Session 5aAA

Architectural Acoustics: Current Topics in Measurement and Modeling

William M. Hartmann, Chair
Michigan State Univ., Dept. of Physics and Astronomy, East Lansing, MI 48824

Contributed Papers

8:00

5aAA1. An experimental study of how paint affects the specific airflow resistance and visual appearance of nonwoven scrims on acoustical ceiling tiles. William Frantz and Anthony Wiker (Armstrong World Industries Inc., 2500 Columbia Ave., Lancaster, PA 17604, whfrantz@armstrong.com)

This study explores how paint affects the specific airflow resistance and visual appearance of nonwoven scrims used on acoustical ceiling tile. Nine different scrims are included. Each scrim is finished with two application rates of paint. The specific airflow resistance is measured by the ASTM C522 method. The painted samples are also scanned and the images processed by ImagePro software to characterize visual aspects such as number of features, area of features, and roundness. It is found that specific airflow resistance and visual responses can be correlated to the resistance of the unpainted scrim, the surface tension characteristics of the scrim and paint, and the application rate of the paint. The resulting empirical models can be used to understand design tradeoffs between airflow resistance and visual impact.

8:15

5aAA2. Maximum length sequences versus random telegraph noise in measuring impulse responses and transfer functions. Neil L. Aaronson and William M. Hartmann (Dept. of Phys. and Astron., Michigan State Univ., 4230 BPS Bldg., East Lansing, MI, 48824)

Maximum length sequences (MLS) are commonly used to measure impulse responses and transfer functions of linear systems. The remarkable utility of the MLS comes from the property that its autocorrelation is approximately a delta function, and this approximation becomes more valid for higher orders (lengths) of MLS. However, the autocorrelation of a random telegraph noise (RTN) sequence also approaches a delta function in the long-sequence limit. Advances in computer hardware make it possible to use long sequences of RTN. Consequently one wonders if the MLS still confers an advantage. This study presents a quantitative analysis of the accuracy of MLS and RTN for all lengths up to $2^N-1=262{,}143$. RTN sequences were generated by scrambling the elements of an MLS. A special case wherein the approximate length of the impulse response is prior knowledge allows the measured impulse response to be windowed. Even with this advantage applied to RTN (but not to MLS), MLS measurements are clearly more accurate. The numerical experiments conclude than any MLS sequence longer than the impulse response outperforms a RTN of any practical order by an order of magnitude. [Work supported by the NIDCD grant DC 00181.]

8:30

5aAA3. Simulations of head-related transfer functions in wideband acoustics. Eira T. Seppälä, Ole Kirkeby, Asta Kärkkäinen, Leo Kärkkäinen (Nokia Res. Ctr., Itämerenkatu 11-13, FI-00180 Helsinki, Finland), and Tomi Huttunen (Univ. of Kuopio, Kuopio, Finland)

Head-related transfer functions (HRTFs) have been simulated in three dimensions for a head-and-torso model for the entire audio frequency range, from 20 Hz to 20 kHz. As opposed to data acquired through measurements, the results derived from computer simulations are free from the effects of noise and imperfections in the electro-acoustic chain. In addition, the spatial resolution can easily be made better than in any practical experiment. The simulations have been performed using an ultraweak variational formulation method for solving Helmholtz equation. The numerical method with its parallel computing capability is efficient, enabling numerical calculations of large physical systems, e.g., of size $0.4\times0.5\times0.8$ m, at high frequencies. The method uses plane-wave basis functions instead of polynomial basis as in standard finite-element methods (FEM), resulting in the need of much sparser volume mesh than in FEM. For HRTF calculations, a so-called perfectly matched layer has been utilized and the solutions are derived in the far field. Thus, exterior problems often solved using boundary-element method are dealt with. The specific HRTF simulations have been performed using fully reflecting, sound hard, boundaries on the head and torso, and also with complex impedance boundaries on the torso such as clothing.

8:45

5aAA4. Reverberation time, mean-free-path, and sound absorption in concert halls—Numerical examination by computer simulation. Takayuki Hidaka and Noriko Nishihara (Takenaka R&D Inst., 1-5-1, Otsuka, Inzai, Chiba 270-1395, Japan)

Sabine and Eyring equations are commonly used to estimate the reverberation time (RT) in concert halls. Derivations of the two equations are based on well-defined and different physical assumptions, and they are both slightly different from the realistic situation in halls. In many researches so far, comparatively simple room shape has been utilized to study the question of which equation is preferable. In this study, the sound fields in rooms with more complex or wide-ranging shape are numerically analyzed by applying the computer simulation technique and CAD models for architecture. Focusing on the fundamental relation between the mean free path as a physical measure of the room and the RT, validity of the two equations in actual halls is discussed. Next, the influence of the room shape or the choice of the RT equation on the effective sound absorption is examined.

9:00

5aAA5. A high-frequency broadband energy-intensity boundary element method for specular reflection in three-dimensional reverberant enclosures. Jerry W. Rouse, Linda P. Franzoni, and Donald B. Bliss (Mech. Eng. and Mater. Sci., Duke Univ., Durham, NC 27708, linda.franzoni@duke.edu)

A new method is presented for predicting the spatial variation of mean-square pressure within three-dimensional enclosures having steady-state, high-frequency broadband sound fields. The enclosure boundaries are replaced by a continuous distribution of uncorrelated broadband directional sources, which provide constituent fields expressed in terms of mean-square pressure and time-averaged intensity variables. Boundary conditions for radiating and absorbing surfaces are recast in energy and intensity variables. Superposition of these source fields in a numerical boundary element formulation leads to the prediction of overall mean-square pressure and time-averaged intensity as a function of position. Both specular and diffuse reflection boundaries can be accommodated. In contrast to the traditional boundary element approach, this method is independent of frequency, and each element has multiple unknowns, namely the strengths of directivity harmonics. For verification, exact analytical solu-

tions for the mean-square pressure distribution in several model problem enclosures were obtained by modal sums and frequency integration and compared to the new method. The comparisons show the method is very accurate, and it is extremely computationally efficient in comparison to classical modal and boundary element approaches. Results also show that diffuse field methods do not provide an accurate simulation for specular reflection cases.

9:15

5aAA6. Energy-based field quantities in reverberation chamber measurements. David Nutter, Timothy Leishman, and JohnPaul Abbott (Brigham Young Univ., N247 ESC, Provo, UT 84602, dave_nutter@hotmail.com)

Traditional reverberation chamber measurements, such as sound power and sound absorption measurements, rely on the evaluation of acoustic pressure at many field positions. Other field quantities could be better suited for these applications if they had greater spatial uniformity and required fewer source-receiver positions to produce accurate measurement results. This paper explores the possibility of using energy-based field quantities for this purpose and provides several analytical and experimental results to show their benefits.

9:30

5aAA7. Analysis method for temporal decay of diffuse broadband sound fields in enclosures. Donald Bliss, Ben Manning, and Linda Franzoni (Mech. Eng. and Mater. Sci., Duke Univ., Durham, NC 27708)

An analysis method to predict the behavior of time-dependent broadband diffuse sound fields in enclosures is described. A formulation utilizing time-dependent broadband energy-intensity boundary sources, including propagation time delays, is developed as a numerical boundary element method. An interpolation method is used to re-express the actual delays in terms of a discrete set of integer-multiple delays, thereby facilitating the numerical solution. The method is demonstrated for the prediction of temporal decay in interconnected one-dimensional channels and in two-dimensional enclosures. Temporal behavior is expressed in terms of a higher-dimensional eigenvalue/eigenmode problem, with boundary source strength distributions expressed as eigenmodes. In temporal decay from steady state, solutions exhibit rapid short-time spatial redistribution of energy, followed by long-time decay of a predominant spatial eigenmode. Long-time decay depends on behavior of the most slowly decaying eigenmode and the relative source-panel strengths do not depend on initial conditions. Short-time adjustment and decay depends on initial source characteristics and the relative distribution of absorbing material. An interesting feature of the short-time behavior is that the time-decaying eigenmodes are primarily a manifestation of energy redistribution rather than absorption.

9:45

5aAA8. Absorption-based scaling method to predict steady-state broadband sound fields in enclosures. Donald Bliss (Mech. Eng. and Mater. Sci., Duke Univ., Durham, NC 27708), Jerry Rouse (Sandia Natl. Labs., Albuquerque, NM 87185-0847), and Linda Franzoni (Duke Univ., Durham, NC 27708)

An absorption-based analysis method is developed for steady-state broadband sound fields in enclosures having diffuse or specular reflection boundaries. The wall absorption is expressed as an overall spatially averaged value, with spatial variations around this mean. Interior pressures and intensities are expressed in a power series of the overall absorption, treated as a small parameter, thereby giving a separate problem at each order. The first problem has a uniform mean-square pressure level proportional to the reciprocal of the absorption parameter, as expected. The second problem gives a mean-square pressure and intensity distribution that is independent of the absorption parameter and is primarily responsible for the spatial variation of the reverberant field. This problem depends on the location of sources and the spatial distribution of absorption, but not absorption level. Higher-order problems proceed at powers of the absorption parameter, but are essentially small corrections to the primary spatial variation. The primary spatial variation problem is solved using two different methods. Although formally developed based on treating absorption as a small parameter, the method is demonstrated to work remarkably well in practice up to large absorptions, and to provide insight into the behavior and design of acoustic spaces.

FRIDAY MORNING, 9 JUNE 2006 ROOM 551A, 7:55 A.M. TO 12:35 P.M.

Session 5aABa

Animal Bioacoustics and Psychological and Physiological Acoustics: Is Hearing All Cochlear?

Neil P. McAngus Todd, Chair

Univ. of Manchester, Psychology Dept., Oxford Rd., Manchester, M60 1QD, UK

Chair's Introduction—7:55

Invited Papers

8:00

5aABa1. Is all hearing cochlear?—revisited (again). Neil P. McAngus Todd (Faculty of Life Sci., Univ. of Manchester, M60 1QD, UK, neil.todd@manchester.ac.uk)

Over 70 years ago John Tait raised a question with regards to the certainty that the line of division between the hearing and vestibular portions of the VIIIth cranial nerve had been drawn at the proper place [J. Tait, Ann. Otol. Rhinol. Laryngol. **41**, 681–704 (1932)]. His reason for raising this question was the evidence which existed at that time that the otolith organs of anamniotes (fish and amphibians), most commonly the sacculus, were demonstrably organs of hearing in many species. He went on to speculate about a possible auditory role for the otolith organs in amniotes (reptiles, birds, and mammals), including in humans, although at this time there was no direct evidence for this. Since this time, however, there has developed a significant literature documenting evidence, not only for anamniote otolithic hearing, but also for a conserved acoustic sensitivity for the amniote otoliths. In the case of humans, this

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sensitivity has been exploited for the development of new acoustic tests of vestibular function. The time is ripe 70 years on, therefore, for us to revisit John Tait's question and explore the possibility that the sacculus may indeed contribute to human hearing. [Work supported by the Royal Society.]

8:25

5aABa2. Structure/function relationships in the saccule of fishes. Arthur N. Popper, Xiaohong Deng, David Zeddies (Dept. of Biol., Univ. of Maryland, College Park, MD 20742, apopper@umd.edu), and Mardi C. Hastings (Office of Naval Res., Arlington, VA 22203)

The saccule is morphologically the most diverse of the otolithic end organs among fish. Each otolithic end organ contains a sensory epithelium with numerous hair cells. The apical ends of the hair cell ciliary bundles contact a solid dense otolith. Relative motion between the sensory epithelium and otolith results in stimulation of the hair cells. Diversity in the saccule includes the overall shape and size of the end organ, the number of hair cells, the length of the ciliary bundles, and the shape and size of the otolith. Functionally, the saccule subserves both hearing and vestibular senses. There is some correlation between the orientation pattern of the sensory hair cells with whether fish are specialized for hearing or not. Yet, there is no real understanding of the relationship between structural diversity and functional differences. In effect, it is not known whether the structural diversity indicates adaptations for different auditory and/or vestibular functions, or whether the structural diversity reflects different ways for taxonomically and ecologically diverse fish to do similar ear functions. This talk will explore these issues using data from highly diverse deep-sea fish and direct measures of responses of the inner ear to sound.

8:50

5aABa3. Acoustically responsive fibers in the mammalian vestibular nerve. John J. Guinan, Jr. (Eaton Peabody Lab., Massachusetts Eye & Ear Infirmary, Harvard Med. School, 243 Charles St., Boston, MA 02114)

There are afferent fibers in the vestibular nerve of cats and guinea pigs that respond to sound at levels within the normal range of hearing. Most acoustically responsive fibers are in the inferior vestibular nerve and have irregular spontaneous activity, although not as irregular as cochlear afferents. Single-fiber labeling shows that acoustically responsive, irregularly discharging (ARID) fibers originate in the saccule. ARID fibers traced centrally arborized extensively in vestibular nuclei and ventromedial to the cochlear nucleus. ARID fibers had broad, V-shaped tuning curves with best frequencies between 500 and 1000 Hz, thresholds of 90 dB SPL or more, and shapes comparable with tuning-curve tails of cochlear afferents. ARID fibers synchronized to a preferred phase of the tone cycle at levels approximately 10 dB lower than their firing-rate threshold. ARID spike rates increased monotonically with sound level without saturating at 115 dB SPL. Acoustic clicks evoked spikes in ARID fibers with minimum latencies less than 1.0 ms. Contraction of the middle-ear muscles decreased responses to sound, consistent with the sound transmission path being through the middle ear. In summary, there is strong evidence that the mammalian saccule responds to sound and sends acoustic information to the central nervous system. [Work supported by NIDCD RO1DC00235.]

9:15

5aABa4. Assessing saccular (otolith) function in man. James G. Colebatch (Dept. of Neurol., Prince of Wales Hospital, Randwick Sydney, NSW 2031 Australia)

Scattered observations have suggest that the vestibular apparatus may have retained sound sensitivity in man and primates [e.g., Young *et al.*, Acta Otolaryngol. **84**, 352–360 (1977)]. In 1964 Bickford *et al.* [Ann. N.Y. Acad. Sci. **112**, 204–218 (1964)] reported the properties of the "inion response," evoked by clicks and which in subsequent publications they deduced was likely to be arising from the saccule. Colebatch *et al.* [Colebatch, *et al.* J. Neurol. Neurosurg. Psychiatry **57**, 190–197 (1994)] reported a new method which reliably demonstrated a short latency vestibulocollic reflex evoked by clicks and recorded from sternocleidomastoid EMG. Consistent with previous evidence, the saccule was thought to be the most likely receptor to mediate the response, a conclusion for which there has been further experimental support. Bone-conducted acoustic stimulation can also evoke a similar reflex and, although saccular receptors are probably excited, this may not be the sole vestibular receptor activated under these conditions. The vestibular apparatus is relatively more sensitive to bone-conducted sound than to air-conducted sound, when compared to the cochlea. [Work supported by NH&MRC Australia.]

9:40

5aABa5. Central connections of anamniote auditory otolith endorgans. Catherine A. McCormick (Dept. Biol. and Dept. Neurosci., Oberlin College, Oberlin, OH, catherine.mccormick@oberlin.edu)

The ascending auditory pathways of anamniotes (fish and amphibians) are organized similarly to those of amniotes (reptiles, birds, and mammals). A common pattern for the flow of acoustic information—inner ear to first-order nuclei to higher-order auditory nuclei in the medulla, midbrain, thalamus, and cerebrum—probably evolved early in vertebrate history and was retained in extant fish and in terrestrial vertebrates (with taxon-specific modifications) despite variability in inner-ear acoustic receptors. Otolith endorgans subserve hearing in jawed fish; auditory fibers segregate from otolith endorgan vestibular fibers and supply auditory divisions of functionally mixed (auditory-vestibular) first-order nuclei. Amphibian auditory otolith endorgans are primarily seismic detectors complementing new sound-pressure receptors of the amphibian and basilar papillae. Otolithic and papillar auditory endorgans supply common areas within functionally mixed (auditory-vestibular) first-order nuclei. Uniquely in anurans, they also supply a discrete first-order auditory nucleus analogous to the amniote cochlear nuclei. Discrete first-order auditory nuclei are correlated with the evolution of a tympanic ear, an independently evolved condition in anurans and among amniotes. Thus, at the first-order level, the amphibian auditory pathway retains primitive organizational features present in fish auditory pathways. Convergent neuroanatomical specializations in anurans and amniotes may reflect common functional requirements.

10:05–10:20 Break

10:20

5aABa6. Evidence for near-field hearing in crocodilian vocal communication: Intensity of the American alligator (*Alligator mississippiensis*) vocal display. Neil P. McAngus Todd (Faculty of Life Sci., Univ. of Manchester, M60 1QD, UK, neil.todd@manchester.ac.uk)

The aim of this study was to carry out an acoustic analysis of alligator vocal displays recorded simultaneously in air and water in order to estimate the role of the sacculus in crocodilian acoustic communication. Crocodilians are interesting animals to study as they are aquatic, highly vocal, and morphologically have a large sacculus. Alligators, in particular, are known to be one of the most vocal species. Previous studies of alligator hearing in air and water have indicated a maximum sensitivity between 800 and 1000 Hz [Higgs *et al.*, J. Comp. Physiol. **188**, 217–223 (2002)]. Analysis of alligator vocalizations, however, indicates that most of the power in adult vocalizations in water is typically between 30–50 Hz. These results leave the very real possibility that the basilar papilla may not be the primary receptor during crocodilian water-borne communication. At 1-m distance peak SPLs of 140 dB can be obtained in water and, in conjunction with existing threshold data, the effective radius of saccular acoustic sensing of the above source can be estimated to be approximately 30 m, which is easily greater than the typical distance between performer and receiver animals during a display.

10:45

5aABa7. Saccular hearing in anurans. Seth S. Horowitz and Andrea M. Simmons (Dept Psych. & Neurosci., Brown Univ., Providence RI 02912)

Acoustic sensing in anurans is mediated by several inner ear organs, including the amphibian papilla, basillar papilla, and lagena. The contribution of the saccule to hearing in adult ranids tends to be constrained to vibration sensitivity and gravity sensing, although there has been some evidence for low-frequency hearing sensitivity as well. The organs that mediate hearing in larval ranids are less well understood. Several studies have postulated a strong role for saccular-mediated hearing in tadpoles, operating by the sensing of particle motion mediated by direct kinetic effects on otolithic organs (the fenestral pathway). Physiological studies show that, in tadpoles, the medial and lateral vestibular nuclei are sensitive to particle motion stimulation, while the dorsal medullary nucleus is more sensitive to pressure stimulation. Anatomical tract tracing with lipophilic cyanocarbanine dyes placed in the saccular branch of the eighth nerve show projections to both auditory and vestibular medullary nuclei, suggesting that input from the saccule may play a role in both particle and pressure sensation.

11:10

5aABa8. The sense of hearing mediated by the saccule in goldfish. Richard R. Fay (Dept. Psychol. and Parmly Hearing Inst., Loyola Univ. Chicago, 6525 N. Sheridan Rd., Chicago, IL 60626)

The saccule mediates the sense of hearing for many fish species, goldfish (*Carassius auratus*) in particular. Motions of the swim bladder in response to pressure fluctuations impinge on the saccule of each ear, where auditory-nerve fibers project to medullar, midbrain, and forebrain nuclei in a circuit analogous to the ascending auditory system of tetrapods. It has been repeatedly shown over the past 40 years that the sense of hearing mediated by the saccule has many fundamental features in common with the sense of hearing of tetrapods. Detection and discrimination thresholds for goldfish are in the vertebrate range quantitatively, and are of the same sorts of simple and complex discriminations that mammals and other tetrapods can perform. Furthermore, goldfish behave as if the perceptual dimensions of spectral pitch, complex musical pitch, timbre, and roughness exist for them. Although goldfish do not vocalize, many fish species that hear using saccular input do vocalize (e.g., mormyrids), and have special adaptations for responding to vocalizations, like many tetrapods. Finally, goldfish have been shown capable of simultaneous stream segregation based on the same principles known for humans and other species. Thus, otolith organs can participate in a sense of hearing that is typical for vertebrates.

Contributed Papers

11:35

5aABa9. Are acoustically induced flows relevant in fish hearing? Charlotte W. Kotas, Peter H. Rogers, and Minami Yoda (Mech. Eng. Dept., Georgia Inst. of Technol., Atlanta, GA 30332-0405, charlotte.kotas@gatech.edu)

The fluid-filled fish ear contains an irregularly shaped, dense, bony otolith that influences the acoustically induced motion of the adjacent hair cells and thus what fish "hear." Because incident sound causes the otolith to oscillate with respect to its surroundings, acoustically induced fluid flows are generated near the otolith. A series of experiments was performed using oscillating spheroids as model otoliths to study the induced steady streaming and periodic flows. The studies focused on how the flows change with spheroid (or otolith) geometry and orientation, as well as oscillation (or sound) frequency ω and amplitude s. Experiments were conducted for Reynolds numbers $\mathrm{Re} \equiv \omega L^2/\nu < 10^2$ and amplitude ratios $\epsilon \equiv s/L = 0.05-0.2$, where L is the product of the spheroid aspect ratio and equivalent radius. Results will be presented for oblate and prolate spheroids oscillating at angles up to 45° relative to the spheroid's axis of symmetry and for flows induced by spheroids driven simultaneously at different frequencies. Understanding these flows may give insight into how fish localize sound sources and lead to new acoustic sensor designs. [Work supported by ONR.]

11:50

5aABa10. Saccular hearing; turtle model for a human prosthesis. Martin L. Lenhardt (Dept. of Biomed. Eng., Otolaryngol., and Emergency Medicine, Virginia Commonwealth Univ., Box 980168 MCV, Richmond, VA 23298-0168)

The saccule is a hearing organ is some vertebrates thought to be responsive to substrate vibration (bone conduction) or low-frequency aerial sound. There was likely some overlap in these functions in the course of evolution after the sensory area to become the cochlea migrated from the saccule. That overlap is preserved in extant turtles by columella (stapes) saccular coupling via fibroelastic strands; thus both organs can respond to air conduction and bone conduction stimulation. Evoked potential data, however, reflect differential AC/BC drive to the inner ear. The columellas inertia provides the force to displace the saccular wall with the tympanum

providing damping. Intense AC stimulation likely stimulates the human saccule. A proprietary stapedial saccular strut is described that serves as a surgically implanted coupling device for humans, allowing more efficient use of AC saccular hearing in clinical deafness. The human saccular resonance is about 350 Hz, which should allow for sufficient speech coding for intelligibility assuming connectivity to the auditory neuraxis. BC stimulation through audible ultrasound also likely activates the saccule in individuals with profound deafness. A stand-alone stapedio-saccular strut or one used in combination with an ultrasonic hearing aid offers the potential of communication through saccular hearing.

12:05–12:35
Panel Discussion

FRIDAY MORNING, 9 JUNE 2006 ROOM 550AB, 8:00 TO 10:00 A.M.

Session 5aABb

Animal Bioacoustics: Classification and Parameter Estimation

John R. Buck, Chair
Univ. of Massachusetts, Dartmouth, SMAST, 706 S. Rodney French Blvd., New Bedford, MA 02744-1211

Contributed Papers

8:00

5aABb1. Online acoustic parameter extraction and annotation for a marine sound collection. Shelagh A. Smith and Jack W. Bradbury (Macaulay Library, Cornell Lab of Ornithology, Ithaca, NY 14850)

After 3 years of archival and with the collaboration of more than 20 researchers and institutions, the online marine sound collection at the Macaulay Library, Cornell Lab of Ornithology, is now in a proof of concept phase. This presentation introduces the new annotation and acoustic parameter extraction and retrieval tools. It is further explained how these tools are used for fine-grain searches and recovery of specific sounds within long recordings while online. Also included is a description of the online open access model and delivery mechanism. [Work supported by ONR and NSDL.]

8:15

5aABb2. Dynamic time warping for automatic classification of killer whale vocalizations. Judith C. Brown (Phys. Dept., Wellesley College, Wellesley, MA 02481 and Media Lab., MIT, Cambridge, MA 02139) and Patrick J. O. Miller (Univ. of St. Andrews, St. Andrews, Fife KY16 9QQ UK)

A large number of sounds from the captive killer whale population at Marineland of Antibes, France have been classified perceptually into call types. [A. Hodgins-Davis, thesis, Wellesley College (2004)]. The repetition rate of the pulsed component of five or more examples of each call type was calculated to give a set of melodic contours. These were compared pairwise using dynamic time warping to give a dissimilarity or distance matrix. The distances were then transformed into a component space using multidimensional scaling, and the resulting points clustered with a k-means algorithm. In grouping 57 sounds into 9 call types, there was a single discrepancy between the perceptual and automated methods. Preliminary measurements on a second set of 74 sounds from northern resident killer whales have been made. Both the high-frequency and low-frequency contours have been measured and classified perceptually into 7 groups. Errors for the automatic classification of this group are over 10%.
[Sound recording supported by grants from WHOI's Ocean Life Institute and the Royal Society to Patrick J. O. Miller.]

8:30

5aABb3. Probability distributions for locations of calling animals, receivers, sound speeds, winds, and data from travel time differences. John Spiesberger (Dept. of Earth and Environ. Sci., Univ. of Pennsylvania, 240 S. 33rd St., Philadelphia, PA 19104-6316, johnsr@sas.upenn.edu)

A new, nonlinear sequential Monte Carlo technique is used to estimate *posterior* probability distributions for the location of a calling animal, the locations of acoustic receivers, sound speeds, winds, and the differences in sonic travel time between pairs of receivers from measurements of those differences, while adopting realistic *prior* distributions of the variables. Other algorithms in the literature appear to be too inefficient to yield distributions for this large number of variables (up to 41) without recourse to a linear approximation. The new technique overcomes the computational inefficiency of other algorithms because it does not sequentially propagate the joint probability distribution of the variables between adjacent data. Instead, the lower and upper bounds of the distributions are propagated. The technique is applied to commonly encountered problems that were previously intractable, such as estimating how accurately sound speed and poorly known initial locations of receivers can be estimated from the differences in sonic travel time from calling animals, while explicitly modeling distributions of *all* the variables in the problem. In both cases, the new technique yields one or two orders of magnitude improvement compared with initial uncertainties. The technique is suitable for accurately estimating receiver locations from animal calls.

8:45

5aABb4. Automatic call recognition of cricket songs using image processing. T. Scott Brandes (Tropical Ecology Assessment and Monitoring (TEAM) Initiative, Conservation Intl., 1919 M St., NW, Washington, DC 20036), Piotr Naskrecki (Harvard Univ., Cambridge, MA 02138), and Harold K. Figueroa (Cornell Lab. of Ornithology, Ithaca, NY 14850)

An automatic call recognition (ACR) process is described that uses image processing techniques on spectrogram images to detect constant-frequency cricket calls recorded amidst a background of evening sounds found in a lowland Costa Rican rainforest. This process involves using image blur filters along with binary filters to isolate calling events. The binary filters are used to isolate potential calls from background noise, and the blur filters are used to unite discrete call fragments as a single continuous call. Features of these events, notably the events central frequency, duration, and bandwidth, along with the type of blur filter applied, are used with a Bayesian classifier to make identifications of the different calls. Of the 22 distinct sonotypes (calls presumed to be species specific)

recorded in the study site, 17 of them are recorded in high enough numbers to both train and test the classifier. The classifier approaches 100% true-positive accuracy for these 17 sonotypes. The high true-positive accuracy of this process enables its use for monitoring singing crickets in tropical forests.

9:00

5aABb5. Feature normalization for robust individual identification of the ortolan bunting (*emberiza hortulana L*). Kuntoro Adi and Michael T. Johnson (Dept. of Elec. and Comput. Eng., Marquette Univ., Milwaukee, WI 53201, kuntoro.adi@marquette.edu)

This paper presents the use of cepstral moment normalization for the purpose of improving the accuracy of individual identification in the ortolan bunting (*emberiza hortulana L*). The underlying classification algorithm is based on hidden markov models (HMMs) using Greenwood function cepstral coefficients (GFCCs), and the proposed feature normalization is implemented via a combination of cepstral mean subtraction and cepstral variance normalization, which are equivalent to a statistical z-score normalization across each example utterance. Results indicate a significant reduction in error on the individual identification task, improving accuracy from 89% to 96% in initial experiments, using a single call type over a small population of 6 individuals.

9:15

5aABb6. Characteristics of nonlinear phenomena in the tonal vocalizations of a North American canid, *Canis rufus*. Jennifer N. Schneider (Dept. of Psych., Univ. of Buffalo, Park Hall Rm. 206, Buffalo, NY, 14260), Rita E. Anderson, and Edward H. Miller (Memorial Univ. of NF, St. Johns, NF, A1B 3X9 Canada)

This study examines the structure and frequency of occurrence of nonlinear phenomena found in tonal vocalizations produced by red wolves. Spectrograms were obtained from audio tracks of digital video recordings of captive wolves from a breeding facility in Graham, WA. Tonal vocalizations were determined to be composed of 1–30 sound units, arranged in 1–5 phrases. Linear units included squeaks (2600–9600 Hz) and wuhs (100–1600 Hz); nonlinear units accounted for 22% of sounds and included between-type frequency jumps, harmonic and pure-tone biphonations, squeaks with sidebands, and squeak jumps. Five tonal vocalization types were identified based on unit composition: squeaks (48.4%), wuhs (19.3%), and three mixed vocalizations: banded squeaks (13.2%), complex squeaks (6.4%), and squeak-wuhs (12.2%). Unit order within a squeak-wuh vocalization was not random; transitions between units following a structural gradient most likely begin with higher frequency units and end with mixed or lower frequency units. The production of nonlinear sounds varied within and between individuals. The linear and nonlinear structure of red wolf tonal vocalizations are similar to that which has been reported in dholes and African wild dogs, and which have been indicated in reviews of published sonograms of gray wolf vocalizations.

9:30

5aABb7. Equine acoustics: Further analysis of a whinny—The expressive component. 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)

A horse's whinny appears to be a very interesting vocalization. Initial acoustic analysis suggested two principal components, one ''emotional'' and the other ''expressive.'' The first seems to follow Morton's criterion: increasing frequency with emotional state. The second component, since many suggest that a whinny is not a threatening gesture, has the potential for a number of specific expressions for such situations as ''calling'' or ''greeting.'' This paper describes the present indications as the data base slowly increases.

9:45

5aABb8. Discovery of Sound in the Sea website: An educational resource. Kathleen J. Vigness Raposa (Marine Acoust., Inc., 809 Aquidneck Ave., Middletown, RI 02842, kvigness@aol.com), Gail Scowcroft, Jill Johnen, Chris Knowlton (Univ. of Rhode Island, Narragansett, RI 02882), and Peter F. Worcester (Univ. of California—San Diego, La Jolla, CA 92093-0225)

The scientific community and the public have become increasingly aware of, and concerned about, underwater sound. There is increasing interest in learning about sources and uses of sound, and potential effects of sound on the marine environment. The Discovery of Sound in the Sea website (http://www.dosits.org) provides scientific information for the general public and K–16 educators and students. It also includes advanced level content appropriate for high school physics or undergraduate classes. The website has three major sections, (1) Science of Sound in the Sea, (2) People and Sound in the Sea, and (3) Animals and Sound in the Sea, introducing the physical science of underwater sound and how people and animals use sound to accomplish various tasks. The Animals and Sound section also summarizes the current state of knowledge of the effects of underwater sound on marine mammals and fishes. Three galleries contain a collection of underwater sounds (Audio Gallery), video interviews with researchers (Scientist Gallery), and descriptions of acoustic equipment (Technology Gallery). Additions to the website are ongoing. Scientists with recordings and/or equipment that have not been included are encouraged to contact the authors. A limited number of free CDs of the website will be available.

FRIDAY MORNING, 9 JUNE 2006 ROOM 550AB, 10:15 A.M. TO 12:00 NOON

Session 5aABc

Animal Bioacoustics: Acoustic Sensing of Animals

Jennifer L. Miksis-Olds, Chair

Univ. of Rhode Island, Graduate School of Ocean Biology, Narragansett, RI 02882

10:15

5aABc1. Tracking fish motion and behavior with acoustic flow. Srinivasan Jagannathan, Deanelle Symonds, Sunwoong Lee, Nicholas C. Makris (MIT, Cambridge, MA 02139), Ninos Donabed, and Purnima Ratilal (Northeastern Univ., Boston, MA 02115)

Fish populations in the New Jersey continental shelf have been continuously monitored over thousands of square kilometers using acoustic waveguide remote sensing techniques [Makris *et al.*, ''Fish population and behavior revealed by instantaneous continental-shelf-scale imaging,'' Science **311**, 660–663 (2006)]. Here, we explore the approaches for tracking both large shoals and smaller isolated fish groups over space and time from images of fish population density. One method uses density thresholds to segment coherent population centers and track their centroids. These data are then used to compute differential velocities between population centers and to determine the rate at which information may be transmitted across a population center. We also examine the possibility of applying optical flow concepts from machine vision [Horn, B. *Robot Vi-*

5a FRI. AM

sion, MIT Press, 1986], as a global approach for estimating fish velocity at every pixel in the fish density image. The analogous acoustic flow is computed based on changes in acoustic intensity over space and time. Acoustic flow automatically tracks fish motion within a school or shoal and migration across fish bridges connecting shoals segments. We also investigate the possibility of using acoustic flow for navigating the ocean environment and correcting for errors in charting acoustic returns, for instance caused by array-heading sensor.

10:30

5aABc2. The roles of density, geometry, and short-range positional order in low-frequency acoustic scattering from fish schools. Thomas R. Hahn and Tanos Elfouhaily (RSMAS, AMP, Univ. of Miami, 4600 Rickenbacker Cswy., Miami, FL 33149)

Fish can be efficient scatterers of low-frequency underwater sound. This results from the large response of their swim bladders to incoming acoustic waves at frequencies close to their natural resonance frequencies. For large and densely packed fish schools the scattering returns deviate significantly from what could be expected by adding the individual fish returns. These deviations are caused by multiple scattering between the individual fish in the school and concern the target strength as well as the spectral and directional characteristics of the scattering cross sections. Acoustically, a fish school can be viewed as a single object with acoustic bulk parameters determined from the properties of the individual fish and their positional arrangement in the school. This effective medium approach incorporates the acoustic coupling of all fish due to multiple scattering and works for a wide range of fish sizes, -numbers, and -densities. In this study, the effective medium approach as well as numerical computations from first physical principles are employed to investigate the coherent and diffusive low-frequency scattering cross sections of large and dense fish schools as a function of fish density, school geometry, and positional order among the individual fish. Inversions for these parameters are explored.

10:45

5aABc3. Assessing fish populations and enhancing biological detection in continental shelf environments with range-dependent scattering and reverberation models. Mark Andrews, Ameya Galinde, Ninos Donabed, Purnima Ratilal (Northeastern Univ., 360 Huntington Ave, Boston, MA 02115), Tianrun Chen, Deanelle Symonds, Sunwoong Lee, and Nick Makris (MIT, Cambridge, MA 02139)

Fish populations in continental shelf environments can be continuously imaged over thousands of square kilometers using acoustic waveguide remote sensing techniques [Makris *et al.*, Science, Feb. (2006)]. A calibrated range-dependent scattering and reverberation model [Ratilal *et al.*, J. Acoust. Soc. Am. **114**, 2302 (2003)] based on the parabolic equation has been applied to assess population densities of fish by inverting long-range acoustic data collected on the New Jersey continental shelf. This model is now applied to predict the types of fish species and zooplankton that are detectable in a general range-dependent continental shelf environment, including the resolution and accuracy that can be expected in estimating fish population densities and for differentiating fish species. We consider different geometries of the source and receiving array to enhance biological detection and reduce background reverberation in highly range-dependent environments. Using multiple source frequencies, the possibility of distinguishing fish species based on their differing scattering characteristics and resonance frequencies will be examined.

11:00

5aABc4. Integrated marine mammal monitoring and protection system: Gray whale target strength measurements. Irena Lucifredi, Amy Vandiver, and Peter Stein (Scientific Solutions Inc., 99 Perimeter Rd., Nashua, NH 03063)

With a broader objective of establishing a global solution for systematic assessment and protection of the marine environment, one of the central objectives of this work is to distinguish whale backscattered responses from the ones generated by the environmental clutter in a waveguide. Furthermore, it aims to identify and analyze the target signature features that are necessary for enhanced active sonar detection and classification of marine mammals. Over the years there have been very few documented attempts to capture and analyze the backscattering response of whales using an active sonar system. Nevertheless, whales, mostly owing to their size, their motion, and the aspect dependence of their backscattered field, possess desirable properties that help distinguish their scattered response from clutter and other environment related false alarms. As an initial step, data collected during the Marine Mammal Active Sonar Test (MAST04) is presented, and long-range gray whale target strength measurements are obtained. Results are compared to the previously published whale target strengths. Additionally, an investigation is conducted in an effort to provide whale feature identification points suitable for automated detection and classification, as a means of relating gray whale active acoustic signatures to their inherent characteristics and motion. [Work supported by ONR.]

11:15

5aABc5. Initial application of spectrogram-based cross correlation to the localization of Orcas. Martin C. Renken (Naval Undersea Warfare Ctr. Div. Keyport, 610 Dowell St., Keyport, WA 98345-7610, renkenmc@kpt.nuwc.navy.mil)

Localization and tracking of sperm whales (*Physeter macrocephalus*) and pilot whales (*Globicephala macrorhynchus*) has been demonstrated using the widely spaced bottom-mounted hydrophone arrays located at the Atlantic Undersea Testing and Evaluation Center (AUTEC), Andros Island, Bahamas [Moretti *et al.*, US Navy J. Und. Acoustics **52**, 651–668 (2002)]. The marine mammal monitoring utilizes a cross correlation of the spectrogram calculated from acoustic calls received at each hydrophone. The spectrogram information is combined with time of arrival pattern estimates to identify specific animals in order to estimate the animal's position through triangulation. This paper explores the initial application of these techniques to the wide variety of acoustic vocalizations recorded from transient killer whales (*Orcinus Orca*) in Dabob Bay of Washington state's inland waters in February of 2005. The recordings were made from a bottom-mounted linear array of hydrophones during a period of low acoustic interference. While this recording was obtained by Naval Undersea Warfare Center Keyport, active operations are not conducted in the presence of whales according to standard operating policy. Monitoring in the Dabob Bay Naval Operating Area is carried out both visually and acoustically by range operators trained in marine mammal identification by NOAA.

11:30

5aABc6. Modeling of acoustically enhanced bubble growth in supersaturated biological media. Yurii A. Ilinskii, Mark F. Hamilton, and Preston S. Wilson (Appl. Res. Labs., Univ. of Texas at Austin, P.O. Box 8029, Austin, TX 78713-8029)

Recent measurements and predictions of dissolved nitrogen concentration in tissues of dolphins and whales following repeated dives have indicated that the gas concentration may reach 300% of saturation [Houser *et al.*, "Can diving-induced tissue nitrogen supersaturation increase the chance of acoustically driven bubble growth in marine mammals?," J. Theor. Biol. **213**, 183–195 (2001)]. This has generated renewed interest in whether underwater sound may increase the likelihood of decompression sickness in these cetaceans. One potential mechanism is bubble growth due to rectified diffusion. Eller and Flynn ["Rectified diffusion during nonlinear pulsations of cavitation bubbles," J. Acoust. Soc. Am. **37**, 493–503 (1965)] modeled rectified diffusion by solving a Rayleigh-Plesset equation for the acoustically driven bubble dynamics, coupled to a quasi-analytical diffusion model that accounts for increase of the equilibrium bubble radius. Crum and Mao ["Acoustically enhanced bubble growth at low frequencies and its implications for human diver and marine mammal," J. Acoust. Soc. Am. **99**, 2898–2907 (1996)] used similar models to consider dissolved gas saturations up to 223% and demonstrated significantly enhanced bubble growth at sound-pressure levels above 210 dB (re: 1 μPa). We are revisiting this problem by extending these models and considering wider ranges of parameters. A principal difference is that the gas diffusion is modeled by solving the full partial differential equation

numerically. Not only does this approach circumvent analytical approximations, it permits taking into account variations in gas concentration associated with dive-profile-dependent hydrostatic pressure variations. [Work supported by ONR.]

11:45

5aABc7. Physics-based volume clutter from GeoClutter biological distributions. Adam Frankel (Marine Acoustics Inc., 706 Giddings Ave., Ste. 1C, Annapolis, MD 21401), Richard H. Love (12315 Fairway Meadows Dr., Fort Worth, TX 76179), Charles Monjo, Bruce Newhall, Juan I. Arvelo, Jr. (Johns Hopkins Univ., Laurel, MD 20723-6099), and William T. Ellison (Marine Acoustics Inc., Litchfield, CT 06759)

Empirical fish species composition and distribution data obtained during a GeoClutter survey were input into scattering models for marine mammals [Love, J. Acoust. Soc. Am. **49**, 816–823 (1971)] and for schools of fish [Feuillade *et al.*, J. Acoust. Soc. Am. 99, 196–208 (1996)]. These scattering models drew additional inputs from a database of biological parameters to generate volume clutter simulations. Model inputs included distribution of fish size, school population size, depth, orientation, and location. The volume clutter model treated each fish school and individual marine mammal as discrete acoustic targets. Motion of the dominant marine organisms (in addition to source and receiver motion) was taken into account to observe clutter map variability during a series of pings. The volume clutter distribution from this physics-based approach was compared against bottom clutter and against measured clutter-rich reverberation from the GeoClutter experiment. [The Office of Naval Research (ONR) is sponsoring this effort.]

FRIDAY MORNING, 9 JUNE 2006 ROOM 553AB, 9:00 TO 11:30 A.M.

Session 5aBB

Biomedical Ultrasound/Bioresponse to Vibration: Targeted Contrast Agents

Tyrone M. Porter, Chair
Univ. of Cincinnati, Biomedical Engineering, Medical Sciences, 231 Albert Sabin Way, Cincinnati, OH 45267-0586

Invited Papers

9:00

5aBB1. Treatment of ischemic stroke with nanobubbles and ultrasound. Evan C. Unger (ImaRx Therapeutics, Inc., 1635 E. 18th St., Tucson, AZ 85719)

The use of perfluorocarbon-filled microbubbles for ultrasound contrast imaging is well known. Now, the next generation of perfluorocarbon-filled bubbles has reached nanoscale and will be evaluated in clinical trials for the dissolution of blood clots in ischemic stroke and other vascular occlusions. MRX-815 nanobubbles are submicron, lipid coated, perfluorocarbon nanobubbles. Due to their small size they are able to penetrate a thrombus. Since they share a lineage with ultrasound contrast agents, their progress can be monitored on diagnostic ultrasound and once they have reached the site of thrombosis ultrasound can be applied externally to activate the bubbles. This activation consists of the bubbles oscillating in the field of ultrasound and cavitating. This mechanical energy works to dissolve the clot and restore blood flow to the ischemic tissues beyond. This process, SonoLysis® therapy, can be used in tandem with lytic agents or can be used alone in cases where lytic agents are contraindicated. A multicenter, Phase II clinical trial is currently evaluating SonoLysisregistered in the treatment of acute ischemic stroke. Preliminary findings have been encouraging and additional studies are planned.

9:30

5aBB2. Echogenic targeted liposomes for transfection and drug delivery. David D. McPherson (Northwestern Univ., 251 East Huron, Galter 8-230, Chicago, IL 60611, d-mcpherson@northwestern.edu)

We have developed targeted, echogenic immunoliposomes. These agents are lipid based, can be targeted to cells, and have encapsulated air bubbles that render them echogenic, and the air bubbles can also be used as cavitation agents. They have the ability to encapsulate drugs and genes. We have developed stable formulations and optimized attachment of antibodies and peptides for targeted molecular imaging using ultrasound. We are determining the pharmacokinetics of drug and gene entrapment and the ultrasonic destruction threshold for therapeutic delivery. With the enclosed cavitation agent, therapeutic ultrasound (sonoporation) potentiates cellular drug and gene delivery. This presentation will focus on optimizing the liposomes for stability, targeting, drug entrapment and release, destruction threshold, and the additive sonoporation effect of the encapsulated cavitation nuclei. Tissue plasminogen activator and papaverine are prototype drugs and beta-galactosidase is the prototype gene for encapsulation. The objective of this work is to use modifications of our echogenic immunoliposomes to better entrap and release drugs and genes, while utilizing the enclosed cavitation nuclei, which allows ultrasound to not only release the drug/gene, but also improve its delivery.

10:00–10:15 Break

10:15

5aBB3. Gas-filled microbubbles—Contrast agents for targeted molecular imaging. Alexander L. Klibanov, John Hossack, Yun K. Cho, William C. Yang, Joshua J. Rychak, and Klaus F. Ley (Univ. of Virginia, Charlottesville, VA 22908)

Microbubble contrast agents suitable for targeted molecular imaging have been described. Commonly used agents are gas-filled micron-size bubbles. Targeting ligands (antibodies, peptides, or proteoglycans) are coupled to the contrast particles and ensure firm attachment of microbubbles to intravascular regions of interest. Targeting is directed to selectins or integrins upregulated on vascular endothelium in the areas of pathology (inflammation, ischemia-reperfusion injury, or angiogenesis). Real-time imaging has been performed in animal models with sub-millimeter spatial resolution. Detection sensitivity for targeted microbubble contrast is superb: single bubbles (pg mass) can be visualized by ultrasound. Successful retention of microbubbles on the target is dependent on the target receptor surface density as well as targeting ligand surface density on the bubbles. To improve targeting efficacy, microbubbles can be outfitted with fast-binding ligands (such as P-selectin glycoprotein ligand-1 analogs) in addition to slow but firmly binding ligands (such as antibodies). Ultrasound irradiation also improves efficacy of targeting, especially in high shear conditions. In addition to imaging, microbubbles can be applied for drug and gene delivery purposes: interaction of ultrasound and bubbles can be used for in situ targeted deposition, activation and release of pharmaceutical agents [Disclosures: ALK, JJR, and KFL: Targeson stock ownership. ALK: Philips Research grant.]

Contributed Papers

10:45

5aBB4. Nonlinear oscillation of microbubble in microtubes for ultrasound imaging and drug delivery. Shengping Qin and Katherine Ferrara (Dept. of Biomed. Eng., Univ. of California, 451 East Health Sci. Dr., Davis, CA 95616)

Ultrasound-assisted microbubble drug delivery is under intensive investigation due to the potential for enhanced local delivery without systemic adverse effects. The efforts in modeling bubble oscillation have been largely focused on using various modified forms of the Rayleigh-Plesset bubble dynamics equation, of which the cornerstone assumption is that a bubble oscillates in an unbounded liquid field and remains spherical until it collapses. Here, nonlinear oscillation of a bubble confined inside a microtube with diameters comparable with capillaries is studied numerically and experimentally. In numerical simulation, the liquid is treated as incompressible viscid fluid and contrast agent is treated as gas bubble obeying the polytropic law. Microbubble oscillation is shown to decrease in a small vessel as compared with oscillation in an infinite fluid or a larger vessel diameter. For the same bubble size, the natural oscillation frequency of a bubble deceases with the decrease of tube size. The natural frequency of a bubble with an equilibrium diameter of 2.5 μm decreases from 3.56 MHz, in unbounded liquid field, to 1.16 MHz in an 8-μm tube. Decreasing the ultrasound center frequency can increase the amplitude of bubble oscillation and thereby produce a higher pressure on the tube.

11:00

5aBB5. *In vitro* and *in vivo* high-frequency ultrasound response of microbubbles. Orlando Aristizábal, Daniel H. Turnbull (Skirball Inst. of Biomolecular Medicine, New York Univ. School of Medicine, 540 First Ave., New York, NY 10016, oarist@saturn.med.nyu.edu), and Jeffrey A. Ketterling (Riverside Res. Inst., New York, NY 10038)

Recently, there has been a growing interest in utilizing contrast agents with high-frequency ultrasound (HFU $>$ 20 MHz). As a first step to using contrast agents *in vivo* with HFU, the acoustic response of several agents was examined using a 40-MHz ultrasound biomicroscope (UBM, Vevo 770, VisualSonics). Optison, Definity, and near-micron and submicron agents from Point Biomedical were investigated. A small volume of the agents was injected into a petri dish of saline solution, and radio-frequency (rf) backscatter data from the UBM were digitized with a digital oscilloscope. Agents were also injected into the cardiovascular system of mouse embryos and rf data were acquired *in vivo*. The echo signals from individual bubbles were windowed in the rf data and the spectra were calculated. The resulting spectra revealed an increasing harmonic content for each of the agents as the acoustic drive amplitude and number of excitation wavelengths increased. Harmonic generation appeared to be due mainly to nonlinear propagation in the water bath and not from a nonlinear response of the contrast agent because reflected spectra without contrast agent showed similar harmonics. Using a modified parameter imaging method, fundamental (40 MHz) and first-harmonic (80 MHz) images were generated.

11:15

5aBB6. Molecular imaging and therapy with perfluorocarbon-based site targeted contrast nanoparticles. Kirk D. Wallace, Michael S. Hughes, Jon N. Marsh, Kathryn C. Crowder, Huiying Zhang, Samuel A. Wickline, and Gregory M. Lanza (Dept. of Medicine, Washington Univ. St. Louis, MO 63130, kirk.wallace@wustl.edu)

Advances in molecular biology and cellular biochemistry are providing new opportunities for diagnostic medical imaging to "see" beyond the anatomical manifestations of disease to the earliest biochemical signatures of disease. The emerging field of "molecular imaging" encompasses the noninvasive *in vivo* diagnosis of complex pathological processes by detection of unique molecular signatures. Localization of specific biochemical epitopes with targeted contrast agents affords the unique opportunity for both targeted delivery and deposition of therapeutics. Liquid perfluorooctylbromide nanoparticles (approximately 250 nm mean diameter) elicit little acoustic reflectivity in suspension and strong acoustic contrast when bound and concentrated on a surface (e.g., fibrin deposits in a thrombus). This nanoparticle platform may be functionalized with homing ligands, like anti-$\alpha v \beta 3$ integrins, and therapeutic agents. Acoustic imaging of densely distributed biomarkers (e.g., fibrin epitopes) is readily accommodated with fundamental imaging, but for sparse biomarkers (e.g., integrins) we have developed and implemented complementary, nonlinear imaging techniques based upon information-theoretic receivers. Moreover, perfluorocarbon nanoparticles also offer targeted drug delivery, based on contact facilitated lipid exchange, which can be further augmented by ultrasonic insonification.

Session 5aMU

Musical Acoustics, Engineering Acoustics, and Signal Processing in Acoustics: Human-Computer Interfaces

Jonas Braasch, Cochair
Rensselaer Polytechnic Inst., 110 8th Street, Troy, NY 12180

William L. Martens, Cochair
McGill Univ., Faculty of Music, 555 Sherbrooke, St., West, Montreal, PQ, H3A 1E3, Canada

Invited Papers

8:00

5aMU1. Musical coffee mugs, singing machines, and laptop orchestras: New paradigms for musical expression. Perry Cook (Comput. Sci., Princeton Univ., 35 Olden St., Princeton, NJ 08544)

Advances in algorithms, hardware, and sensors now allow us to build new types of expressive musical devices based around small computer systems. These new ''instruments'' can leverage and extend the expertise of virtuoso performers, expand the palette of sounds available to composers, and encourage new ideas and participation from the young or the musically untrained. New musical instuments can take the form of augmentations of traditional musical instruments, whimsical objects made by augmenting everyday nonmusical objects, or entirely new paradigms for making and controlling sound. Wired and wireless networking can provide opportunities, yet pose significant challenges, for new forms of ''orchestral'' expression. This talk will look at a variety of musical technologies, devices, systems, compositions, and performances we have created at Princeton in the last 10 years or so. Lots of audio/video examples, as well as actual live demonstratons, will be presented.

8:30

5aMU2. Mapping strategies for sound synthesis, digital audio effects, and sonification of performer gestures. Vincent Verfaille and Marcelo M. Wanderley (SPCL & IDMIL, Music Technol. Area, McGill Univ., 555 Sherbrooke St. West, Montreal, Canada, H3A 1E3)

Mapping strategies are an essential step when designing realtime musical performance systems, as well as offline digital sound processing. These strategies define how we relate input device parameters to sound synthesis or audio effect parameters. This implies the ability to combine input parameters among themselves (parameter combination) and valid control signals in terms of range, variation type, etc. (signal conditioning). Recent works highlighted the interest of multi-layer mapping strategies in the context of digital musical instruments, which can also be applied in the context digital audio effects. In this presentation, three strategies will be discussed in order to illustrate the role of mapping strategies in various contexts. The first example concerns an additive synthesizer called *Ssynth*, a further development of *Escher*, a prototyping system aiming at studying the effect of mapping strategy in instrument design. The second example is a general mapping strategy for digital audio effects, allowing for both adaptive and gestural control. The final example concerns sonification of gestures, used to provide cues about ancillary movements of performers. For each example, mapping strategies will be explained in terms of their structure and functionality. [Work supported by FQRNT and MDEIE PSR-SIIRI (Québec, Canada), CNRS and PACA (France).]

9:00

5aMU3. On the evolution of controllers for sound spatialization. Jonas Braasch (Program in Architectural Acoust., Rensselaer Polytechnic Inst., Troy, NY 12180)

Dynamic changes in spatial sound attributes have played a role in classical Western music for a long time. It is known that choreographic movements of opera singers were sometimes made for acoustic considerations. Probably the first mechanical spatial sound controller is the so-called wind swell that is found in pipe organs. Already in 1712, Renatus Harris mentions how *swelling* enables the player to project the sound of the pipes *ad libitum* to nearby or further distances. With the invention of electroacoustic music, a number of electromechanical devices were developed to control spatial aspects of sound (primarily positioning sound sources in 3-D space). Typical examples are Stockhausen's rotational table (developed in the 1950s) and Manfred Krause's sound mill (1960). In this presentation, the evolutionary steps in the design of spatial sound controllers will be outlined—beginning with early, purely mechanical devices up to recent approaches including the author's participation in the development of a gestural controllable sound system based on virtual microphone control (ViMiC).

9:30

5aMU4. Deriving individualized control over music synthesis via inversion of psychophysical scaling results. William L. Martens (Schulich School of Music, McGill Univ., Montreal, QC, Canada)

Computer-based interactive music generation systems have an advantage over acoustic instruments with respect to the flexibility of the human interface for their control. Whereas human performers must conform to the relatively invariant constraints of acoustic instruments, a computer interface can be rapidly updated with regard to changes in user preferences and performance requirements.

This paper will describe a software environment that executes miniature psychophysical scaling experiments for a single user of an interactive music generation system in order to derive individualized control over music synthesis via inversion of the obtained psychophysical scaling results. A case study will be presented in which the system has been used to control parametric synthesis of musical timbres using an electronic musical keyboard, providing automatic perceptual mapping of synthesis parameters within their musically useful range. More in-depth exploration of the timbral similarities between a user-selected set of synthesis patches resulted in low-dimensional control structures that could be used to organize musical timbres in preparation for composition or performance.

10:00–10:15 Break

Contributed Papers

10:15

5aMU5. The method for capturing tilt signal of musical instruments using light sensor. Moonseok Kim (CCRMA, Dept. of Music, Stanford Univ., Stanford, CA 94305-8180)

Players of musical instruments used to express their intuition and emotion not only by playing style but also the motion of the instrument's body. Furthermore, on the ensemble, this gesture plays a role as a language to communicate to each other. Substituting some players for a computer is one part of the information to make interactions between human players and computer. To get them, tilt sensors which catch accelerations are used in general. They are used to generate underdamped signal and have to apply LPF for acquiring stable responses. This method results in delay and may affect temporal difficulties on ensemble. If the light source of the performance room is directional, the light intensity signal that is caught at photoresistors on two or more points of the instrument's surface can be used to calculate the amount of tilting. The photoresistor senses the light intensity which is determined by the distance from the light source, and differences on each value represent relative angle according to the light source direction.

10:30

5aMU6. Time marching spectral analysis of a swept sine melody model. Rama B. Bhat (Concordia Univ., 1455 de Maisonneuve West, QC H3G 1M8, Canada)

A melody is a progression of musical notes in a scale. Since the notes have identified frequencies, a melody rendering involves a quick sweep of frequencies between any two neighboring notes, and a dwell at the notes. A melody rendering is modeled as a dwell at the notes and a sinusoidal sweep in between. A time marching spectral analysis shows that the dwell time and the rate of sweep between notes have significant influence on the melody. Any two notes form the not-so-high harmonics of a periodic process which may or may not necessarily contain the fundamental and some of the lower harmonics. If a three-note combination of a first, a fifth, and an octave higher notes are taken, as the drone in Indian classical music, they have ratios of 1:3/2:2. The combination is a periodic process with half of the first note frequency as the fundamental which is missing, while the second, third, and fourth harmonics are present. A melodic rendering of seven notes or less may be construed as a periodic process with some of the harmonics varying or dancing with time. Results on a melody in pentatonic scale is presented and discussed.

10:45

5aMU7. Ignoring irrelevant dimension variability in frequency discrimination. Blas Espinoza-Varas (Commun. Sci. and Disord., OU Health Sci. Ctr., Oklahoma City, OK 73190), Hyunsook Jang (Hallym Univ., Chuncheon, South Korea), and Praveen Jajoria (OU Health Sci. Ctr., Oklahoma City, OK 73190)

The ability to ignore variability in irrelevant dimensions while attending to pitch plays a significant role in music perception. In frequency-discrimination tasks, this study examined how this ability depends on four factors: the presentation order and time proximity of target and irrelevant dimensions, the dimension conveying irrelevant variability, and training. Each observation interval displayed pairs consisting of a target (T) followed by a nontarget sinusoid (N) or the reverse (N leading T), with 20–250-ms interstimulus intervals (ISI). Frequency discrimination thresholds (FDTs) were measured in the presence or absence of irrelevant variability in the frequency, duration, or level of either T or N. The ability to ignore irrelevant variability depended on all four factors studied. Irrelevant variability in frequency or level of trailing N tones induced large and persistent elevations in the initial FDTs, especially at short ISIs. Irrelevant variability in T or N duration, T level, or leading N-tone frequency yielded only small and transient FDT elevations. With T-N pairs, irrelevant variability was harder to ignore when it was displayed after, rather than simultaneously with, the target increment. Prolonged training in irrelevant-variability conditions nullified all FDT elevations. The results are interpreted in terms of memory, attention, and discrimination-strategy constraints.

11:00

5aMU8. Application of cochlear analysis techniques to finding percussive events in electro-acoustic music. Anderson Mills (The Univ. of Texas, 702 W 32nd St., Austin, TX 78705, nodog@mail.utexas.edu)

Electro-acoustic music is a style of music for which often the pressure signal itself is the only consistent, objective representaiton of the music. A project has been undertaken at The University of Texas to automatically extract audio properties from recordings of electro-acoustic music, and the first property chosen to be extracted is a measure of the "single damped percussive events" (SPDEs) found in a piece of music. Because electro-acoustic music often contains sounds which are not considered musical in traditional contexts, the analysis of this type of music must begin by approaching the pressure signal without traditional musical foreknowledge, and therefore this analysis is focused on the extraction of any type of percussive event found in the pressure signal. The explored methods of analysis include bandpass filtering, spectrogram, and Patterson's auditory image model (AIM) as front-ends. The results of these different methods are compared. A human experiment to determine the nature of an SPDE and to generate data to use as a measure for the algorithms is also discussed. Self-similarity at different time scales within the measure of SPDEs during a piece of music provides useful information to a music theorist and therefore is the next step in this analysis research.

11:15

5aMU9. Frequency tracking of ecclesiastical Byzantine music frequency intervals. Kyriakos M. Tsiappoutas (Dept. of Psych., Illinois State Univ., Normal, IL 61790, kmtsiap@ilstu.edu), George E. Ioup, and Juliette W. Ioup (Univ. of New Orleans, New Orleans, LA 70148)

Two Byzantine music (BM) pieces performed by two well-established traditional chanters are quantitatively analyzed using modern frequency tracking methods to estimate the tone frequencies of the diatonic scale, one of the three main scales of BM. The two experimental versions of the diatonic scale are compared with each other and with two of many theoretical diatonic scales proposed by traditional BM theorists. This constitutes a measure of how well theoretical scales model performance. Then a direct comparison between Byzantine and common European music scales is attempted. Statistical techniques are employed to reveal any significant differences among the different scales. The attraction effect (AE) or "elk-

sis,'' i.e., when a specific secondary tone intentionally takes on different frequency values (micro-sharpened or micro-flattened) depending on its direction in the musical script, is measured. This is part of a continuing attempt to quantify the AE empirically. Both main and secondary tones are independently examined to assess their reliance on the AE, but only secondary tones are found subject to it. A connection to the psychophysics of pitch perception illustrates whether the differences among scales are distinguishable by the human ear.

11:30

5aMU10. Idiomatic guitar synthesis. Leandro L. Costalonga and Eduardo R. Miranda (Interdisciplinary Ctr. for Comput. Music Res., Univ. of Plymouth, PL4 8AA, UK, leandro.costalonga@plymouth.ac.uk)

The aim of this research is to add intelligence to the interface between systems for computer-aided composition and synthesizers. In this paper we introduce a system that is able to play guitar music on a synthesized guitar idiomatically. There are a great number of sound synthesis techniques and computer-aided composition systems available. However, there is a general lack of intelligent interfacing between these two types of systems. For example, when one auditions the different parts of an orchestral piece, the software treats these materials identically; that is, the violin part is not played violinistically or the clarinet part clarinetistically. Musicians tend to blame the synthesizers for such poor performance quality, but if the synthesizer was controlled idiomatically the sound synthesis technique itself would be just one of the design aspects to be considered in such systems, and not the single one. Our idiomatic guitar player system includes knowledge about the biophysical constraints of the performer, the performance characteristics associated with different musical styles, and the performance styles of different individuals. [Research sponsored by CAPES Ministry of Education of Brazil.]

11:45

5aMU11. Synthesis and dynamic behavior of diffracted audio spectra. C. Eswaran and V. Vijayakumar (Faculty of Information Technol., Multimedia Univ., Cyberjaya 63100, Selangor, Malaysia)

It is shown in this paper how the intensity and phase of the harmonics of a spectrum containing a discrete set of input frequecies of unit intensity is modified by diffraction through a circular aperture. The resulting spectrum further shows dynamic behavior which is discussed in terms of a modulation parameter. Examples of synthesized audio spectra which show similarity to known musical instruments are provided as illustration of the synthesis method.

12:00

5aMU12. A system for understanding violin bowing technique through measurement of physical dynamics. Diana Young (MIT, E15-443, 20 Ames St., Cambridge, MA 02142, young@media.mit.edu)

Presented here is a custom-built sensor system for the measurement of violin bowing parameters. An improvement on previous designs, the system includes a set of accelerometers, gyroscopes, and force sensors, installed on a carbon fiber violin bow. In addition, another set of inertial sensors, as well as an electric field position sensor, are mounted on the test violin. When taken together, the data from the violin and bow are combined to achieve estimates (relative to the violin) of the three primary bowing parameters: force, speed, and bow-bridge distance. Because the system is relatively compact and lightweight, it is suitable for operation in realistic performance environments outside the laboratory setting. As such, it is possible to obtain detailed physical gesture data from real performances. The potential of the system is explored in several early studies to investigate the interaction of bowing parameters in common strokes. Distinctions between novices and experts, as well as differences between technique and style among experts, are discussed.

FRIDAY MORNING, 9 JUNE 2006 ROOM 554AB, 8:30 A.M. TO 12:00 NOON

Session 5aSC

Speech Communcation: Perception and Modeling of Speech Processes

Douglas H. Whalen, Chair
Haskins Laboratories, 300 George St., New Haven, CT 06511

Contributed Papers

8:30

5aSC1. The recent origin of human speech. Philip Lieberman (Cognit. and Linguistic Sci., Brown Univ., Providence, RI 02912-1978, philip_lieberman@brown.edu), Robert C. McCarthy (Florida Atlantic Univ., Boca Raton, FL 33431), and David Strait (State Univ. of New York at Albany, Albany, NY)

Studies of human development and swallowing show that a supralaryngeal vocal tract (SVT) capable of producing quantal vowels involves (1) facial restructuring yielding a short oral cavity; (2) a tongue that moves down into the pharynx carrying the larynx down with it; and (3) a long neck. The skeletal features of hominid fossils suggest the absence of fully human speech anatomy until 100 000 years ago. With long faces and short necks, Neanderthals and early anatomically modern humans could not have possessed SVTs capable of swallowing and of producing fully human speech. Probable [i] SVT shapes were computer-modeled from perturbations of vocal-tract area functions obtained from MRIs of adult humans. When normalized to the length of an adult human SVT, $F1/F2$ patterns fall outside of the range of Peterson and Barney's [''Control methods used in a study of the vowels,'' J. Acoust. Soc. Am. **30**, 739–742 (1952)] data used by Lieberman and Crelin, ''On the speech of Neanderthal man,'' Linquistic Inquiry **2**, 203–222 (1971) to assess Neanderthal speech), but at the extreme range for [i] as measured by Hillenbrand *et al.* [''Acoustic characteristics of American-English vowels,'' J. Acoust. Soc. Am. **97**, 3099–3111 (1995)]. In neither case are the high-frequency spectra characteristic of human quantal [i]'s. $F1/F2$ patterns for unnormalized SVTs fall outside the human range for both plots.

8:45

5aSC2. Better model and decoding methods for automatic speech recognition. Douglas D. O'Shaughnessy and T. Nagarajan Li (INRS-EMT, 800 de la Gauchetiere west, Ste. 6900, Montreal PQ, H5A 1K6 Canada)

Different ways to improve the accuracy and performance of automatic speech recognition (ASR) systems are examined. Earlier research in the INRS group concentrated on improvements in adaptation techniques, to try to reduce the mismatch between system training and operating conditions. More recently, a concentration was made on improvements to the

5a FRI. AM

modeling and decoding methodology of the ASR systems. A two-level syllable model-based decoding approach is proposed here. At the first level, a syllable model-based decoding is performed to segment the utterances into syllable segments, and to identify which syllable group this segment belongs to. Then, at the second level, each segment is rescored using the phoneme model, where the possible phoneme sequences are constrained by the syllables in that syllable group. A new heuristic score is proposed to be added in hidden Markov model (HMM) decoding that indicates the degree of competition among different HMM states. Speech features are obtained from the posterior of this HMM approach and these features are used to further improve recognition results. The state evolution tracks are incorporated as features in an HMM-based recognizer. Experiments were carried out on a phoneme classification task (TIMIT) and these methods were found to show improvements compared to a baseline performance. [Work supported by NSERC-Canada and Prompt-Quebec.]

9:00

5aSC3. Development of a binaural speech transmission index. Sander J. van Wijngaarden and Rob Drullman (TNO Human Factors, P.O. Box 23, 3769 ZG Soesterberg, The Netherlands)

Although the speech transmission index (STI) is a well-accepted and standardized method for objective prediction of speech intelligibility in a wide range of environments and applications, it is essentially a monaural model. Advantages of binaural hearing to the intelligibility of speech are disregarded. In specific conditions this leads to considerable mismatches between subjective intelligibility and the STI. A binaural version of the STI was developed, based on interaural cross correlograms, which shows a considerably improved correspondence with subjective intelligibility in dichotic listening conditions. The new binaural STI is designed to be a relatively simple model which adds only few parameters to the original standardized STI, and changes none of the existing model parameters. For monaural conditions, the outcome is identical to the standardized STI. The new model was validated on set of 39 dichotic listening conditions, featuring anechoic, classroom, listening room, and cathedral environments. For these 39 conditions, subjective intelligibility (CVC-wordscore) was measured, as well as the binaural STI. The relation between binaural STI and CVC-wordscores in dichotic listening conditions closely matches the STI reference curve (standardized relation between STI and CVC-wordscore) for monaural listening. The monaural STI performs poorly in these cases.

9:15

5aSC4. New directions for a speech-based speech transmission index. Rob Drullman and Sander J. van Wijngaarden (TNO Human Factors, P.O. Box 23, 3769 ZG Soesterberg, The Netherlands)

The standardized method for determining the speech transmission index (STI) involves the use of a specific intensity-modulated test signal. The STI is obtained from measurements on the transmission channel, usually showing reductions of the modulation depths in the received test signal. Instead of using an artificial signal, various approaches have been suggested in the literature to use speech as a test signal [cf. Payton *et al.*, J. Acoust. Soc. Am. **111**, 2431 (2002)]. Such a speech-based STI has several advantages, e.g., predicting intelligibility differences due to speaking style and enabling the evaluation of vocoders. As we encountered shortcomings and inaccuracies in the existing speech-based STI methods, we propose a new procedure for estimating the speech-based modulation transfer function (MTF) which approaches the accuracy of conventional STI implementations. The new procedure uses the cross spectrum between the transmitted and received speech signals, with special phase weighting to address the relative importance of shifted modulations of the temporal envelope. Evaluation of the algorithm on a vocoder database showed promising results, yielding an average correlation coefficient of 0.93 between the subjective CVC scores and the speech-based STI for male speech. Details of this new speech-based STI algorithm will be discussed.

9:30

5aSC5. Statistical characterization of target-to-interferer ratio of co-channel speech. Brett Smolenski (Res. Assoc. for Defense Conversion, 525 Brooks Rd., Rome, NY 13440)

Recent research in speaker identification technology suggests that it can operate in co-channel environments provided the system can have access to only the less-corrupted segments of speech. In order to identify the uncorrupted speech segments as accurately as possible, it is necessary to fully characterize the statistics of the random processes generating the uncorrupted segments. In a co-channel environment the uncorrupted speech segments are produced when one speaker's voiced speech overlaps with the other speaker's silence or unvoiced speech. Hence, if one has a statistical model of voiced, unvoiced, and silence segments, one can use this information to obtain a model of the uncorrupted speech segments. To accomplish this, statistical models that account for the observed voiced, unvoiced, and silence segment lengths are first developed. Markov models are used to account for dependencies between voiced, unvoiced, and silence segments. In addition, a model of the sampling distribution of the segmental target-to-interferer ratio (TIR) is developed and the short- and long-term correlation present in the segmental TIR signal is also explored.

9:45

5aSC6. Local versus distal speaking-rate normalization effects. Matthew Winn and Rochelle Newman (Univ. of Maryland, College Park, 0100 Lefrak Hall, College Park, MD 20742)

Individuals vary their speaking rate, and listeners typically use the speaking rate of precursor sentences to adjust for these changes (Kidd, 1989). This study examines the temporal window over which these effects occur, by comparing the size of rate normalization effects for local, distal + local, and global rate changes in a precursor phrase. A single talker produced the precursor sentence "Sarah brought a bag so Paul could get the _" at three different speaking rates, followed by a final word containing a duration-based contrast (bin versus pin). Listeners heard trials in which either the whole precursor sentence was spoken at a fast, medium, or slow rate, or (via cross-splicing) only portions of the sentence varied in rate. Subjects rated the target phoneme in the final word as either a "b" or a "p" using a six-step rating scale. Local rate changes (changes in the speaking rate of "get the") yielded significant normalization effects, seen as the shifting of the VOT boundary between p and b. The addition of distal rate changes, however, yielded no significant changes in perception on top of this local effect. This is in contrast to earlier work supporting two separate (local and distal) effects of rate changes.

10:00–10:15 Break

10:15

5aSC7. Speaker identification in hybrid synthesis: Implications for speech perception. Susan R. Hertz (NovaSpeech LLC, 215 N. Cayuga St. #48, Ithaca, NY 14850 and Dept. of Linguist., Cornell Univ., Ithaca, NY 14853) and Isaac C. Spencer (NovaSpeech LLC, Ithaca, NY 14850)

As part of an evaluation of hybrid synthesis [Hertz, Proc. IEEE Workshop on Speech Synthesis (2002)], perceptual experiments were conducted that tested the hypothesis that stressed vowels are the primary cues to speaker identification. Hybrid sentences were constructed for eight voices, including child and adult and male and female, in which stressed vowels were taken from a single human speaker, but other segments were replaced by surrogates from different sources. Some surrogates were natural speech segments; others were formant synthesized. Some matched the age or gender of the stressed vowel speaker; others did not. After being trained on six human target voices, listeners were asked to identify the hybrid stimuli, and also fully synthetic and natural stimuli (for target and nontarget voices), in terms of age, gender, and whether and how much they matched a target voice. For all categories, hybrid stimuli, in contrast to synthetic ones, were identified as accurately as natural speech, both by listeners familiar with a target voice (e.g., family members) and those unfamiliar. The experiment demonstrated how little attention is paid to consonants and reduced vowels in determining speaker identity, and supported our hybrid approach to modeling voices. [Work supported by NIH Grant 1 R43 DC006761-01.]

10:30

5aSC8. On the relation of apparent naturalness to phonetic perceptual identification of voicing. Robert E. Remez, Claire Landau, and Daria F. Ferro (Dept. of Psych., Barnard College, 3009 Broadway, New York, NY 10027-6598)

Does the qualitative experience of speech sounds influence phonetic perception? Our studies of consonant place perception have revealed a dichotomous relation between phonetic sensitivity and naturalness. Although natural quality and phonetic sensitivity sometimes co-vary, in other conditions phonetic sensitivity is indifferent to huge variation in naturalness. New tests are reported here extending the research to the dimension of voicing, a contrast correlated with qualitatively distinct acoustic constituents in normal production. Two acoustic methods were used to create naturalness variants: (1) variation in the excitation of a synthetic voicing source and (2) variation in the bandwidth of the formant centers. A naturalness tournament was composed of items drawn from the test series, and the sensitivity of perceivers to the voicing contrast was estimated with the cumulative d' across the series in identification tests. Together, the findings show how intelligibility and naturalness can be either orthogonal or contingent aspects of consonant perception for the dimensions of place and voicing. These measures offer a tool to understand the contribution of normative functions in the perception of speech. [Research supported by NIH (DC00308).]

10:45

5aSC9. Garden-path phenomena in spoken word recognition: Gradient sensitivity to continuous acoustic detail facilitates ambiguity resolution. Bob McMurray (Dept. of Psych., E11 SSH, Univ. of Iowa, Iowa City, IA 52242), Michael K. Tanenhaus, and Richard N. Aslin (Univ. of Rochester, Ithaca, NY 14850)

Studies have demonstrated that lexical activation is gradiently sensitive to continuous acoustic detail [McMurray *et al.*, "Gradient effects of within-catagory phonetic variation on lexical access," Cognition **86**, B33–B42 (2002)]. This study investigates the consequences of this for online recognition, hypothesizing that such processes help maintain lexical alternatives and reactivate them if early interpretations are disfavored by subsequent input. Stimuli were pairs like barricade/parakeet, where, if the initial segment was misperceived, disambiguation would occur quite late. The effect of VOT on recognition was assessed for tokens favoring the competitor (barakeet with VOTs between 15 and 0). A categorical system would categorize these as equally [b]; garden path to barricade; and show difficulty reactivating parakeet after /eet/. A gradient system would activate parakeet more for VOTs near the boundary, and recover from the garden path faster. Subjects heard ten 8-step continua of this form and selected the target from screens containing the target (parakeet), competitor (barricade), and unrelated objects, while eye movements were recorded. Fixations revealed a gradient pattern of recovery time to switch from the competitor to the target was linearly related to VOT. Subsequent experiments verified that this was not due to stimulus range, nor to the visual presence of the competitor. Thus, lexical activation is sensitive to continuous detail, and this facilitates ambiguity resolution.

11:00

5aSC10. Not all feedback is created equally: The role of multisensory feedback in the perceptual learning of speech under a novel acoustic transformation. Kevin T. Webster and Lorin Lachs (California State Univ., Fresno, 2576 E. San Ramon, M/S ST11, Fresno, CA 93740, llachs@csufresno.edu)

Previous research has investigated the perceptual learning of speech modified by the frequency-inversion transformation, in which the spectral content of an acoustic pattern is rotated about an arbitrary frequency. The resulting transformed speech is almost completely unintelligible. However, given appropriate training, individuals can learn to perceive the linguistic content of the transformed spoken message. Recent work by Webster and Lachs has indicated that providing multisensory feedback during training appears to enhance the rate at which individuals adapt to the transformation. Furthermore, different types of multisensory feedback give rise to different patterns of learning. For example, learners presented with concurrent visual displays of a talkers lip movements (i.e., articulatory information) demonstrate better learning of training materials than learners presented with concurrent visual displays of text. The goal of the present experiment was to refine the technique for training perceivers in order to exaggerate differences between feedback groups in overall learning. A new control condition was also included to test the hypothesis that nonarticulatory but dynamic visual displays yield qualitatively different patterns of learning that those exhibited by perceivers trained with articulatory displays. The results provided interesting insights into the methodology necessary to test perceptual learning of frequency-inverted speech under multisensory conditions.

11:15

5aSC11. Lexical influences on the progressive facilitation during perception of assimilated speech. Cheyenne Munson, Bob McMurray (Dept. of Psych., Univ. of Iowa, E11 SSH, Iowa City, IA 52242, cheyenne-munson@uiowa.edu), and David Gow (Massachusetts General Hospital, Boston, MA 02114)

Phonological processes such as place assimilation, in which coronal sounds partially adopt the place of a subsequent noncoronal (e.g., green boat becomes green/m boat), may create ambiguity during speech comprehension, but may also paradoxically facilitate word recognition. Gow (2001) demonstrated that partial assimilation facilitates perception of postassimilation context. Two eye-tracking experiments investigated whether this progressive effect is influenced by lexical processes. Assimilated and nonassimilated adjectives (e.g., green) were spliced onto coronal or noncoronal nouns (e.g., boat or dog) to create phonologically plausible and implausible assimilation. In experiment 1, assimilation resulted in nonwords; in experiment 2, words assimilated into other words, cuing lexical competition (e.g., cat box became cat/p box). Subjects viewed a screen showing 4 pictures: a coronal noun, 1 noncoronal, and 2 fillers. Eye movements were monitored as subjects heard instructions to select a picture with a mouse. In both experiments subjects were reliably faster to look at the noncoronal target following the assimilated adjective than the nonassimilated one. This progressive effect occurred approximately 100 ms later when assimilation created competitors (experiment 2). These results support an interactive view of word recognition in which lexical processing interacts in real time with phonological and perceptual grouping and integration processes.

11:30

5aSC12. The effect of tonal information and word frequency on spoken word processing. Yao-ju Lin (Dept. of English, Linguist. Div., Natl. Taiwan Normal Univ., 162, HePing East Rd., Sec. 1, Taipei, Taiwan, 69221021@ntnu.edu.tw) and Janice Fon (Natl. Taiwan Univ., Taiwan)

This paper aims to examine the role of tonal information and word frequency in processing spoken Mandarin lexicon. Experiment 1 uses a lexical decision paradigm in which subjects are to hear high- and low-frequency bisyllabic word pairs whose segmental makeups are similar (e.g., [high frequency] jing4-zheng1 competition vs. [low frequency] jin4-sheng1 promotion). The F0 information of the stimuli are either uncontaminated, transformed for the second syllable, or transformed for both syllables. Experiment 2 uses a shadowing task with the same set of stimuli in which subjects are asked to repeat the words as quickly as possible. Preliminary results showed that tonal information is essential for lexical retrieval, especially for low-frequency words. Based on the TRACE model, it is also expected that subjects are more likely to substitute similar-sounding words of higher frequency for low-frequency words whose F0 information is manipulated.

11:45

5aSC13. Auditory, but perhaps not visual, processing of Lombard speech. Eric Vatikiotis-Bateson, Victor Chung, Kevin Lutz, Nicole Mirante, Jolien Otten, and Johanna Tan (Dept. of Linguist., Univ. of British Columbia, E270-1866 Main Mall, Vancouver, British Columbia V6T 1Z1, Canada)

Perception results for three studies are presented that address the role of Lombard speech in auditory, visual, and auditory-visual speech perception. Predictably, when presented in auditory-only conditions with masking noise, listeners recover speech recorded in noise (Lombard speech) better than speech recorded in quiet and presented with the same level of masking noise. However, there is almost no difference in listener performance when Lombard and quiet speech are presented audiovisually with masking noise. Both conditions are enhanced compared to auditory-alone conditions, but there is no indication that the facial motion correlates, demonstrated previously for quiet speech [H.C. Yehia, *et al.*, Speech Commun. **26**, 23–44 (1998)], play as strong a role in enhancing auditory-visual processing of Lombard speech, even though Lombard speech is accompanied by larger facial motions. Perhaps it is no surprise that, at a cocktail party, one leans in with an ear rather than with the eyes. [Research supported by CFI and NSERC.]

FRIDAY MORNING, 9 JUNE 2006 ROOM 556AB, 7:55 A.M. TO 12:00 NOON

Session 5aSP

Signal Processing in Acoustics, Underwater Acoustics, and Engineering Acoustics: Processing of Acoustic Vector Sensors

Paul Hursky, Cochair

Heat Light and Sound Research, 12730 High Bluff Dr., Suite 130, San Diego, CA 92130

Gerald L. D'Spain, Cochair

Scripps Inst. of Oceanography, Marine Physical Lab., 291 Rosecrans St., San Diego, CA 92106-0701

Chair's Introduction—7:55

Contributed Papers

8:00

5aSP1. Radiated noise measurements with vector sensor arrays. Joseph Clark and Gerald Tarasek (NSWCCD, Code 7340, 9500 MacArthur Blvd., West Bethesda, MD 20817-5000, joseph.a.clark1@navy.mil)

The use of arrays of vector sensors to measure source levels of radiated noise from submarines is currently being studied. Ambient noise limits the accuracy of these measurements. The directional character of the ambient noise field (anisotropy) has been found to affect measurement accuracy differently for alternative kinds of processing. In this talk, two types of vector sensor processing, linear matrix and intensity processing, will be compared theoretically and experimentally. Theoretical comparisons of achievable measurement accuracies will be developed in terms of the fourth moment of the measured signal amplitude (the variance of the sound power). Experimental comparisons will be based on data obtained during recent testing at SEAFAC, the U.S. Navy acoustic signature measurement facility in Behm Canal, Alaska.

8:15

5aSP2. Changes in ocean infrasonic energy density and vector intensity with changes in wind and ocean surface wave conditions as measured by a freely drifting vector sensor. Gerald L. D'Spain and Scott A. Jenkins (Marine Physical Lab., Scripps Inst. of Oceanogr., 291 Rosecrans St., San Diego, CA 92106)

During an experiment in 1.3-km-deep water in the Southern California Bight, the wind speed decreased from about 7 to 1 m/s over a 10-h period, and then increased to 9 m/s over the next 8-h period. Measurements made by a neutrally buoyant, freely drifting vector sensor at 920-m depth show that the corresponding changes in the potential and kinetic energy density levels of the infrasonic sound field have a frequency-dependent delay. The levels in a 0.1-Hz-wide bin at 3 Hz respond to wind-speed changes with approximately a 2-h delay, and the delay in response increases with decreasing frequency at a rate of about 1 h per hertz down to 1 Hz. The corresponding evolution in directionality of the active and reactive acoustic intensity vectors in these same frequency bins and their relationship to the surrounding bathymetry will be presented. The directional spectra of the ocean surface wave field during the experiment were reconstructed using back-refraction modeling with wave data from shore stations and offshore platforms archived by the Coastal Data Information Program to obtain estimates of the temporal variability of the nonlinear wave-wave interactions generating the ocean infrasound. [Work supported by ONR.]

8:30

5aSP3. Optimal detection with vector sensors and vector sensor line arrays. Gerald L. D'Spain (Marine Physical Lab., Scripps Inst. of Oceanogr., 291 Rosecrans St., San Diego, CA 92106), James C. Luby (Univ. of Washington, Seattle, WA 98105), Gary R. Wilson, and Richard A. Gramann (Univ. of Texas, Austin, TX 78113)

The detection performance of single vector sensors and vector sensor line arrays is degraded by nonacoustic self-noise and spatial coherence of the noise between vector sensor components. Results based on optimizing the directivity index for a single vector sensor show that the particle motion channels should always be included in the processing for optimal detection, regardless of self-noise level, assuming these levels are properly taken into account. The vector properties of acoustic intensity can be used to estimate the levels of nonacoustic noise in ocean measurements. Application of conventional, minimum variance distortionless response, and white-noise-constrained adaptive beamforming methods with ocean acoustic data collected by a vector sensor illustrates increase in spatial resolution but corresponding decrease in beamformer output and introduction of bias with increasing beamformer adaptivity. Expressions for the spatial coherence of all pairs of vector sensor components in homogeneous, isotropic noise show that significant coherence exists at half-wavelength spacing between particle motion components. For angular intervals about broadside, an equal spacing of about one wavelength for all components provides maximum directivity index, whereas each of the component spacings should be different to optimize the directivity index for angular intervals about endfire. [Work supported by ONR.]

8:45

5aSP4. An electronic windscreen for microphones. Gary W. Elko and Jens Meyer (mh acoustics LLC, 26 Blackburn Pl., Summit, NJ 07901)

Wind-noise pickup in microphones can be highly disturbing, especially for directional microphones that attain their directivity by using velocity or spatial derivatives of the acoustic pressure. Since small directional microphones are common in audio communications systems, the problem of wind noise is well known and the typical solution is to use a windscreen. Conventional windscreens that block the wind but do not significantly alter the sound field can be very effective, but need to be relatively large to attain a good reduction in wind-noise pick-up. Small directional microphone arrays are more commonly used in hearing aids since they have been clinically shown to improve hearing in noise for hearing-impaired users. Since hearing aids are required to be compact, an effective conventional windscreen is not practical. This talk describes a novel multi-microphone wind-noise reduction algorithm that we refer to as an ''electronic windscreen.'' This new algorithm and its variants have low computational cost and are therefore applicable to commercial digital hearing aids that use multi-microphone arrays to attain directionality.

9:00

5aSP5. Vector sensor processing results using data from the Makai experiment. Paul Hursky, Michael B. Porter, Martin Siderius (Heat, Light, and Sound Res. Inc., 12730 High Bluff Dr., Ste. 130, San Diego, CA 92130), and Bruce M. Abraham (Appl. Physical Sci. Corp., New London, CT 06320)

Interest in small mobile platforms such as AUVs and gliders, and fixed autonomous systems, whose small size and power budget severely limit their ability to deploy larger arrays, has stimulated interest in vector sensors (that sense both pressure and particle velocity). Such combined sensors provide performance that can only be duplicated by larger pressure-only hydrophone arrays. In September 2005, during the Makai experiment, we deployed a four-element vector sensor array in waters off the coast of Kauai in Hawaii. We will present comparisons of processing a towed source and acoustic communications packets recorded on the four-element vector sensor array and a comparable pressure-only sensor array. We will also discuss modifications to propagation models based on Gaussian beams and normal modes to predict particle velocity, showing comparisons between model predictions and our experiment data.

9:15

5aSP6. Imaging marine geological environments with vector acoustics. Dennis A. Lindwall (Marine Geosciences Div., Naval Res. Lab., Stennis Space Ctr., MS, 39529)

Using vector acoustic sensors for marine geoacoustic surveys instead of the usual scalar hydrophones enables one to acquire 3D survey data with instrumentation and logistics similar to current 2D surveys. The addition of a collocated hydrophone allows the signal to be rectified, removing the ghost images. This concept was tested by a scaled experiment in an acoustic water tank that had a well-controlled environment with a few targets. Using vector acoustic data from a single line of sources, the three-dimensional tank environment was imaged by directly locating the source and all reflectors. [Office of Naval Research program element 61153N.]

9:30

5aSP7. Sediment sound speed measurements using buried vector sensors. Anthony P. Lyons (Appl. Res. Lab., The Penn State Univ., P.O. Box 30, State College, PA 16804-0030, apl2@psu.edu), John C. Osler, David M. F. Chapman, and Paul C. Hines (Defence R&D Canada—Atlantic, Dartmouth, NS, Canada, B2Y 3Z7)

As part of the SAX04 sediment acoustics experiment conducted off the coast of Florida in the Gulf of Mexico, vector sensors containing three-axis accelerometers and pressure sensors were buried in the seabed. These sensors were used as the receivers to measure sediment sound speed dispersion from 300 to 3000 Hz a band in which it has been particularly difficult to make these measurements. The acoustic intensity measured by the vector sensors was combined with the experimental geometry and propagation conditions, to relate the angle of arrival to sediment sound speed as a function of frequency. Lower frequency measurements, 300 to 1200 Hz, used noise radiated from a moored research vessel as the acoustic source. Higher frequency measurements, 800 to 3000 Hz, used transmissions from an acoustic projector in a three-point mooring in the water column that could be adjusted to change the angle of ensonification. Results from the two techniques are compared with each other and previous measurements of dispersion in sandy sediments. [Work performed under ONR Grants N00014-04-1-0013 and N00014-03-1-0883.]

9:45

5aSP8. Development of a triaxial pressure-velocity probe for deep submergence applications. J. A. McConnell, P. D. Lopath, S. C. Jensen (Acoustech Corp., 4900 South Broad St., Philadelphia, PA 19112), D. H. Trivett, V. B. Biesel, J. J. Caspall (Georgia Inst. of Technol., Atlanta, GA 30332), J. F. McEachern, and R. Coughlan (Navmar Applied Science Corp., Warminster, PA 18974)

A triaxial pressure-velocity probe has been developed for the purpose of measuring the ambient acoustic noise field in the ocean at frequencies below 500 Hz. The probe consists of an omnidirectional hydrophone, a compliantly suspended sphere containing a triaxial geophone, and an electronics housing that contains a four-channel preamplifier and a digital compass. All of these components are packaged in a frame which in turn is attached to a deployment cable (i.e., a strength member) and submerged to a prescribed depth. The performance specifications for the transducers include a submergence depth rating of 6 km and a noise floor of 40 dB *re*: 1 mPa^2 per hertz. These two parameters essentially drive the design of the probe and the paper will address these topics from the standpoint of the design evolution that took place in order to meet the specifications. Other aspects of the probe's development include efforts to mitigate flow-induced noise and shielding the compass from the magnetic field emitted by the geophones. Test data obtained on a prototype unit will be presented. [Work supported by C. M. Traweek at the Office of Naval Research Code 321MS.]

10:00–10:15 Break

10:15

5aSP9. Forming first- and second-order cardioids with multimode hydrophones. J. A. McConnell (Acoustech Corp., 4900 South Broad St., Bldg. 6, Ste. LL00, Philadelphia, PA 19112), R. C. Haberman, T. A. McCormick (BBN Technologies, Mystic, CT 06355), and J. Rudzinsky (Appl. Physical Sci. Corp., New London, CT 06320)

Most everyone in sonar signal processing is familiar with forming a first-order cardioid using scalar and vector sensors, which are defined as the reciprocal analog of a monopole and dipole radiator, respectively. The second-order cardioid can be formed using scalar, vector, and dyadic sensors, in which the latter sensor is defined as the reciprocal analog of a quadrupole radiator. Multimode hydrophones configured as piezoelectric cylinders and spheres with segmented electrodes have the ability to form the both types of cardioids. This paper compares and contrasts the use of multimode hydrophones to produce each beam pattern and presents the results of several sensors that were developed. Topics relating to the sensitivity, bandwidth, noise floor, beamwidth, beam steering, and directivity index will be covered. [Work supported by C. M. Traweek at the Office of Naval Research Code 321MS and L. Shumway at the Department of Homeland Security.]

10:30

5aSP10. Development of a miniature uniaxial pressure-acceleration probe for bioacoustic applications. J. A. McConnell and S. C. Jensen (Acoustech Corp., 4900 South Broad St., Bldg. 6, Ste. LL00, Philadelphia, PA 19112)

The bioacoustics community has recently expressed an interest in using collocated measurements of the acoustic pressure and the particle velocity to categorize how certain species of marine life transmit and receive sound. For most applications the frequency range of interest spans from tens of hertz to a few kilohertz. Because the test specimens and the test facilities can be relatively small, a miniature pressure-acceleration $(b-a)$ probe was developed. The probe conforms to the geometry of a cube having a principal side length of 31.75 mm (1.25 in.) and contains flexural mode piezoelectric transducers to measure the acoustic pressure and one component of the acoustic particle acceleration vector. The cube also houses a low-noise two-channel preamplifier and is compliantly suspended inside a light frame. This paper highlights the design evolution of a few prototypes and addresses issues associated with the electronic noise floor of the accelerometer, mechanical cross talk between the transducers, damping of the external suspension system and transducers, and the dynamics of the signal cable. [Work supported by Acoustech Corporation IR&D.]

10:45

5aSP11. Position-invariant close-talking microphone array. Jens Meyer and Gary W. Elko (mh acoustics LLC, 26 Blackburn Pl., Summit, NJ 07901)

Close-talking microphones are commonly used for speech pick-up in high-noise environments since they can significantly improve the signal-to-noise ratio (SNR) relative to conventional pressure microphones. Increased SNR is based on the assumption that unwanted noise sources are in the far field. However, close-talking microphones have to be carefully positioned or otherwise they can have reduced SNR improvement. A new close-talking microphone array is proposed that is orientation invariant with respect to the attenuation of far-field noise sources. It is shown that by appropriate processing of the close-talking microphone array signals, one can adaptively compensate for the distance and orientation of the microphone for a near-field source. The proposed close-talking array is based on a spatially orthonormal decomposition of the sound field for a near-field source. It consists of four or more pressure microphones that are mounted in the surface of a small rigid sphere. Finally, it is shown that decomposing the sound field into spatially orthonormal modes leads to a computationally efficient implementation. The orientation compensation is applicable not just to close-taking microphones but to directional microphones in general, e.g., podium microphones or hand-held vocal microphones.

11:00

5aSP12. Intensity processing of vector sensors in the bistatic regime. Nathan K. Naluai, Gerald C. Lauchle, and Thomas B. Gabrielson (Grad. Program in Acoust., The Pennsylvania State Univ., 217 Appl. Sci. Bldg., University Park, PA 16802)

Acoustic intensity processing of signals from directional frequency analysis and recording (DIFAR) acoustic subsystems is used to enhance the detection of submerged bodies in bistatic sonar applications. In some directions, the scattered signals may be completely dominated by the incident blast from the source, depending upon the geometry, making the object undetectable by traditional scalar pressure measurements. Previous theoretical derivations suggest that acoustic vector intensity sensors, and the associated intensity processing, are a potential solution to this problem. Deep-water experiments conducted at Lake Pend Oreille in northern Idaho are described. A large body of revolution is located between a source and a number of SSQ-53D DIFAR sonobuoys positioned from 5 to 30 body lengths away from the scattering body. Scalar pressure measurements change by less than 0.5 dB when the scattering body is inserted in the field. However, the phase of the acoustic intensity component formed between the omnipressure and directional channel orthogonal to the direction of incident wave propagation varies by as much as 75. This metric is shown to be a repeatable and strong indicator of the presence of the scattering body. [Work supported by ONR, code 321MS.]

11:15

5aSP13. Processing of vector fields in underwater waveguides. Brian R. Rapids (Appl. Phys. Lab., Johns Hopkins Univ., 11100 Johns Hopkins Rd., Laurel, MD 20723-6099, brian.rapids@jhuapl.edu)

The ability to measure the scalar and vector properties of underwater acoustic fields has improved over the past decade with advancements in transducer technology. Investigations regarding the processing of such sensors have principally focused upon the exploitation of a vector sensor as a directional hydrophone and the ability to achieve significant spatial gains in an unusually small form factor (i.e., superdirectivity). These superdirective elements may then be examined in the framework of standard linear array processing techniques. Alternatively the scalar and vector sensors can be combined multiplicatively to generate estimates of the acoustic intensity vector. Several investigations regarding the detection of signals in noise with these multicomponent sensors have found that optimal linear processing produces a superior signal-to-noise ratio at the back end of the processor when compared to the intensity processor. While the intensity-based processing may not provide an optimal detector, it does provide insight into the fundamental nature of the underwater acoustic field by characterizing the energy flux density. This talk will discuss the effects of signal-model mismatch on the linear and multiplicative process by examining cases in which the assumed phase difference between pressure and particle velocity is not zero.

11:30

5aSP14. Comparative beamforming studies employing acoustic vector sensor data. Kevin B. Smith (Dept. of Phys., Naval Postgrad. School, Monterey, CA 93943), Roger T. Richards, and Philip V. Duckett, Jr. (Naval Undersea Warfare Ctr., Newport, RI 02841-1708)

Data obtained from a vector-sensor line array (VSLA), while deployed vertically at Lake Pend Oreille during calibration, are examined in the context of propagation phenomena. Both broadband (FM sweep) and cw signals are analyzed. Only linear plane-wave beamforming is considered, based on well-established array processing techniques for vector sensors [Cray and Nuttall, J. Acoust. Soc. Am. **110**, 324 (2001)]. Normalization is addressed, as is the determination of the vertical plane of propagation for proper steering of the vector components. Results show that the vector sensor array can be properly scaled and steered using a strong, single, cw source. Peak-to-sidelobe improvements of several dB were observed using vector beamforming rather than pressure-only beamforming. Arrival direction resolution using cardioid null is also presented. The vertical configuration allowed testing of predicted multipath interference behavior. As expected, arrival angle determinations with a single vector sensor is possible with the broadband signal. The difficulty of similarly resolving multipath arrival angles with one vector sensor and a single cw tone is shown. An array of vector sensors will resolve these angles. [Work completed with the support of Michael Traweek, Thomas Curtin, and Ellen Livingston, all from the Office of Naval Research, and by Daniel Deitz of NAVSEA-PMS403.]

11:45

5aSP15. Acoustic source separation via particle velocity vector measurement. Kenbu Teramoto and M. D. Tawhidul Kahn (Dept. of Mech. Eng., Saga Univ., Saga, Japan)

A novel blind source separation of a mixture of two or more voice signals has been proposed in the present paper. The separation system has been focused based on the spatio-temporal blind source separation algorithm. The proposed algorithm utilizes the linearity among the signals of sound pressure and the components of three-dimensional (x, y, and z directional) particle velocity, all of which are governed by the equation of motion. Principally, as the mechanism of blind source separation uses no *a priori* information about the parameters of convolution, filtering, as well as mixing of source signals, some simple assumptions such as the statistical independency of the linearly combined (mixed) observed signals containing zero mean as well as unit variance have been implied in the present separation algorithm. Therefore, the proposed method has successfully simplified the convoluted blind source separation problem into an instantaneous blind source separation problem over the spatio-temporal gradient spaces. An acoustic experiment with two female voices has been carried out to compare the simulated data as well. A Micro-Flown system has been adopted to evaluate the voice signals efficiently.

FRIDAY MORNING, 9 JUNE 2006 ROOM 552A, 8:00 TO 9:45 A.M.

Session 5aUWa

Underwater Acoustics: Acoustic Propagation in Sediments

Nicholas P. Chotiros, Chair
Univ. of Texas, Applied Research Lab., P.O. Box 8029, Austin, TX 78714

8:00

5aUWa1. Acoustic mode at low frequencies in marine sediments: Derivation from first principles using the method of homogenization. Allan D. Pierce, William M. Carey, and Paul E. Barbone (Boston Univ., Boston, MA 02537)

The original low-frequency Biot model factors into three characteristic modes, one being the acoustics mode, having only three significant aggregate parameters: apparent density and bulk modulus, and a numerical coefficient relating attenuation to ω^2. This prediction, although from a heuristic multiparametered model, finds an independent basis in a modification of the derivation by Burridge and Keller [“Poroelasticity equations derived from microstructure,” J. Acoust. Soc. Am. **70** 1140–1146 (1981)] of the poroelasticity equations from microstructure, using only the equations of elasticity and of those for a fluid with viscosity, with no exotic mechanisms. The overall mathematical procedure is the method of homogenization. Isolation of the acoustic mode is achieved by introducing a reference velocity scale, c_{ref}, the intrinsic sound speed in the fluid alone, with the macroscopic length scale H taken as c_{ref}/ω. Mass densities of fluid and elastic media are assumed comparable, and the elastic modulus is assumed comparable to bulk modulus of fluid. The nature of sediments suggests that one needs two inner scales, the larger of which is a representative grain dimension, and the smaller of which is associated with the contact areas of adjacent grains caused by gravitational compression.

8:15

5aUWa2. Broadband sound velocity and attenuation in sandy seabottoms in shallow water. Ji-Xun Zhou and Xue-Zhen Zhang (Georgia Inst. of Technol., Atlanta, GA 30332-0405 and Inst. of Acoust., Chinese Acad. of Sci., Beijing 100080, China)

A previous paper analyzed and reviewed low-frequency (LF) field measurements conducted at 17 locations in different coastal zones around the world, and indicated that the effective acoustic attenuation curves for sandy seabottoms, derived from different field characteristics at these locations, exhibited similar magnitude and nonlinear frequency dependence in a frequency range of 50–2000 Hz. [Zhou and Zhang, “Nonlinear frequency dependence of the effective seabottom acoustic attenuation from low-frequency field measurements in shallow water,” J. Acoust. Soc. Am. **117**, 2494 (2005)]. This paper aims to report the sound velocity in sandy bottoms, derived from the LF field measurements. The resultant LF velocity ratio in the sandy bottom and in the water (at the water/bottom interface) is in a range of 1.04–1.08. Combining the LF field-derived sound velocity and attenuation with the benchmark data of bottom acoustic parameters at mid- to high frequency, obtained from the SAX99 [Williams *et al*, IEEE J. Ocean. Eng. **27**, 413–428 (2002)], offers a reference data set of sound velocity and attenuation in sandy bottoms for a broadband of 50–300 000Hz. The broadband sound velocity data show a significant amount of dispersion. The attenuation data are robust enough to cast doubt on a universal assumption of a linear dependence of attenuation on frequency. [Work supported by ONR and CAS.]

8:30

5aUWa3. Dispersion and attenuation due to scattering from heterogeneities in the frame bulk modulus of sand sediments. Brian T. Hefner and Darrell R. Jackson (Appl. Phys. Lab., Univ. of Washington, Seattle, WA 98105)

A model that predicts the sound speed and attenuation of Biot fast compressional waves when there is scattering into the slow compressional wave has been developed. The scattering is due to small scale heterogeneities in the sediment properties such as the porosity or the bulk modulus of the frame that arise due to the random packing of the sediment grains. This model may account for the deviation of measured fast wave attenuation from the predictions of Biot theory that have been observed in both ocean and laboratory sediments. To compare the predictions of this theory with sound speed and attenuation measurements, the spatial statistics of the sediment properties must be known. Despite work in the granular physics community to understand the distribution of the contact forces between the grains, no direct measurements have been made on the spatial variations in the bulk modulus for dry or fluid-saturated sediments. In order to estimate these variations, we have performed molecular dynamics simulations of spherical bead packs which may approximate simple glass bead sediments. The progress of these simulations and the techniques used to estimate the bulk modulus on scales approaching the grain diameter will be presented. [This work is supported by ONR.]

8:45

5aUWa4. Seabed sediment properties from single-channel seismic profiling data. Klaus C. Leurer, Brian O'Connell, and Colin Brown (Dept. of Earth and Ocean Sci., Natl. Univ. of Ireland, Galway, Galway, Ireland, klaus.leurer@nuigalway.ie)

An acoustic model is presented for the calculation of velocity and attenuation of compressional waves in unconsolidated marine sediments. The model uses the concept of a standard sediment structure that assumes spherical grain shapes and a bimodal grain-size distribution in which the

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small particles represent the clay fraction. The description of the acoustic response of this bimodal sediment model is based on Biot's poroelastic theory and includes two viscoelastic extensions to the original formulation to enable the resulting combined model to cover a wide range of unconsolidated marine sediments. The first extension, an effective-grain model, is introduced to include the fine-grained spectrum of these sediments and the second, a viscoelastic grain-contact model, to account for their unconsolidated granular nature. The acoustic model provides the basis for a neural-network (NN) inversion scheme to extract sediment physical properties from measured single-channel seismic reflection profiling data. Input parameters for the NN are travel times, attenuation, and reflectivities, which are determined from the seismograms. Preliminary application to real reflection data has provided reasonable results within the framework of our standard sediment model for sediment thickness, density, porosity, sound velocity, and concentration of fine-grained material.

9:00

5aUWa5. Gassy sediment acoustics: Low-frequency sound-speed measurements paired with computed x-ray tomography imaging in reconstituted natural sediments. Preston S. Wilson (Mech. Eng. Dept., Univ. of Texas at Austin, 1 University Station C2200, Austin, TX 78712-0292, pswilson@mail.utexas.edu), Allen H. Reed, Warren T. Wood (Stennis Space Ctr., MS 39529-5004), and Ronald A. Roy (Boston Univ., Boston, MA 02215)

Various models of sound propagation in gas-bearing sediments have appeared in the literature but they are largely unverified by experiment. This is primarily due to the difficulty of assessing the bubble-size distribution and global void fraction in optically opaque sediments. We successfully measured the sound speed in reconstituted natural mud and kaolinite sediments containing varying fractions of biogenic gas bubbles. We used a one-dimensional acoustic resonator technique to determine the low-frequency (100–2000 Hz) sound speed of sediment samples contained within cylindrical acrylic core liners. High-frequency (400 kHz) time-of-flight sound-speed measurements were obtained on a subset of the samples. The sediment samples were contemporaneously imaged with a computed X-ray tomography system that yielded independent estimates of the void fraction and bubble-size distribution. Measured sound speeds ranged from 1520 m/s for gas-free mud to as low as 280 m/s for gassy mud and 110 m/s for gassy kaolinite. CT-derived mud void fractions ranged from 0.001 to 0.002, and the kaolinite void fraction was 0.005. The results of these measurements will be compared to existing theory, but a preliminary analysis shows reasonable agreement between measurement and Wood's suspension model. [Work supported by NRL and the UT College of Engineering.]

9:15

5aUWa6. An experimental laboratory investigation of the low-frequency acoustics of seagrasses. Preston S. Wilson (Mech. Eng. Dept., Univ. of Texas at Austin, 1 Univ. Station C2200, Austin, TX 78712-0292, pswilson@mail.utexas.edu) and Kenneth H. Dunton (Univ. of Texas, Port Aransas, TX 78373-5015)

The acoustics of seagrass beds can impact sonar operation in shallow water and may also be exploited for remotely assessing the health of estuarine and littoral ecosystems. The acoustics are dominated by air channels (aerenchyma) within the plants and air bubbles produced by the plants during photosynthesis. *In situ* studies of seagrass beds have confirmed their unique acoustic properties with respect to side-scan sonar backscatter and propagation effects. The results of these studies suggest a richness of exploitable phenomena, but corresponding forward theoretical models have yet to appear. To address this question, low-frequency acoustic laboratory experiments were conducted on three freshly collected seagrass species, *Thalassia testudinum*, *Syringodium filiforme*, and *Halodule wrightii*. A one-dimensional acoustic resonator technique was used to assess the effective acoustic properties and gas content of individual plants, leaves, and rhizomes. We successfully tracked bubble formation and transport from above- to below-ground tissues and present our results with respect to model predictions based on independent optical gas content estimates obtained via microscopic cross-section imagery. [Work supported by the UT College of Engineering and the Texas Sea Grant College Program Grant R/ES-87.]

9:30

5aUWa7. Ultrasonic velocity measurements for synthetic gas-hydrate samples. Hailan Zhang, Dong Wang, Haibo Zhao, Weijun Lin (Inst. of Acoust., Chinese Acad. of Sci., Beijing 100080 China), and Xiuming Wang (CSIRO Petroleum, Bentley WA 6102, Australia)

Laboratory ultrasonic methods offer a way of studying acoustic velocity of a gas-hydrate bearing formation. By measuring ultrasonic velocities of the gas-hydrate samples in various temperature and pressure conditions, more effective inversion techniques can be developed to quantitatively evaluate gas-hydrate concentration and distributions. Low-temperature laboratory measurements of compressional velocities in compacted samples are conducted. These gas-hydrate samples are synthesized by using various densities at various pressures and temperatures. At -10°C, the compressional velocities of the compacted gas-hydrate samples are from 2440 to 3570 m/s with the density range from 475 to 898 kg/m^3. Compressional velocity measurements are made where the temperature and pressure can be controlled. When the pore pressure increases from 10 to 40 MPa, the compressional velocities of the sample increases from 2340 to 2600 m/s at 1.5°C. When the temperature decreases from 10° to -13°C, the compressional velocity will increase from 3600 to 3800 m/s at a pore pressure of 6 MPa. Our experimental results are qualitatively in agreement with those of weighted average model and the Biot-Gassmanns model when the gas-hydrate concentration in a sediment bearing sand is about 20%. [Work supported by National Natural Science Fundation of China, No. 10534040.]

Session 5aUWb

Underwater Acoustics: Acoustic Backscattering and Clutter

Nicholas P. Chotiros, Chair
Univ. of Texas, Applied Research Lab., P.O. Box 8029, Austin, TX 78713

Contributed Papers

10:30

5aUWb1. Statistical characterization of sonarlike clutter observed on the STRATAFORM during the 2003 Acoustic Clutter Experiment in the 400–1500-Hz region. John R. Preston (Appl. Res. Lab., The Penn State Univ., P.O. Box 30, State College, PA 16804)

In 2003 ONR sponsored the Acoustic Clutter Experiment to study shallow-water scattering and clutter in the STRATAFORM area off New Jersey. Sources were bistatically received coherent pulses. The receiver was the five octave research array (used horizontally). The STRATAFORM has benign surface morphology but contains many subsurface features. MIT researchers have shown fish to be a primary source of the observed clutter and reverberation. *K* distributions, with their shape and scale parameters, are used to describe non-Rayleigh behavior. Statistical characterization is presented versus location. The ''bandwidth'' effect is shown where the shape parameter first decreases inversely proportional to bandwidth but then increases back toward the Rayleigh distribution at higher bandwidths. The shape parameter estimates for the 2003 data are well fit by an elongated patch model of Abraham and Lyons. It is shown that shape parameter estimates are about the same for the 2003 and 2001 data taken in the same area. The main differences between the data sets are that the typical scatter sizes/strengths seem to have increased. This is consistent with observations by Nero of NRL and by MIT researchers of higher and more concentrated fish populations observed in 2003. [Work supported by ONR Code 32, Grant N00014-05-1-0156.]

10:45

5aUWb2. The effect of multi-path on synthetic aperture sonar image statistics. Shawn F. Johnson, Anthony P. Lyons, Douglas A. Abraham (Appl. Res. Lab., The Penn State Univ., 117 Appl. Sci. Bldg., State College, PA 16804), and Kevin Williams (Univ. of Washington, Seattle, WA 98105)

The K-distribution shape parameter has been shown to be a useful metric of seafloor reverberation statistics. Using this metric, synthetic aperture sonar (SAS) data collected during the Office of Naval Research sponsored SAX04 experiment in the Gulf of Mexico during October 2004 has been processed to examine image statistics as a function of range from the sonar. SAS images are unique in that the resolution cell size does not increase with range as for traditional sonar. Analysis of the SAX04 data has shown that in spite of the fixed resolution cell size, shape parameter estimates were found to increase with range from the sonar. The increase can be attributed to multi-path propagation. At increasing distance from the sonar, scattered returns arising from additional propagation paths arrive in a specified time window, with the result that two or more resolution cells are contributing, which leads to a larger estimate of the shape parameter than would be expected when only the direct path contributes. Model results using the sonar system geometry and seafloor scattering theory will be used to demonstrate this effect. [Work supported by ONR Codes 321 and 333. First author supported by a National Defense Science and Engineering Graduate Fellowship.]

11:00

5aUWb3. Time evolution of acoustic scattering from a bioturbated seafloor. Anthony L. Gerig, Anthony P. Lyons (Appl. Res. Lab., Penn State Univ., P.O. Box 30, State College, PA, 16804, alg14@psu.edu), and Eric Pouliquen (NATO Undersea Res. Ctr., La Spezia, Italy)

Knowledge of the structure, composition, and time evolution of the seabed is essential to modeling acoustic seafloor scattering. The purpose of this work is to quantify the change in a bioturbated seafloor profile with time, and use the results to predict concomitant changes in a backscattered acoustic wave. To accomplish this task, digital stereo photograph pairs of an approximately 0.5-– m^2 patch of bioturbated seafloor were taken every 10 min over the course of several days at several different locations off the Florida coast as a part of the Office of Naval Research sponsored SAX04 experiment. Photographs were captured, calibrated, and processed according to standard photogrammetric methods to produce surface height fields of the seafloor. The time sequence of fields was then used to estimate height decorrelation with time. Fields were also used to calculate height spectra for the seafloor, and to estimate spectral decorrelation with time. These estimates were subsequently compared for differences in decorrelation time between frequencies. Finally, quantitative results were used as input for the small perturbation surface scattering model to predict the time evolution of acoustic backscatter. Preliminary results indicate that seafloor decorrelation occurs on the scale of hours. [Work supported by ONR.]

11:15

5aUWb4. Statistical and change-detection analyses using quantitative side-scan-sonar data. Jerald W. Caruthers (The Univ. of Southern Mississippi, 1020 Balch Blvd., Stennis Space Ctr., MS 39529)

This paper discusses recent basic, high-frequency, bottom-backscattering research conducted in conjunction with sediment-classification projects and the ONR bottom scattering research projects, KauaiEx and SAX04. This work has a bearing on statistical analyses and bottom change detection relevant to the detection of objects on or near the bottom in shallow water. The research discussed here includes the analyses of data from three surveys made between 2002 and 2004. The primary contributions of this work are based on the use of probability density functions (PDFs) of the backscatter signals to specify the various regimes of bottom sediments and to detect spatial changes in these regimes based on chi-square and other tests applied to these PDFs. A simple, quantitative test for non-Rayleighness of a PDF was also applied. [This work was supported by the ONR.]

11:30

5aUWb5. Low-frequency scattering from a mud volcano located offshore Sicily. Anthony L. Gerig and Charles W. Holland (Appl. Res. Lab., Penn State Univ., P.O. Box 30, State College, PA, 16804)

Mud volcanoes are objects that form on the seafloor due to the emission of gas and fluidized sediment from the earths interior. They vary widely in size, can be exposed, proud, or buried, and are of interest to the underwater acoustics community as potential sources of active sonar clutter. Coincident seismic reflection data and low-frequency bistatic scattering data were gathered from one such volcano buried in the Straits of Sicily. The bistatic data were generated using a pulsed piston source and a 64-element horizontal array, both towed over the top of the volcano. The purpose of this work was to appropriately model low-frequency scattering from the volcano using the bistatic returns, seismic bathymetry, and knowledge of the general geoacoustic properties of the area's seabed to guide understanding and model development. Ray theory, with some approximations, was used to model acoustic propagation through overlying layers. Due to the volcano's size, scattering was modeled using geometric acoustics and a simple representation of volcano shape. Modeled bistatic data compared relatively well with experimental data, although some features remain unexplained. Results of an inversion for the volcanos reflection coefficient indicate that it is acoustically softer than expected. [Work supported by ONR, NURC.]

11:45

5aUWb6. Backscattering by layered elastic seafloors: The equivalent input backscattering strength approach. Laurent Guillon (Ecole Navale, IRENav, BP 600, 29240 Brest Armees, France, guillon@ecole-navale.fr)

In low-frequency experiments, an acoustic signal may penetrate with great depth into the seafloor and interact with submerged structures. Consequently, for many applications, there is need for a model to predict the backscattering strength (BS) of a layered seafloor containing elastic layers. A few models exist; however, two problems exist for their use: first, as most of them are based on specific configurations they are poorly adaptable, and second, very few deal with both layered and elastic seafloors. This paper proposes to extend to the case of elastic layers the equivalent input backscattering strength (EIBS) approach defined for a fluid layered seafloor [L. Guillon and X. Lurton, "Backscattering from buried sediment layers: The equivalent input backscattering strength model," J. Acoust. Soc. Am. **109**, 122–132 (2001)]. The BS is calculated as follows: each layer BS is defined with a specific model, then each local BS is modified to take into account the various effects of stratification, and finally the different modified BS are added to provide the total BS. Two different local models for the individual BS are tested within this general frame. The numerical results obtained show the flexibility of the EIBS approach and its ability to reveal specific effects of layering on the global BS. [Work supported by IFREMER.]