young children can process word duration and fundamental frequency ($F0$) information in an adultlike manner to disambiguate simple conjoined English phrases [Beach *et al.*, J. Acoust. Soc. Am. **99**, 1148–1160 (1996)]. However, the processing of pause duration has not been investigated in this manner. The present study investigates how children use pause duration and $F0$ to perceive word boundaries, and whether the manner of information processing differs between children and adults. Adults and children (5 and 7 years old) completed a picture-pointing task based on auditory stimuli. Stimuli were computed-edited tokens of the words "sun," "flower," and "pot." From a recording by a male talker, vocoded tokens were created with five steps of pause duration and three steps of fundamental frequency. Both continua ranged between patterns suggesting the interpretations "sun, flowerpot" and "sunflower, pot." Preliminary results indicate a cue-trading relation between duration and $F0$ for all listeners, with pause duration playing the dominant role. These data suggest adultlike processing in children as young as 5 years old. Experiments with younger children and a formal evaluation of the pattern recognition processes used by listeners are currently underway in our laboratory.

5aSC12. Perception of stress in Thai. Rattima Nitisaroj (Dept. of Linguist., Georgetown Univ., 37th and O St., NW, Washington, DC 20057)

This study assesses the relative contribution of four parameters—duration, amplitude, fundamental frequency and vowel quality—to the perception of stress in Thai. Previous acoustic studies [e.g., S. Potisuk *et al.*, Phonetica **53**, 200–220 (1996)] have identified these four parameters as varying with stress, but perceptual tests in which these parameters are independently controlled have been needed. For this experiment, stimuli were created by digitally manipulating the four parameters in minimal pairs (compounds versus phrases) that differ only in stress pattern. Subjects were asked to choose between two contexts signaling whether the stimuli they heard contained a compound or a phrase token. Duration was found to be the strongest perceptual cue to stress. Sometimes, stress was signaled by the combination of duration and vowel quality. Results also showed an effect of subjects bias to judge noun–verb tokens as phrases and noun–noun tokens as compounds. If the bias conflicted with decision signaled by duration and vowel quality, effects caused by amplitude variation were observed. Finally, fundamental frequency did not appear to play any role in signifying stress. The results thus support the following hierarchy of perceptual correlates to stress in Thai: duration, vowel quality/bias, amplitude, and fundamental frequency.

5aSC13. Comparing rhythm and melody in speech and music: The case of English and French. Aniruddh D. Patel, John R. Iversen, and Jason C. Rosenberg (The Neurosci. Inst., 10640 John Jay Hopkins Dr., San Diego, CA 92121)

Does the prosody of a nation's language leave an imprint on its music? We address this question by comparing British English and French, a stress-timed vs syllable-timed language with salient intonational differences. We previously showed that an empirical difference between speech rhythm in the two cultures is reflected in instrumental music. In this study we expand on these rhythmic measurements and provide new data on melody. We compare English and French intonation using a measure which can also be applied to musical melodies. In a database of read speech we converted the intonation contour of each sentence into a sequence of vowel pitches, using the mean fundamental frequency of each vowel to represent its pitch. We found that the size of pitch intervals between successive vowels varied more in British English than in French speech. We then examined classical instrumental music and found that pitch intervals between successive notes varied more in English than in French music. We also examined differences in the way rhythm and melody are aligned in both speech and music. Overall, we find that the prosody of a culture's language is reflected in the structure of its instrumental music. [Supported by Neurosciences Research Foundation.]

5aSC14. Acoustic modifications in choral reading. Meredith A. Poore and Sarah H. Ferguson (Dept. of Speech-Lang.-Hearing, Univ. of Kansas, 3001 Dole Ctr., 1000 Sunnyside Ave., Lawrence, KS 66045, mpoore@ku.edu)

Choral reading is the condition in which one or more talkers read an assigned text aloud and in synchrony. Choral reading has several practical and theoretical applications, including use with beginning readers to enhance oral literacy, with people who stutter to evoke fluency, and as a tool for investigating speech rhythm. However, little is known about the speech acoustic modifications that occur when talkers perform the task of choral reading, or how different methods of eliciting choral reading may affect these modifications. In stuttering therapy, choral reading is usually elicited in pairs, either with two live talkers (choral reading) or one talker live and one prerecorded (track reading). While previous studies have assumed that these two reading conditions are equal, a recent study on speech timing suggests otherwise [F. Cummins, ARLO **3**, 7–11 (2002)]. Measurements of pitch, amplitude and vowel duration will be compared across solo, track and choral reading of texts which vary in their perceived rhythmic organization. Acoustic data will be interpreted in reference to various theories regarding the effectiveness of choral reading for fluency enhancement. [Work supported by a University of Kansas Honors Program Undergraduate Research Award.]

Session 5aUW

Underwater Acoustics: Localization, Classification and Processing

Harry A. DeFerrari, Chair
RSMAS Applied Marine Physics, University of Miami, 4600 Rickenbacker Causeway, Miama, Florida 33149

Contributed Papers

8:00

5aUW1. Multi-array passive search in a littoral environment. Donald R. DelBalzo, Erik R. Rike, and David N. McNeal (Neptune Sci., Inc., 40201 Hwy. 190 E, Slidell, LA 70461 delbalzo@neptunesci.com)

Acoustic barrier tactics were developed during the Cold War for deep, uniform underwater environments. Oceanographic and acoustic conditions in littoral environments are extremely complex and dynamic. The spatial and temporal variability of low-frequency signal and noise fields in these complex environments destroys the basic homogeneous assumption associated with standard tactical search concepts. Genetic algorithms (GAs) have been applied to both the signal and noise parts of this problem to produce near-optimal, nonstandard search tracks that maximize probability of detection in such inhomogeneous noise fields. In the present work, a GA was used to optimize tactics of several passive searchers by constructing optimal barriers in complex, littoral environments. The dynamic ambient noise model (DANM) was used to produce realistic, low-frequency, directional noise fields, based on discrete ship tracks. The cost function to be minimized was the probability that a target crossed the barrier undetected. Both standard and GA-derived tactics were evaluated and compared. The results show the importance of nonstandard tactics and careful consideration of environmental complexity when designing optimal passive search tactics. [Work sponsored by NAVSEA under the LCS project.]

8:15

5aUW2. Beamformer waveguide invariant source localization on the New Jersey shelf (USA) during winter acoustic propagation conditions—RAGS03. Altan Turgut, Bruce H. Pasewark, Marshall H. Orr (Naval Res. Lab., Acoust. Div., Washington, DC 20375), and Daniel Rouseff (Univ. of Washington, Seattle, WA 98105)

Horizontal and vertical beamformer (waveguide) invariants are used to robustly localize broadband noise sources (ships) using signals in the 50–200-Hz band. The localization is realized by using signatures recorded on one horizontal and three vertical arrays. In brief, waveguide invariant theory applied to beamforming by two vertical arrays provides a range ratio of the source to the receivers. Beamforming by a horizontal array provides time-evolving spectrum for a particular look direction (LOFARgram). As a result the trajectories of the striations observed in LOFARgrams can be used to estimate the speed, range and direction of a broadband source. In December 2003 the Naval Research Laboratory moored three vertical arrays and one horizontal array at ranges of 10, 20 and 30 km from two fixed acoustic sources on the New Jersey (USA) shelf. Acoustic signatures from passing merchant and research vessels were recorded on the arrays and used to demonstrate source localization. [Work supported by ONR.]

8:30

5aUW3. A new invariant method for instantaneous source range estimation in an ocean waveguide from passive beam-time intensity data. Sunwoong Lee and Nicholas C. Makris (MIT, 77 Massachusetts Ave., Cambridge, MA 02139, makris@mit.edu)

A new method is derived for instantaneous source range estimation in an ocean waveguide from passive beam-time intensity data received on a horizontal or vertical line array. While the method is as simple and robust as the "waveguide invariant" method described by Brekhovskikh and Lysonov (2003), it does differ significantly from the latter. Some advantages are that (1) it only requires source signal measurement at a single rather than many ranges so that range can be instantaneously estimated, and (2) it is invariant over all horizontally stratified ocean waveguides. The paramter β of the waveguide invariant method, on the other hand, varies significantly with vertical sound-speed structure. Since the invariant parameter of this new method varies only with the geometry of the array, the term "array invariant" method is here introduced. The method does not rely on near-field processing techniques but is also applicable at arbitrary far-field ranges. The method is applied to data from the Main Acoustic Clutter Experiment 2003 for source ranges between 2 to 10 km, where it is shown that simple, accurate, and efficient source range estimates can be made using this new method.

8:45

5aUW4. Source localization in environments with deterministic and stochastic uncertainties. Ralph N. Baer and Michael D. Collins (Naval Res. Lab., Washington, DC 20375)

Focalization is an approach for localizing an acoustic source when there are uncertainties in the sound speed or other environmental parameters [J. Acoust. Soc. Am. **90**, 1410–1422 (1991)]. The original work on focalization was based on deterministic parameters, but the concept was recently applied to problems involving uncertainties associated with internal waves [J. Acoust. Soc. Am. **115**, 2550 (2004)]. Due to a parameter hierarchy in which source position outranks environmental parameters, it is often possible to localize a source without determining the true environmental parameters. This is a very fortunate situation, especially when dealing with stochastic parameters that would be essentially impossible to determine. Focalization is currently being extended to problems involving uncertainties in both deterministic and stochastic parameters. [Work sponsored by ONR.]

9:00

5aUW5. Optimal sensor placement in highly variable acoustic fields to increase detection. Erik R. Rike and Donald R. DelBalzo (Neptune Sci., Inc., 40201 Hwy. 190 E, Slidell, LA 70461)

Optimal placement of sensors for detection depends on spatial and temporal properties of ambient noise (AN) and transmission loss (TL). These acoustic quantities are highly variable in littoral environments and their predictions are often overly smoothed when estimating detection probabilities. This variability was characterized using the dynamic ambient noise model (DANM) and the parabolic equation (PE) model. DANM was used to produce space and time estimates of directional noise based on discrete ship tracks and the PE was used to characterize spatial variability in TL. Then, Brown's algorithm for search planning was modified to include realistic ordered search against diffuse target distributions and used to determine the relative importance of non-homogeneous noise statistics and TL in optimizing search tactics. The new capability is COBRA (coherent optimization for Baysean resource allocation), which is efficient and accurate. The results in one littoral area show that spatial variations in

TL are twice as important as those in AN for optimal search planning. This work has immediate application to design of optimal sonobuoy field patterns and future application to mobile sonar search planning. [This work was sponsored by ONR under the LADC project.]

9:15

5aUW6. Horizontal array gain variability measured on the New Jersey shelf during late fall and early winter propagation conditions—RAGS03. Bruce H. Pasewark, Altan Turgut, Marshall H. Orr, Jeffery A. Schindall, Michael McCord, and Earl Carey (Naval Res. Lab., Code 7120, Washington, DC 20375-5350)

A conventional beamformer has been applied to 300 Hz CW and 300 Hz center frequency LFM acoustic signals received on a 96-channel 465 m bottomed horizontal array. The temporal history of the array gain for five subapertures (30, 60, 120, 240, 465 m) will be presented. The gain of each subaperture varied over time scales ranging from less than 1 min to 13 days. The time dependence is being correlated to sound speed variability induced by shelf slope front movement, the internal tide, nonlinear internal waves and atmospheric forcing. The longer apertures did not achieve ideal array gain. Temporal variability of the horizontal spatial coherence lengths will be discussed. The acoustic sources were moored 18 m above the bottom in 64 m of water. The propagation path was cross shelf. The receiving array was located near the New Jersey (USA) shelf break 20 km from the source. Water depth at the array location was 89 m. [Work supported by ONR.]

9:30

5aUW7. Hybrid joint probability density functions for active sonar. James M. Gelb and Brian R. La Cour (Appl. Res. Labs., Univ. of Texas, 10000 Burnet Rd., Austin, TX 78758)

Methods are presented for estimating joint probability density functions (PDFs) of statistically dependent features with a focus on sparse data with planned application for computing likelihood functions for feature-based target classification in active sonar. The estimators involve a new class of hybrid models, i.e., adjusted combinations of low-dimensional, nonparametric PDFs and high-dimensional, multivariate, parametric models. One goal is to preserve physically meaningful feature identity. The efficacy of the methods to model PDFs and to classify data sets using these PDFs will be presented for simulated and actual data. Two general forms for the PDFs are presented: (1) hybrid models, $\Pi_i p_i(f_i) \times[M_N(\mathbf{f})/\mathbf{\Pi_i M_i(f_i)}]$, where $\mathbf{f}$ represents a set of N features; $p_i(f_i)$ are the marginal probabilities, and $M_N(\mathbf{f})$ is a model for the N-dimensional multivariate PDF with model marginals $M_i(f_i)$. (2) Expansion models: the multivariate PDF is expanded in terms of its correlations, $\Pi_i p_i(f_i)[1+\Sigma_{i<j}\xi_{ij}+\Sigma_{i<j<k}\zeta_{ijk}+\cdots]$, with, for example, the two-point correlation functions defined as $\xi_{ij}=p_{ij}(f_i,f_j)/p_i(f_i)p_j(f_j)-1$. In both cases successes and failures of the models and necessary modifications to them are presented to improve their general utility. [This work is funded by the ONR.]

9:45

5aUW8. Object classification using low frequency scattering characteristics obtained from structural acoustic target models. Alessandra Tesei and Mario Zampolli (NATO Undersea Res. Ctr., Viale San Bartolomeo 400, 19138 La Spezia, Italy)

The capability to distinguish between different objects using low frequency scattered signals is a topic of fundamental interest to the sonar research community. To address this issue, a 3-D frequency domain finite element structural acoustics tool (FESTA) is used to synthesize multistatic scattered field time series from frequency sweeps. The time-frequency plots for scattering from a variety of manmade objects, like elastic spherical shells and cylindrical shells of various types, are analyzed, and the characteristic elastic effects are exploited to obtain low frequency classification clues. Comparisons between FESTA and other models are shown, and where available, experimental data are used to validate the results. Guidelines for the optimal level of modeling accuracy required to capture the relevant scattering details using the finite element tool are established, and the tradeoffs between accuracy, computation time and memory use are addressed.

10:00–10:15 Break

10:15

5aUW9. M-sequences and bi-static active sonar. Harry DeFerrari and Hien Hguyen (RSMAS, Univ. of Miami, FL)

The application of m-sequences to bi-static active sonar is evaluated with data from acoustic propagation experiments in the Florida Straits. Over the past 20 years, several numerical methods for processing M-sequences have been developed by Birdsall, Metzger and others. These methods come together for a possible sonar application by the following approach: (1) continuous transmission and reception of long m-sequences, (2) synchronous sampling to form a CON (complete ortho-normal) data set, (3) direct blast removal by HCCO (hyperspace cancellation by coordinate zeroing), and (4) a full range waveform Doppler search. Ultra-fast Hadamard transforms speed up the direct waveform pulse m-sequence pulse compression and the inverse pulse waveform transform and thereby allow timely execution of the intensive computational burden. The result is a numerically demonstrable approach that produces a gain of 36 dB over a simple pulse and 16 dB over other active signals. In addition, the direct arrivals and their Doppler leakage are eliminated, thus shifting the detection problem from a reverberation limited to a noise limited Doppler process. Here, m-sequence data from propagation experiments are reprocessed to test the fundamental concepts. Data results are shown to be in close agreement with ideal numerical simulations.

10:30

5aUW10. Multisensor registration for distributed active sonar systems. Brian R. La Cour, Kevin Johnson, and Son Quach (Appl. Res. Labs., Univ. of Texas, P.O. Box 8029, Austin, TX 78713-8029)

A technique is described whereby discrepancies between and biases within active sonar receivers may be corrected using acoustic signals. The method utilizes information contained in the direct blast recorded on each receiver over multiple active pings to estimate the ranges and bearings of the one or more sources relative to each receiver. Each sensor is assumed to have limited beamforming capability. In addition, the use of known acoustic scatters to resolve sensor misalignment will be discussed. Finally, results of an application of these techniques to active sonar data from the Deployable Experimental Undersea Multistatic Undersea Surveillance system (DEMUS) will be discussed and used to assess the algorithm. [This work was funded under ONR Contract No. N00014-00-G-0450-21.]

10:45

5aUW11. Target detection using a beam space time-reversal operator. Geoffrey F. Edelmann, Joseph F. Lingevitch, David M. Fromm, Charles F. Gaumond (Naval Res. Lab., 4555 Overlook Ave., Washington DC 20375, geoff@ccs.nrl.navy.mil), and David C. Calvo (Naval Res. Lab., Washington DC 20375)

Time-reversal operator methods such has DORT have recently garnered interest as promising techniques to ensonify ocean targets. However, measuring the time-reversal operator in the ocean by independently firing each element of a time-reversal mirror is difficult due to low signal-to-noise levels. This talk will discuss a portion of the shallow-water TREX-04 experiment in which energy was focused on a target using a vertical array of 64 source/receive transducers. The time-reversal operator was measured by transmitting energy from the full aperture of the time-reversal mirror instead of from just a single element. The time-reversal operator, made in beam space, was then used to successfully focus energy on a target 1 km away. [Work supported by the Office of Naval Research.]

11:00

5aUW12. An underwater acoustic detection experiment based on forward scattering in a time reversal mirror. J. Mark Stevenson, Alessandra Tesei, Piero Guerrini, Pierangiolo Boni (NATO Undersea Res. Ctr., La Spezia, Italy), Philippe Roux, Heechun Song, William S. Hodgkiss, William A. Kuperman, and Tuncay Akal (Scripps Inst. of Oceanogr., La Jolla, CA)

Following earlier work by Song *et al.* [IEEE, J. Ocean. Eng. **28**, (2003)] we conducted an underwater acoustic barrier experiment based upon forward scattering in a time reversal mirror. The experiment was conducted at a carrier frequency of 15 kHz in a shallow water waveguide. The waveguide length was about 16 water depths with one interface being an undulating, soft seafloor. After establishing a focus at one end of the waveguide, stationary and moving objects were placed in the water column, including an autonomous undersea vehicle (AUV). Before and after comparison of the received acoustic signal, including the temporal and spatial abberation of the focus, provided object detection. Several anomaly detection techniques were applied to the acoustic data in post-processing. Target speed was observed to be a discriminator in anomaly detection performance, suggesting that a combination of different anomaly detection techniques is probably desirable in addressing the moving target problem. [Work supported by NATO and ONR.]

11:15

5aUW13. Examination of bit error characteristics due to a drifting source in passive phase conjugation underwater acoustic communications. Jong R. Yoon, Kyu-Chil Park (Div. of Electron., Comput. and Information Eng., Pukyong Natl. Univ., Korea 608-737, jryoon@pknu.ac.kr), and Daniel Rouseff (Univ. of Washington, Seattle, WA 98105)

Recent experimental work in acoustic communications using passive phase conjugation has shown that the demodulation error depends on the relative drift rate between the transmitter and receiver [Rouseff *et al.*, IEEE J. Ocean Eng. **26**, 821–831 (2001)]. The observed effect involves the mismatch between the initial impulse response and the subsequent response after the source or receiver has changed locations. The result can also be interpreted in terms of the spot size of the retro-focused field. In the present work, the effect of relative drift between source and receiver is studied by numerical simulations. The communications bit error rate is quantified as a function of drift rate, carrier frequency, array geometry and source-receiver range. [This work was supported by Pukyong National University Research Abroad Fund in 2003.]

11:30

5aUW14. Focal depth shifting of a time reversal mirror in a range-independent waveguide. Shane C. Walker, Philippe Roux, and William A. Kuperman (Marine Phys. Lab., S.I.O., UCSD, 9500 Gilman Dr., Mail Code 0238, La Jolla, CA 92039-0238, shane@physics.ucsd.edu)

A time reversal mirror refocuses back at the original probe source position. A goal has been to refocus at different positions without model based calculations. A method to refocus at different ranges has been developed earlier [Song *et al.*, J. Acoust. Soc. Am. **103**, 3224–3240 (1998)] using frequency shifting. Here we present a technique to refocus at different depths than the original probe source in a shallow ocean range-independent waveguide. The requirement is to collect data from various ranges at a single depth, as from a moving broadband radiator, over a distance sufficient to construct the relevant frequency-wavenumber structure of the waveguide. With this information, it is then possible to focus at arbitrary depth at any of the ranges that the probe source data were taken. Theory and laboratory measurements are presented.

FRIDAY AFTERNOON, 19 NOVEMBER 2004 PACIFIC SALON 3, 1:00 TO 4:45 P.M.

Session 5pUW

Underwater Acoustics: Noise, Sound Sources and Sonar Systems

David L. Bradley, Chair

Applied Research Laboratory, Pennsylvania State University, P.O. Box 30, State College, Pennsylvania 16804

Contributed Papers

1:00

5pUW1. Underwater sound pressures from marine pile driving. Richard B. Rodkin and James A. Reyff (Illingworth & Rodkin, Inc., 505 Petaluma Blvd. South, Petaluma, CA 94952, rrodkin@illingworthrodkin.com)

Marine pile driving of steel shell piles has resulted in high underwater sound pressures that have been lethal to fish. High sound pressures have resulted in harassment of pinnepeds under the marine mammal protection act. Most waterways in the nation include fish and marine mammals that are protected by State and/or Federal agencies. Impacts from pile driving have contributed to costly construction delays for some major bridge projects. Recent construction activities in the marine environments of Northern California have provided the opportunity to characterize these sound pressures and evaluate control measures to protect fish and marine mammals. Sound control measures evaluated include different pile-driving methods, cofferdams (with and without water), confined air bubble curtain systems, and unconfined bubble curtain systems. Some control measures have achieved over 30 dB of noise reduction. Each situation, however, presents difficulties in achieving targeted reduction goals.

1:15

5pUW2. Sounds and vibrations recorded on and beneath landfast sea ice during construction of an artificial gravel island for oil production in the Alaska Beaufort Sea. Charles R. Greene, Jr. (Greeneridge Sci. Inc., 1411 Firestone Rd., Goleta, CA 93117)

Underwater and airborne sounds and ice-borne vibrations were recorded from sea-ice near an artificial gravel island during its initial construction. Recordings were made in winter 2000 when BP began constructing Northstar Island about 5 km offshore. Underwater and airborne sounds were recorded over frequencies from 10 to 10 000 Hz; particle motion in the ice was recorded from 10 to 500 Hz. Ice thickness was about 1.6 m.

Water depth at the island was 12 m, and depths ranged between 4 and 13 m at recording stations. Recording distances were from about 100 m to over 5 km. Activities recorded included ice augering, pumping sea water to flood the ice and build an ice road, a dozer plowing snow, cutting ice with a Ditch Witch, trucks hauling gravel over an ice road to the island site, and a backhoe trenching the sea bottom for a pipeline. Dominant frequencies were generally below 500 Hz. The Ditch Witch cutting ice was one of the stronger sounds and was audible underwater and as ice vibration to distance 3 km. Other sounds diminished to background levels by a distance of 2 km. Airborne sound detection depended on wind speed and direction. [Work supported by BP.]

1:30

5pUW3. Issues for improved characterization of the acoustic field due to radiated noise of ships in shallow water environments. Christopher Barber and Kyle M. Becker (Appl. Res. Lab., Penn. State Univ., P.O. Box 30, State College, PA 16804-0030)

Estimation of the sound pressure levels radiated from ships to underwater sensors in littoral environments is a problem of increasing importance. Current areas of interest range from underwater surveillance and homeland security of ports, to the threat posed to naval vessels by influence mines. Recent literature addressing shallow water propagation has focused primarily on object detection using active sonar systems or on environmental characterization through inverse methods. While conclusions from these studies regarding the impact of environmental variability on acoustic propagation may be generally applicable, there are several additional areas that need to be investigated in order to understand the acoustic field at short ranges in shallow water. This presentation seeks to identify areas where further investigation may be warranted. In particular, existing propagation codes are examined for their applicability and/or limitations for the modeling of near-field radiation from nonsimple distributed sources, evaluation of complex source signals containing both broadband noise and narrow-band tones over a broad frequency range, and the impact of small-scale local variability. This presentation seeks to motivate a discussion toward new approaches and methodologies for addressing gaps between modeling requirements and capabilities. [Work supported by PEO-LMW, AIMS Program.]

1:45

5pUW4. Modeling of angular and range acoustic energy distribution from seismic exploration array in the Northern Gulf of Mexico. Arslan Tashmukhambetov, Natalia A. Sidorovskaia (Phys. Dept., Univ. of Louisiana at Lafayette, UL BOX 44210, Lafayette, LA 70504), George E. Ioup, and Juliette W. Ioup (Univ. of New Orleans, New Orleans, LA 70148)

Broad-band experimental recordings (up to 25 kHz) of the acoustic energy arrivals from surface seismic exploration array by a bottom-moored hydrophone were gathered by the Littoral Acoustic Demonstration Center (LADC) in the Northern Gulf of Mexico during the summer of 2003. Experimental data are available for a wide range of horizontal distances (up to 7 km) and arrival angles. The numerical reconstruction of the acoustic radiation pattern and the acoustic energy decay law in range subject to the actual propagation conditions in the Northern Gulf of Mexico based on the calibrated recorded data by the identification of the zero-to-peak value in the direct pulse arrival is presented. The comparison of the measured energy losses with the expected ones due to spherical and cylindrical spreading is given. The identification of different types of arrivals (direct pulse, surface ghost, bottom reflected, etc.) and evaluation of their frequency content depending on the azimuth and emission angles and the distance to the array are conducted. The complexity of propagation and source models required to predict the identified arrivals is discussed.

2:00

5pUW5. Noise characterization at the international station of Crozet Islands (H04). Antoine Roueff, Pierre-Franck Piserchia, Jean-Louis Plantet, Yves Cansi, and Gerard Ruzie (CEA/DASE/LDG, BP12, 91680 Bruyeres-le-Chatel, France)

In the context of the Comprehensive Nuclear-Test-Ban Treaty (CTBT), the hydroacoustic network gives the ability to state parties to check the natural and artificial activity in the oceans. To achieve this purpose, 11 hydroacoustic stations are going to be established. One of these stations has recently been launched by the CEA (France) near the Crozet Islands in the Indian Ocean. In this paper, we characterize the ambient noise of this new station. This study will enable one to estimate the detectability of the station. In this analysis, we associate the tide height with the noise level on the recorded signals. The lunar calendar perfectly matches the hydroacoustic noise variation. In addition, we will show the influence of this noise evolution on the rate of detection. In order to have a good statistic, we process a hundred days of data (recorded at 250 Hz). The tide heights supplied by "le Laboratoire de gophysique et Ocanographique de Toulouse," are sampled with a 1-h step. Furthermore, the spectral characteristic of the ambient noise is also given. Finally, a comparison with other international stations, Diego Garcia (H08) and Cape Leeuwin (H01), is given.

2:15

5pUW6. A swept-frequency implosive source. LeRoy M. Dorman and Allan W. Sauter (Marine Physical Lab., Scripps Inst. of Oceanogr., UCSD, 9500 Gilman Dr., La Jolla, CA 92093-0220)

Implosive underwater sound sources, such as the 20-liter single-shot device we have constructed, generally emit a rarefaction followed by a compression when the implosion is complete. Some purposes, however, are better served by a swept-frequency, rather than an impulsive, signature, since the lower peak pressures of the swept source are less likely to cause environmental damage. Our current plan for a repeating implosive source operates by venting a bubble (initially at ambient pressure) into a low-pressure receiver through a control valve. Performing this process in steps with abrupt pauses induces oscillations characteristic of the several sizes of the evolving bubble. From modeling the bubble oscillations using the Gilmore bubble equation, it appears feasible to produce a range of a factor of 2 in frequency, with a peak pressure of a megapascal. The design of the control valve, however, is challenging, as it was for the single-shot version. The repeater, however, operates on air, whose density and viscosity are lower.

2:30

5pUW7. An airgun array source signature model for environmental impact assessments. Alexander O. MacGillivray, N. Ross Chapman (School of Earth and Ocean Sci., Univ. of Victoria, P.O. Box 3055 STN CSC, Victoria, BC V8W 3P6, Canada, alexm@uvic.ca), and David E. Hannay (JASCO Res. Ltd., Victoria, BC V8Z 7X8, Canada)

Environmental impact assessments for seismic surveys often include estimates of radially propagating sound levels, which are used to determine marine mammal impact zones. Sound levels may be estimated using computer-based acoustic modelling techniques but these require an accurate description of the survey source signature—arrays of airguns, in particular, have complex, highly directional source functions that depend on the array layout. To address this requirement, an airgun array source signature model has been developed for the purpose of underwater noise level prediction. The source model is based on published descriptions of the physics of airgun bubble oscillations and radiation [A. Ziolkowski, Geophys. J. R. Astron. Soc. **21**, 137–161 (1970)] and includes the effects of port throttling, motion damping and bubble interactions. The output of the model is a collection of notional signatures which may be used to compute the source function of the array in any direction. Free parameters in the model have been fit to a large collection of existing airgun signature data, for airguns ranging from 5 to 185 in^3. The output of the model is suitable for estimating sound levels resulting from airgun survey activity. [Work supported by NSERC.]

2:45

5pUW8. Measured and predicted array response to the vertical directivity of ambient noise. Andrew Holden (DSTL Winfrith, Dorchester, DT2 8WX, UK, apholden@dstl.gov.uk)

A great deal has been published on ambient noise. Most of this has covered (a) omni directional levels, and (b) the vertical and horizontal directivity of shipping noise at low frequencies. There is some published material on the vertical directivity of wind generated noise at lower frequencies, but very little at higher frequencies. In order to study wind generated ambient noise at higher frequencies, a small planar array from QinetiQ Bincleaves has been used to make ambient noise measurements. As well as measurements, a model called CANARY has been written to predict ambient noise vertical directivity and array responses to this noise. This paper contains some comparisons between CANARY predictions and (a) previously measured array response to ambient noise vertical directivity at 4.5 kHz, and (b) analysis of the planar array measurements at 25 kHz. The paper shows the nature of the ambient noise vertical structure and that the CANARY predictions are in good agreement with the measurements.

3:00–3:15 Break

3:15

5pUW9. Noise audit model for acoustic vector sensor arrays on an ocean glider. Michael Traweek (Office of Naval Res., 800 N. Quincy St., Arlington, VA 22217-5600), John Polcari (AETC, Inc., Arlington, VA 22202), and David Trivett (Georgia Inst. of Technol., Atlanta, GA 30332-0405)

The design of any new sensing array is a complex endeavor, demanding clear understanding of both signal exploitation options and the underlying noise environment (including ambient, structural, flow and electronic components), including cross-channel characteristics. The goal of this effort is to outline an approach for assessing potential advantages of vector sensors to provide ocean gliders with the acoustic sensory input necessary to execute ISR, characterize the environment, and support other mission-enhancing behaviors. At the heart of the approach is development of a noise audit model (NAM) which (coupled with appropriate signal characterizations) enables realistic and comparable evaluation of optimal processing performance for different classes of sensors (e.g., scalar, vector, or tensor). The NAM is a design tool that permits balanced design of the various array components so that the array output is ambient noise limited in the quietest operating environment. Noise components are broken into different source-transmission path chains; correct transfer functions are applied along each path; and sensor/array outputs are then obtained by incoherent sum. Properly structured, rapid evaluation of performance limits can also be supported. The approach will be discussed, and examples of both NAM components and associated performance evaluations will be presented. [Work supported by ONR.]

3:30

5pUW10. Simulation of synthetic aperture sonar performance under experimentally measured environmental fluctuations. Timothy H. Ruppel and Steve Stanic (Naval Res. Lab., Stennis Space Ctr., MS 39529)

In June 2003, NRL conducted a series of coherence measurements in very shallow water (8 m depth) off the coast of Panama City, FL under a wide range of ocean conditions. Low-frequency (1–10 kHz) and high-frequency (10–200 kHz) measurements were taken. The results show appreciable variability even during fairly calm conditions. This paper will report on the effects of this experimentally measured variability on the performance of a simple computer-simulated synthetic aperture sonar (SAS). SAS works by rapidly and repeatedly pinging a target from a moving acoustic array platform. If the underwater acoustic environment changes appreciably between pings, random phase errors are introduced on the individual SAS receiving elements, with the result that the SAS image might be significantly degraded. The use of a computer-modeled SAS allows for the investigation of these effects in the absence of such phenomena as random receiver motion (motion compensation) or specific auto-focusing algorithms. [Work supported by the ONR Program Element No. 62435N.]

3:45

5pUW11. Multibeam sonar observations of hydrodynamic forcing functions and bubble persistence in a ship wake. Thomas C. Weber, Anthony P. Lyons, and David L. Bradley (Appl. Res. Lab. and the Grad. Prog. in Acoust., P.O. Box 30, State College, PA 16804, tcw141@psu.edu)

High frequency (240 kHz) upward-looking multibeam echosounder data were collected from the wake of a twin-screw 50-m research vessel. Volumetric backscatter in each of the 101 1.5×1.5-deg beams was recorded at a repetition rate of approximately 10 s as the ship passed over the moored sonar system. The multibeam sonar's wide field of view and the low wake drift rate made it possible to image the wake over its entire lifetime of 30 min. The wake quickly separates into distinct port and starboard components. It is suggested that this is caused by vortices shed from the hull that propagate orthogonally to the ship's direction, entraining bubbles and creating a convergence zone on the outboard side of the vortices as they oppose the buoyant rise of the bubbles. The speed at which the vortices separate is calculated from the data and used in a simple model describing the evolution of the bubbles in the wake. Results show that the hull shedded vortex model is plausible, and that the gas transfer rate from the bubbles must be approximately 25 times less than it would be for surfactant-free bubbles in order to explain the bubble persistence observed in the data.

4:00

5pUW12. Synthetic aperture sonar applied to a multibeam volume imaging sonar. Michael Hamilton (SAIC, 10260 Campus Point Dr., MS-C4, San Diego, CA 92121)

The possibility of using synthetic aperture sonar (SAS) processing techniques on a multibeam volume imaging sonar (the Buried Object Search Sonar, or BOSS) is explored. A system algorithm is developed which involves first forming several parallel synthetic apertures from a translated array oriented in the cross-track direction, then beamforming the synthetic apertures to form a fully 3D volumetric representation. The validity of the technique is verified by taking data from a subaperture of an experimental cable-guided multibeam system and applying the SAS processing string. Test targets are examined, including spheres and cylinders. The process proved quite robust, particularly considering that the system was not originally designed for SAS processing, and no sophisticated motion compensation schemes were employed. Implications for multibeam SAS processing of future AUV-based systems are discussed. [Work supported by ONR's National Research Enterprise Internship Program, and performed at Naval Surface Warfare Center, Panama City, FL.]

4:15

5pUW13. High-frequency passive sonobuoy designs with PVDF wires and their predicted performance. Juan Arvelo, Jr., Patrick Ferat, and Ron Mitnick (Appl. Phys. Lab., Johns Hopkins Univ., 11100 Johns Hopkins Rd., Laurel, MD 20723-6099)

A novel piezoelectric material technology is used to design high-frequency (>10 kHz) air-deployed sonobuoys that exploit ambient noise anisotropy to enhance their passive performance. High frequencies have the advantages of smaller arrays and reduced clutter from distant shipping noise. PVDF wires are used to design vector sensors in azimuth with vertical directionality. Such hydrophones are used to design vertical and volumetric array configurations. Performance predictions of developed array designs are presented for selected environmental conditions. Results clearly indicate that vertical aperture is necessary to increase array gain against wind-driven noise by exploiting its anisotropy while azimuthal discrimination is required to enhance gain against nearby shipping interference. [This work is sponsored by the Office of Naval Research (ONR).]

4:30

5pUW14. Hair cell transducer design based on optical fiber Bragg gratings. Francois M. Guillot, D. H. Trivett, and Peter H. Rogers (George W. Woodruff School of Mech. Eng., Georgia Inst. of Technol., Atlanta, GA 30332-0405, francois.guillot@me.gatech.edu)

The development of a biomimetic transducer that reproduces the sensing mechanism of a single hair cell is presented. When light from a broadband source is input into an optical fiber terminated by a Bragg grating, the undisturbed grating reflects a fraction of the incident light at the Bragg wavelength, and the corresponding light intensity can be measured by a photodiode. The end of the fiber onto which the grating is written is adhered to two spacers separated by a 1-mm gap, and this assembly is adhered to two plates connected by a hinge, which is located below the gap. One plate is held rigidly and the tip of the other (free) plate experiences transverse vibrations, when ensonified. These vibrations induce longitudinal strains in the length of the fiber situated over the gap. Each unstrained portion of the grating reflects a small amount of light at the Bragg wavelength, resulting in two interfering light signals. The strain modulates the interference signal and produces an ac voltage at the photodiode output. The characteristics and performance of the transducer are presented, and its applications as the building block for underwater vector and tensor sensors are discussed. [Work supported by ONR.]