Session 1aAA

Architectural Acoustics: Perception and Binaural Effects in Architectural Acoustics

David H. Griesinger, Chair
23 Bellevue Ave., Cambridge, MA 02140

Contributed Papers

10:00

1aAA1. The importance of the direct to reverberant ratio in the perception of distance, localization, clarity, and envelopment, Part one. David Griesinger (23 Bellevue Ave., Cambridge, MA 02140, dgriesinger@verizon.net)

The component of hall sound that travels directly from a musician to a listener carries information on both the location and distance of the sound source and is vital to both the clarity of the sound and the emotional involvement of listeners. Experiments in laboratories and a hall show that the human ability to separately perceive the direct sound in a reverberant field depends both on the time delay between the direct sound and the reverberant energy and on the direct-to-reverberant ratio (D/R). Threshold data for direct sound perception as a function of time delay and D/R will be presented, and their implications for listening room and hall design will be discussed. In large halls the delay can be long enough that the direct sound is perceptable at D/R values less than −10 dB, but in small halls the threshold becomes as high as 0 dB. Consequences for hall design include: early reflections in the best halls need not be lateral, small halls sound better with fewer early reflections (implying the use of non-shoebox shapes, absorbent stage houses, and shorter reverberation times), and in small halls electroacoustic enhancement should increase only the late reverberation.

10:15

1aAA2. The importance of the direct to reverberant ratio in the perception of distance, localization, clarity, and envelopment, Part two. David Griesinger (23 Bellevue Ave., Cambridge, MA 02140, dgriesinger@verizon.net)

Part two explores the practical consequences of the observations in part one. An experimental method will be presented for objective tests on direct sound detection that yield the threshold data in part one. The rate at which the forward integrated reverberation increases to the level of the direct sound from a short note, and the D/R from short notes or syllables is found to be of particular importance. The insensitivity of current acoustic measures to the perception of direct sound will be discussed, and new measures will be proposed that can be used with live music and speech. We will also present subjective and objective results from a series of experiments in a small hall at the Longy School of Music in Cambridge, Massachusetts. Selective addition of absorption to the stage area and to the lateral walls greatly improved subjective clarity and the balance between musicians on stage and in the hall. Electroacoustics was applied to restore the late reverberation to the value of the unmodified hall without altering the improved clarity. The changes were enthusiastically received by the participants. Audio examples from these experiments will be demonstrated.

10:30

1aAA3. Comparisons of different microphones, microphone orientation, and spacing on late lateral energy measurements. Clothilde Giacomoni, Amy Scherma, and Michelle C. Vigeant (Dept. of Mech. Eng., Acoust. Prog. and Lab., Univ. of Hartford, 200 Bloomfield Ave., West Hartford, CT 06117, vigeant@hartford.edu)

A strong correlation exists between listener envelopment and late lateral energy [J. S. Bradley and G. A. Soulodre, "Objective measures of listener envelopment," J. Acoust. Soc. Am. **98**, 2590–2597 (1995)], but limited work has been conducted on measuring this parameter in actual spaces. Measuring late lateral energy (GLL) requires a figure-of-eight pattern microphone to measure the lateral energy only, but an omnidirectional microphone is also used simultaneously to measure other common room acoustic parameters. An investigation was conducted to determine the effects of various different microphones, microphone orientation, and the spacing between the bidirectional and omnidirectional microphone on measured GLL values. Eight figure-of-eight pattern microphones were used, including both ribbon [(1) Cascade Fat Head II, (2) Royer R-121] and condenser [(2) AKG C414, (1) Neumann KM120, (2) Shure KSM44] microphones. Measurements were taken in an 80-seat lecture-style classroom, using the sine sweep method. The microphones were positioned at three different spacings from the omnidirectional microphone, in both orientations at one spacing, and two different receiver locations. In general, differences of 1–2 dB were found between all of the microphones, while minimal differences were found between the two orientations and various spacings for each microphone. Implications of the results for GLL measurements are discussed. [Work supported by University of Hartford WELFund.]

10:45

1aAA4. Effects of room reverberation time and receiver position on measured binaural room impulse responses. Lauren M. Ronsse and Lily M. Wang (Architectural Engr. Prog., Univ. of Nebraska-Lincoln, Peter Kiewit Inst., 1110 S. 67th St., Omaha, NE 68182-0681, lronsse@unomaha.edu)

This research examines the effects of room reverberation and receiver position on the fine detail in measured binaural room impulse responses, which can affect psychoacoustic perception in rooms such as speech segregation. Binaural room impulse responses have been gathered in four spaces with a range of shapes and reverberation times for three different receiver positions (center, 1 m from side wall, and 1 m from back wall) and three different source orientations (0, 45, and 90 deg from receiver). The directional loudspeaker is located approximately 0.5 m in front of the receiver for all conditions. Relationships between room reverberation times and measures impacting source localization such as interaural level differences and frequency-to-frequency fluctuations are analyzed. The results will determine if conclusions from previous research in this area conducted in a classroom [B.G. Shinn-Cunningham, N. Kopco, and T.J. Martin, J. Acoust. Soc. Am. **117**, 3100–3115 (2005)] are applicable to spaces with varying shapes and reverberation times.

11:00

1aAA5. Perceptual alignment of early-to-late-reverberant energy ratio and reverberation time to match visual environmental cues of a music performance. Daniel Valente and Jonas Braasch (Commun. Acoust. and Aural Architecture Res. Lab., School of Architecture, Rensselaer Polytechnic Inst., 110 8th St., Troy, NY 12180)

The subject of auditory-visual integration has become increasingly important with the introduction of high-resolution playback devices for on-screen object segregation. What effect does the visual location of an object

have on a participant's expectations of that object's auditory stimulus? Several studies have looked at the phenomenon in recent years, but these studies rely on static photographic images to represent the visual scene and mostly use speech signals for testing. Building on these studies, our aim is to propose a testing method using monochromatic compositing (blue-screen technique) to locate a music performance recorded in a studio in a number of virtual acoustical environments being assessed by a listener. In this study, the participant is asked to perceptually adjust two acoustical parameters, early-to-late reverberant energy ratio and reverberation time, of two music performances in five contrasting visual environments according to his or her expectations of how the room should sound given its visual appearance. This study reveals participant resiliency in the presence of forced auditory-visual mismatch; participants are able to adjust the presented acoustic modeling algorithm in a statistically similar way despite randomized starting values for the monitored parameters. Subjective results of the experiment are presented along with objective measurements for verification.

11:15

1aAA6. The impact of visual cues on perceived intimacy in an acoustic ecology. Shane A. Myrbeck, Daniel Valente, Jonas Braasch, and Bobby Gibbs (School of Architecture, Rensselaer Polytechnic Inst., 110 8th St., Troy, NY 12180, myrbes@rpi.edu)

Crossmodal studies of the effects of visual cues on the perceived acoustics of a performance space have suggested the inextricable relationship between vision and audition. When designing a building or virtual environment for music, critical attention must be paid to a visual aesthetic to ensure a desired level of intimacy between the audience and performer. This visual aesthetic is ordinarily left entirely to the main architect or visual artist. However, given the body of research suggesting crossmodal influences on the human auditory system, it is clear that acousticians must also carefully consider the visual effects of their recommendations for acoustic improvement. The objective of this study is to establish the effects of various visual cues on audition as pertaining to intimacy in an ecological context. This study uses digital compositing (bluescreen) techniques to place a musical performance in various virtual environments, and employs real time audio-video processing to create an interactive, cross-modally immersive environment. Participant-adjusted direct-to-reverberant ratio as well as visual cues adapted from stage lighting techniques are used to establish crossmodal relationships. It is hypothesized that the ability to adjust visual cues will significantly effect subjects' perception of intimate acoustic settings.

11:30

1aAA7. Accurate timbre and localization of binaural recordings through headphones using sound pressure measurement at the eardrum. David Griesinger (23 Bellevue Ave., Cambridge, MA 02140, dgriesiner@verizon.net)

Individual HRTFs are almost universally measured with blocked or partially blocked ear canals, and individual headphone equalization is obtained in the same way. Using this method binaural reproduction without head tracking usually results in a significant alteration of timbre as well as inaccurate and/or in the head localization. The error is important—auralizations using incorrect timbre can lead to misleading conclusions. This problem was studied with the help of a dummy head that precisely models the author's pinna, ear canals, and eardrum resonance. A new type of probe microphone was also developed that allows comfortable recordings and measurements at the eardrum. Data from this equipment show that while the spatial variation in HRTFs can be captured with a blocked ear canal up to a frequency of 7 kHz, errors in headphone equalization made with a blocked canal are typically in error by more than 10 dB at midfrequencies. Eardrum measurements of HRTFs and headphones result in superior performance for an individual, and a pair of headphones correctly equalized for an average individual produces out-of-head localization for at least 50% of other individuals in our tests. Data and theory and a noninvasive method for equalization will be presented.

11:45

1aAA8. Influence of vibration on the perception of musical instruments for various stage constructions. Clemeth L. Abercrombie and Jonas Braasch (Rensselaer Polytechnic Inst., Greene Bldg. 110 8th St., Troy, NY 12180, clem@loudensound.com)

Expanding use of multisensory music presentation calls for increased knowledge of human response to audio-tactile stimulus. This presentation outlines an experiment to explore the human ability to distinguish differences in tactile signals generated by musical sources on typical floor constructions. A double bass and bass drum are used to generate binaural audio and tactile signals on various floor constructions. Vibration signals are then compared for differences in arrival time, magnitude, and frequency content. Audio and vibration signals are reproduced in different combination using a calibrated motion platform and headphones. Test participants are asked if they can identify the differences in tactile signals given a fixed audio environment. Test results are compared to those found in other studies obtained using musical signals as well as harmonic and noise signals. The importance of calibrating tactile musical signals is discussed along with the implications of the results with respect to tactile signal synthesis for multimodal music presentation.

Session 1aABa

Animal Bioacoustics and Psychological and Physiological Acoustics: Fish Bioacoustics: Sensory Biology, Sound Production, and Behavior of Acoustic Communication in Fishes I

Joseph A. Sisneros, Cochair
Dept. of Psychology, Univ. of Washington, Seattle, WA 98195

Richard R. Fay, Cochair
Parmly Hearing Inst., Loyola Univ., Chicago, IL 60626

David Zeddies, Cochair
Marine Acoustics Inc., 4100 Fairfax Dr., Arlington, VA 22203

Chair's Introduction—8:00

Invited Papers

8:05

1aABa1. The future of fish bioacoustics. Arthur N. Popper (Dept. of Biology, Univ. of Maryland, College Park, MD 20742, apopper@umd.edu) and Richard R. Fay (Loyola Univ. of Chicago, Chicago, IL 60626)

In 1993 (Brain, Behav. Evol. **41**, 14–38 (1993)] we evaluated changes in our knowledge of fish bioacoustics since our 1973 review and proposed ten issues that still needed to be answered. This presentation asks whether we have have made progress on these issues since 1993 and poses questions for future fish bioacoustics research. Many of the topics suggested in 1993, such as sound localization mechanism, the roles of the swim bladder' and other peripheral structures in hearing, and the contributions of pressure and particle motion in hearing are still open. There are also "newer" questions not apparent in 1993, such as mechanisms of ultrasound detection and the effects of increased human-generated noise on fish. The usefulness of thinking of fish as "hearing specialists" and "hearing generalists" still needs to be assessed. Finally, an overriding question related to the significance of the diversity of fishes and diversity in ear structure, has not been resolved. Indeed, we still do not know whether the diversity found in the fish auditory periphery and CNS are convergent mechanisms to glean the same kind(s) of information about sound, or whether the variation reflects different uses of sound by different species.

8:20

1aABa2. Fish hearing and bioacoustics since 1973. Richard Fay (Parmly Hearing Inst., Loyola Univ. Chicago, IL 60626) and Arthur Popper (Univ. of Maryland, College Park, MD 20742)

In 1973, we wrote a review in *J. Acoust. Soc. Am.* that summarized our view of fish hearing and bioacoustics. This presentation looks back on the the major questions we had at the time and how the field has dealt with them, or not, to the present. Some of the issues identified included: (1) the roles of the various otolith organs for hearing; (2) lateral line function and its relation to hearing; (3) sound source localization; and (4) how sound stimulation works for fishes without obvious specializations for pressure reception. There have been major advances in these and other aspects of fish sound reception, but in every case, there are significant unanswered questions. For example, it is still controversial whether the utricle plays a role in hearing in most species, and the functional role of the lagena remains an enigma. However, we now have a much better understanding of lateral line stimulation and function, and we now understand that it is not an "accessory auditory organ," as was widely assumed initially. Many questions of sound localization are now thought to be answered theoretically, and in some cases empirically, but many critical and fundamental experiments on localization remain to be done.

8:35

1aABa3. Response of fish sound production to short-term ocean upwelling events at the LEO-15 (Long-term Ecosystem Observatory at 15 m depth) ocean observatory. David A. Mann (Coll. of Marine Sci., USF, 140 7th Ave. S., St. Petersburg, FL 33701, dmann@marine.usf.edu) and Thomas M. Grothues (Rutgers Univ., Tuckerton, NJ 08087)

Understanding factors controlling the distribution and reproduction of fishes is crucial for developing ecosystem models of fish populations. Yet these observations are difficult to make on physically relevant time and space scales. A hydrophone was used to record fish sound production associated with reproduction at the LEO-15 ocean observatory to determine the relationship between variation in fish calling behavior and oceanographic variation. Sound production was dominated by Atlantic croaker (*Micropogonias undulatus*), weakfish (*Cynoscion regalis*), and striped cusk-eel (*Ophidion marginatum*). Striped cusk-eels called with a crepuscular pattern, with a strong peak at dusk, less sound production during the night, and a lesser peak in sound production at dawn. Sciaenids called mostly at dusk and night. Nine advection events bringing colder waters to the LEO-15 site were correlated with greatly reduced levels of sound production in Atlantic croaker and weakfish on daily time scales. Our results show how ocean observatories with passive acoustics can study tightly coupled physical oceanographic influences on fish behavior on daily time scales.

8:50

1aABa4. Stereotypy and variability in the mating call of the Lusitanian toadfish, *Halobatrachus didactylus*. M. Clara P. Amorim, J. Miguel Simoes, Vitor C. Almada (Behavioural Ecology Res. Unit, I.S.P.A., Rua Jardim do Tabaco 34, 1149-041 Lisbon, Portugal), and Paulo J. Fonseca (Univ. of Lisbon, 1749-016 Lisbon, Portugal)

During the breeding season, Lusitanian toadfish males become territorial and emit a tonal advertisement call (the boatwhistle) to attract mates to their nests. We recorded 16 males in the Tagus River estuary (Portugal) during the breeding season and studied boatwhistle variability in a short (minutes) and long time scale (days). We also related boatwhistle acoustic features with male physical characteristics. The recorded males were inside closed artificial nests in groups of eight individuals. Boatwhistles showed individuality in short periods of time but some males showed significant intraindividual variability in a longer time scale. Both social interactions (number of calling males) and tide level changed significantly male calling rate and acoustic characteristics of boatwhistles. The fundamental frequency of boatwhistles (equivalent to the sonic muscle contraction frequency) and boatwhistle emission rate were correlated with male condition (somatic and lipid content). Sound emission rate was also correlated with sonic muscle index. These results suggest that different toadfish males produce distinct boatwhistles, and that they have the ability to modulate the acoustic characteristics of their sounds with the level of social interactions, which increase during rising and high tide. Moreover, boatwhistles characteristics and acoustic activity seems dependent on male condition and sonic muscle index.

Contributed Paper

9:05

1aABa5. Recordings of sonic muscle activity of the Lusitanian toadfish *Halobatrachus didactylus* during sound production. Joana M. Jordão, Maria Clara Amorim, and Paulo Fonseca (Centro de Biologia Ambiental, FCUL, Campo Grande, Lisboa, Portugal)

Male Lusitanian toadfish, *Halobatrachus didactylus*, are territorial and have a wide acoustic repertoire. The sound production apparatus consists of a pair of intrinsic large sonic muscles on the two lobes of the swimbladder. Sonic muscle activity and the variability of the vocalizations were related using electromyograms (EMGs). EMG electrodes were cirurgically implanted on the sonic muscles of several territorial males. After recovery, the subjects were released into an experimental tank with an available nest. Sonic muscle activity was registered and the vocalizations were simultaneously recorded using a hydrophone. Social context was manipulated by releasing different kinds of intruders, male(s) or female(s). This action elicited the production of boatwhistles, while grunts were generated during fish disturbance. Preliminary analysis shows that, unlike previously reported, the bilateral sonic muscles contract simultaneously during the generation of both signal types. The muscle contraction period is halved during the production of the grunt relative to the boatwhistle. The EMG period matches the sound pulse period and determines the fundamental sound frequency. [JMJ was supported by a Post-doc grant (SFRH/BPD/36510/2007).]

Invited Papers

9:20

1aABa6. Toadfish boatwhistles also function as territorial keep-out signals. Raquel O. Vasconcelos (Dept. de Biologia Animal and Centro de Biologia Ambiental, Faculdade de Ciências, Univ. de Lisboa. Bloco C2 Campo Grande, 1749-016 Lisbon, Portugal, rfvasconcelos@fc.ul.pt), J. Miguel Simoes (Unidade de Investigacao em Eco-Etologia, 1149-041 Lisbon, Portugal), Paulo J. Fonseca (Univ. de Lisboa, 1749-016 Lisbon, Portugal), and M. Clara P. Amorim (Unidade de Investigacao em Eco-Etologia, 1149-041 Lisbon, Portugal)

Boatwhistles produced by the toadfish *Halobatrachus didactylus* seem to function as mate attraction calls during the breeding season. Recent observations, however, indicate that their emission is not restricted to this period, suggesting that boatwhistles may have other functions. The possible dual function of boatwhistles was tested by eliciting sound production from reproductive males in advertisement and territorial defence contexts. These were simulated by creating aggregations of confined nesting males in an intertidal area and by conducting territorial intrusion experiments in tanks, respectively. Furthermore, we investigated whether parental care (eggs in the nest) affected the behavioral responses of territorial males. Nesting males kept in aggregations emitted boatwhistles spontaneously for several days. The relation between calling rate and number of eggs in the nest is under analysis. During territorial intrusions, resident males prevented the entrance of intruders in their shelters by producing mostly boatwhistles. Parental males revealed higher aggression levels, exhibiting additional threatening and attack behaviors. Agonistic boatwhistles differed from the mating calls by presenting less amplitude modulation and lower dominant and fundamental frequencies. These results suggest that, apart from attracting mates, the boatwhistles of batrachoidids may also function as an active keep-out signal during territorial defence.

9:35

1aABa7. Evolution and diversity of batrachoidid vocalizations. Aaron N. Rice and Andrew H. Bass (Dept. Neurobiology and Behavior, Cornell Univ., Ithaca, NY 14853, arice@cornell.edu)

Despite the fact that toadfishes are one of the best-studied taxa for understanding fish vocal communication, vocalizations have only been recorded from a low proportion of taxa within the family. Here, we describe the diverse vocal repertoire and novel swimbladder morphology of a tropical Pacific euryhaline toadfish, *Batrachomoeus trispinosus*. *B. trispinosus* produces four classes of vocalizations: long duration hoots, grunts, grunt trains, and grunts exhibiting acoustic beats; the majority exhibiting prominent harmonic features. Calls with acoustic beats, previously unreported for any individual actinopterygian taxon and convergent with songbirds, typically had a predominant frequency around 2 kHz with a beat frequency around 300 Hz. *B. trispinosus* possesses a bilaterally divided swimbladder, forming two completely separate bladders. Grunts exhibiting beats may be produced by bilateral control of the bladders. Phylogenetic analysis based on available molecular data show that *B. trispinosus* and *Halobatrachus didactylus* represent basal lineages in the Batrachoididae, and the separated swimbladders are an independently derived feature of *B. trispinosus*. Thus, the toadfishes may possess higher levels of vocal, motor, and morphological diversity than previously appreciated, and now provide a framework for understanding integrated mechanisms underlying the evolution of acoustic novelty in fishes.

9:50

1aABa8. Sound production and hearing in coral reef butterflyfishes. Timothy C. Tricas and Kelly Boyle (Dept. of Zoology and Hawaii Inst. of Marine Biol., Univ. of Hawaii at Manoa, Honolulu, HI 96822)

Butterflyfishes (family Chaetodontidae) exhibit a diversity of close affiliative social behaviors. All 87 species of *Chaetodon* have a laterophysic connection and extensions of the anterior horns of the swim bladder near the otic capsule, which are proposed to impart sound pressure sensitivity to the normally mechanosensitive lateral line and acceleration-sensitive inner ear. Video kinematic analyses reveal that the motor action patterns used to produce pulsed sounds vary among species. The sound production mechanisms in *Chaetodon* and *Forcipiger* do not appear to be conserved, thus sound production may have evolved independently within the family. Species with long swim bladder horns have lower AEP hearing thresholds than those without, and deflation of the horns and bladder decreases hearing sensitivity in the former group. When the laterophysic connection and swim bladder horns are physically disrupted in wild fish, movement within their territories decreases and fish pairs swim closer together. Thus, the low intensity sounds produced by butterflyfishes may favor close social behaviors that increase communication efficiency among individuals on a noisy coral reef. We are currently investigating how hydrodynamic flow fields produced during social interactions may be used for intraspecific communication during these natural behaviors.

10:05—10:25 Break

10:25

1aABa9. Head and body kinematics of pulse sound generation and feeding in longnose butterflyfishes (Genus *Forcipiger*). Kelly S. Boyle and Timothy C. Tricas (Dept. of Zoology, Hawaii Inst. of Marine Biology, Univ. of Hawaii, 2538 McCarthy Mall, Honolulu, HI 96822, kboyle@hawaii.edu)

Both species of butterflyfish (*Chaetododntidae*) in the genus *Forcipiger* produce pulsatile sounds during territorial interactions. Sounds of one species, the forcepsfish (*F. flavissimus*), are approximately 20 ms and occur with rapid dorsoventral head motion (headbob). We performed laboratory sound production and feeding experiments with high-speed video (600 fps) to test predictions of the hypothesis that sound production in this genus results from evolutionary modification of stereotyped feeding behavior. Cranial elevation is an important component of prey capture for a variety of teleost fishes and facilitates opening of the buccal cavity and oral jaws. Studies on kinematics of feeding in *Forcipiger* by other authors revealed a minor contribution of cranial elevation compared to other fishes. Preliminary results indicate cranial elevation during headbobs involves a comparable range of motion relative to prey capture (approximately 3°–7°), but maximal elevation occurs within 12 ms during the headbob compared to 25 ms in feeding. Additionally, sound emission during feeding is weak to nonexistent. Electroymyography has revealed involvement of anterior epaxialis muscles with cranial elevation during feeding and headbob sound production. These experiments will allow further kinematic comparisons of head movements and associated muscle firing patterns during feeding and acoustic behaviors.

10:40

1aABa10. Sound production and localization in loricariid catfishes. Michael E. Smith, Patrick C. Stewart, Amanda L. Webb, and Brian D. Rogers (Dept. of Biology, Western Kentucky Univ., 1906 College Heights Blvd. #11080, Bowling Green, KY 42101, michael.smith1@wku.edu)

Catfishes of the family Loricariidae have bilobed swim bladders that are adjacent to their ears. We hypothesize that this anatomical design assists these fishes in sound localization and/or sound production. Loricariid catfishes produce short, broadband clicks via pectoral spine stridulation. To test whether these clicks can be localized, we used classical conditioning to train groups of *50 Otocinclus affinis* to come to a conspecific sound on one side of a 200-L aquarium. Four types of experiments were performed—naive, trained, test-1, and test-2. During trained trials, food and a conspecific sound stimulus were presented simultaneously, while only the sound stimulus was presented during naïve (untrained) trials. During the test-1 and test-2 trials, only the conspecific sound stimulus or a 500-Hz sound stimulus, respectively, was presented. *O. affinis* were attracted to the conspecific sound post-training, with 60–80% of the fish remaining on the speaker side of the tank during the stimulus. In summary, *O. affinis* are able to localize a conspecific sound source. Future experiments have been designed to examine the acuity of this sound localization using four speakers in a circular tank, and the effects of swim bladder deflation on the sound localization ability of *O. affinis*.

Contributed Paper

10:55

1aABa11. Bocaccio sounds in the Southern California Bight. Ana Širović (Alaska Pacific Univ., 4101 Univ. Dr., Anchorage, AK 99508, asirovic@alaskapacific.edu) and David A. Demer (Southwest Fisheries Sci. Ctr., NOAA Fisheries, La Jolla, CA 92037)

Sound production by many fish species has been studied extensively, but little is known about sound production by rockfishes (genus *Sebastes*), and only a few species have been reported to be soniferous. Passive acoustic recordings were made during 2007 and 2008 at Hubbs-SeaWorld Research Institute and Southwest Fisheries Science Center tanks containing Bocaccio (*S. paucispinis*). Sounds in tanks were recorded using preamplified hydrophones (HTI-94-SSQ) and were digitized at sample rates of 44,100 or 8000 Hz. Recordings were collected at 14 locations in the Southern California Bight (SCB) using passive acoustic moorings, AURALs, sampling at 8000 Hz from August through October 2007. Three low frequency (<900 Hz), short duration (<4 s) sounds were recorded in tanks containing only Bocaccio multiple times. One of these sounds was also commonly recorded in the SCB and is the same sound that was recorded first by Thompson off San Clemente Island in the 1960s. This sound exhibited the same diel pattern as noted by Thompson, with maximum calling at night, peaking at dusk and dawn. This long-term persistence of the same sound indicates that passive acoustic tools may be used for efficient monitoring of changes in rockfish populations over long time periods.

Invited Paper

11:10

1aABa12. Sound source localization by the plainfin midshipman fish, *Porichthys notatus*. David Zeddies (Marine Acoust. Inc., 4100 Fairfax Dr., Arlington, VA 22203), Richard R. Fay (Loyola Univ. Chicago, Chicago, IL 60626), Peter W. Alderks (Univ. of Washington, Seattle, WA 98195), Kiel Shaub (Univ. of Washington, Seattle, WA 98195), and Joseph A. Sisneros (Univ. of Washington, Seattle, WA 98195)

Sound source localization of the plainfin midshipman fish (*Porichthys notatus*) was studied using the phonotactic response of gravid females to synthetic male advertisement calls. Playback experiments were conducted in a 3.65-m-diameter outdoor concrete tank at the UC Bodega Marine Laboratory in Bodega Bay, CA using a J-9 transducer placed near the center of the tank. The sound field in the tank was measured at 5-cm intervals using an eight-hydrophone array to measure the pressure gradients from which particle motion vectors were calculated. The acoustic measurements confirmed that the J-9 projector was operating as a monopole source. One hundred twenty-two gravid females were released 100 cm away from the center of the sound source while their behavioral responses were video taped during experiments with playback sound "On" (90 Hz tone) and sound "Off" (Controls). A total of 45 positive phonotactic responses from naïve gravid females were recorded from the 62 trials with sound "On." The phonotactic responses consisted primarily of straight to slightly curved tracks to the sound source while abrupt changes in trajectory to the sound source were rarely observed. The results confirm that fish can locate sound sources in the near field. [Work supported by NSF.]

Contributed Paper

11:25

1aABa13. Passive acoustic determination of herring size. Thomas R. Hahn, Jennifer Wylie, and Xiaojun Chen (Rosenstiel School of Marine and Atmospheric Sci. Univ. of Miami, 4600 Rickenbacker Cswy., Miami, FL 33149)

Over the last decade, acoustic signatures of a variety of fishes have been recorded and analyzed. More recently, these vocalizations have been used to passively detect aggregations of fish, demonstrating the potential to supplement and enhance traditional active acoustic surveys. Based on previously published work on acoustic emissions of herring, this paper discusses the possibility of not merely passively detecting absence or presence of aggregations of herring but, additionally, assessing abundance and size. Theoretical considerations as well as data collected in Prince William Sound, Alaska, on Pacific herring (*Clupea Pallasii*) are presented.

Invited Paper

11:40

1aABa14. Assessment of cumulative sound exposure level as criterion for exposure of fish to impulsive sound. Michele B. Halvorsen (Dept. of Biology, Univ. of Maryland, College Park, MD 20742, mb_halvorsen@yahoo.com), Thomas Carlson (Pacific Northwest Natl. Labs., Battelle, Portland, OR 97204), and Arthur N. Popper (Univ. of Maryland, College Park, MD 20742)

Acceptable exposure of fish to pile driving impulsive sound is currently determined by cumulative sound exposure level (SEL). The implicit assumption in this regulatory approach is that cumulative SEL, regardless of how reached, is one of the critical measures of sound exposure for fish health. We are testing this assumption by exposing juvenile Chinook salmon to sequences of impulsive sound differing in individual impulse SEL and number of impulses, but with the same cumulative SEL value. Test fish are exposed in a unique device that permits accurate simulative of impulsive pile driving sound. Following exposure, the hearing of test fish is evaluated using auditory evoked potential methods followed by necropsy for assessment of barotrauma. The null hypotheses we are testing is that the probability of hearing threshold shift $\geqslant$6 dB and the probability of barotrauma mortal injury is the same for all cumulative SEL exposures. The results of these tests will determine the experimental design of additional tests to derive criteria for exposure of fish to pile driving sound. [Work supported by National Cooperative Highway Research Program (NCHRP) and Minerals Management Service (MMS).]

Session 1aABb

Animal Bioacoustics: An Integration of Bioacoustics, Neuronal Responses, and Behavior I

Terry Takahashi, Chair
Inst. of Neuroscience, Univ. of Oregon, Eugene, OR 97403

Chair's Introduction—8:00

Invited Papers

8:05

1aABb1. Spatial aspects of acoustic processing in complex environments. H. Steven Colburn, Barbara Shinn-Cunningham, and Kamal Sen (Hearing Res. Ctr., Boston Univ., 44 Cummington St., Boston, MA 02215, colburn@bu.edu)

From the perspective of acoustical signal processing, it is a challenge to analyze a complex sound environment and segregate the acoustic mixture into individual sources, each with its own spatial characteristics and reverberation pattern. However, auditory systems perform this perceptually critical task with impressive ease. This talk will review how acoustical array processors work to separate sound sources. We then consider what most vertebrates, who have two point receivers separated in space, can accomplish by combining the signals that reach their two ears. The nature of the information available to bird and mammalian auditory systems is briefly reviewed, and the computational approaches these systems take to separating sound sources is assessed, including the logic of processing sound within narrow frequency channels over brief time epochs (rather than, e.g., undertaking broadband analysis). Some observations about the ability of classical binaural processing models to implement earlier stages of processing for these complex environments will be discussed. Finally, performance of human and zebra-finch subjects in tasks requiring judgments about individual sources embedded in complex sound mixtures will be considered in the context of physiologically motivated processing schemes. [Work supported by NIDCD: DC00100 (Colburn), DC05778 (Shinn-Cunningham), and DC07610 (Sen).]

8:30

1aABb2. Neural basis of spatial hearing in reverberant environments. Sasha Devore (Eaton Peabody Lab., Boston, MA 02114 and Harvard-MIT Div. of Health Sci. and Technol., Cambridge, MA 02139), Antje Ihlefeld, and Barbara Shinn-Cunningham (Boston Univ., Boston, MA 02215), and Bertrand Delgutte (Eaton Peabody Lab., Boston, MA 02114)

In reverberant environments, acoustic reflections interfere with the direct wavefront reaching a listener's ears, distorting the spatial cues for sound localization. Yet, human listeners have little difficulty localizing sounds in everyday settings. Our research aims to elucidate the neural basis of spatial hearing in reverberant environments. Using a virtual acoustic space (VAS) technique, we investigated the effects of reverberation on the directional sensitivity of binaural neurons in the inferior colliculus (IC) of anesthetized cats and awake rabbits, focusing on neurons sensitive to interaural time differences. Consistent with the buildup of reverberant energy in a sound stimulus, we find that reverberation increasingly degrades the directional sensitivity of single neurons over time, although the amount of degradation depends on the characteristic frequency and the type of binaural cues available. The neurophysiological data can account for results from recent human psychophysical studies of spatial hearing in reverberant environments. In particular, we show how a population rate model for decoding the observed midbrain responses predicts the main trends in human lateralization performance, suggesting a subcortical origin for robust sound localization in reverberant environments. [Work supported by National Institutes of Health (NIH) Grant Nos. R01 DC002258 (BD), P30 DC005209 (BD), and R01 DC05778-02 (BGSC). SD partially supported by NIH Grant No. T32 DC00038 and the Helen Carr Peake Fund.]

8:55

1aABb3. Comparative studies of the cocktail party problem using birdsong. Micheal L. Dent (B76 Park Hall, Dept. of Psych., Univ. at Buffalo, SUNY, Buffalo, NY 14260), Barbara G. Shinn-Cunningham, and Kamal Sen (Boston Univ., Boston, MA 02215)

Many animals are adept at identifying communication calls in the presence of different masking sounds, much like humans navigating a cocktail party. In order to compare human and avian abilities in analyzing signals in complex acoustic mixtures, we measured identification of birdsong by humans and birds in the presence of different maskers and compared results to physiological recordings in zebra finches. In humans, spectrotemporal similarity of targets and maskers causes difficulties in segregating and selecting the target out of the sound mixture; however, for similar targets and maskers, spatial separation alleviates this interference. Field L recordings in finches reveal differences in the forms of neural interference caused by the various masker types, consistent with human behavior. However, avian identification performance is similar for the different maskers. Moreover, spatial separation of target and masker causes little improvement in avian target identification unless target and masker have distinct temporal envelopes. When target and masker envelopes differ, avian identification performance improves significantly, regardless of masker type. These experiments demonstrate that both birds and humans can segregate communication signals from a sound mixture, but suggest that the acoustic attributes that the two species use to solve the cocktail party problem differ.

9:20

1aABb4. Localization and identification of concurrent sounds in the owl's auditory space map. Clifford H. Keller and Terry T. Takahashi (Inst. of Neurosci., Univ. of Oregon, Eugene, OR 97403)

In nature, sounds from multiple sources sum at the eardrums, generating complex cues for sound localization and identification. In this clutter, the auditory system must determine "what is where." We examined this process in the auditory space map of the barn owl's inferior colliculus using two spatially separated sources simultaneously emitting uncorrelated noises, which were uniquely identified by different frequencies of sinusoidal amplitude modulation. The neurons responded when either source was placed within the receptive field, suggesting two clearly segregated foci of activity at appropriate loci on the space map. The spike trains were locked strongly to the AM of the source within the receptive field, while the other source had minimal influence. Two sources AMed at the same rate were successfully resolved, suggesting that source separation is based on differences of fine structure. The spike rate and synchrony were stronger for whichever source had the stronger average binaural level. A computational model showed that neuronal activity was largely proportional to the degree of matching between the momentary binaural cues and the neuron's preferred values. Individual neurons respond to and synchronize with sources in their receptive field if there are frequencies having an average binaural level-advantage over a second source. [NIH-DC003925.]

9:45

1aABb5. Localization of amplitude modulated sounds and their echoes in the barn owl. Brian S. Nelson and Terry T. Takahashi (Inst. of Neurosci., Univ. of Oregon, Eugene, OR 97403)

Echoes that arrive after short delays are not perceived as events separate from sounds arriving directly from an active source, a phenomenon known as localization dominance. Studies have focused on the time-advantage at the onset of an echo/direct-sound pair to explain localization dominance. The head-saccades of barn owls and the responses of neurons in their auditory space maps were measured when sound pairs were gated simultaneously, so that the lead/lag relationship could be discerned only from the ongoing features of the envelopes. The owls behaved as though they experienced localization dominance at short delays, but only if the sounds were deeply amplitude modulated, suggesting the importance of the ongoing envelope. Correspondingly, neurons in the space map fired preferentially to the leading sound, thus paralleling the behavior, but again, deep amplitude modulations were necessary. Analyses of waveforms at the ears suggest that the acoustic features that caused the space map neurons to fire were more abundant for the leading than for the lagging stimuli, thus explaining localization dominance without the need for explicit echo-suppression circuitry. [Work supported by NIDCD F32-DC008267 and RO1-DC003925.]

10:10—10:30 Break

10:30

1aABb6. Coding of sound features in primary and secondary auditory areas of songbirds. Frederic E. Theunissen (Dept. of Psych., Univ. of California Berkeley, 3210 Tolman Hall, Berkeley, CA 94720)

We have used a combination of neuroethological and classical auditory neurophysiological approaches to study how behaviorally relevant sounds are processed in the avian auditory system. One of the contributions of the neuroethological approach has been the discovery of highly specific auditory neurons that appear to be specialized for detecting very specific behaviorally relevant sounds. On the other hand, many auditory neurons recorded in the auditory system of nonspecialists do not exhibit such specificity. At the same time, animals and humans hear and process a large space of sounds and are able to categorize these in much broader perceptual terms, describing them in terms of their pitch, timbre, and rhythm. By systematic analyzing neural responses to song in the ascending avian auditory system and relating receptive fields to the statistics of natural sounds, we have shown that these two approaches can be unified: we found that the spectrotemporal receptive fields of auditory neurons tile a subset of the acoustical space that is particular important for natural sounds. In addition, we found that neurons could be classified into functional clusters. Neurons in different clusters were sensitive to different song features and, we will argue, are involved in mediating distinct perceptual attributes.

10:55

1aABb7. Sound localization acuity in the cat: Integrating data from psychophysical, physiological, and acoustical cue studies. Daniel J Tollin (Dept. of Physio., Univ. of Colorado Med. Sch., 12800 E. 19th Ave., Aurora, CO 80045, daniel.tollin@uchsc.edu)

The minimum audible angle (MAA), a psychophysical measure of auditory spatial acuity, is the smallest angle separating the locations of two sound sources, presented sequentially, that can be reliably discriminated. In humans and cats, the MAA for tone stimuli depends jointly on source azimuth and tone frequency. It is hypothesized that this results from the spatial and frequency dependence of the acoustical cues to location. Thus, for high-frequency ($>$1.5 kHz) stimuli, where the predominant cue for source azimuth is the interaural level difference (ILD), the MAA would be determined by the spatial and frequency dependence of the ILD cue. However, for this to be true, the underlying neural acuity for ILD cues must be frequency independent. Here the hypothesis was tested that the frequency dependence of the psychophysical MAA in cats can be explained by the frequency dependence of the ILD cue with source azimuth and the frequency independence of the acuity of neurons (or neural MAA) in the lateral superior olive (LSO) to ILDs. The LSO is a brainstem structure responsible for the initial encoding ILDs. The data from acoustical cue measurements and physiological and psychophysical studies in cats support this hypothesis. [Work supported by NIH Grants Nos. DC006865 and DC02840.]

11:20

1aABb8. Some brain mechanisms for auditory scene analysis. John C. Middlebrooks, Chen-Chung Lee (Dept. of Otolaryngol., Univ. of California, Irvine, CA 92697-5310, j.midd@uci.edu), and Ewan A. Macpherson (Univ. of Western Ontario, London, ON Canada, N6G 1H1)

Detection and discrimination of sounds in complex auditory environments is facilitated by knowledge of the locations of sound sources and by spatial separation of signals and maskers. In studies of the spatial sensitivity of auditory cortical neurons in anesthetized cats, we find that location-dependent variation in temporal spike patterns can signal sound-source locations throughout up to 360 deg of space. In awake cats, spatial receptive fields can be similarly broad, although cortical neurons show a greater diversity of temporal patterns than is seen in anesthetized conditions. Our recent results demonstrate that spatial tuning tends to sharpen when a cat is engaged in an auditory task, compared to idle conditions, and that tuning can sharpen even further when the task involves sound localization. In human psychophysical studies, spatial hearing permits perceptual "stream segregation" of multiple temporally interleaved sequences of sounds. In parallel studies of the auditory cortex in anesthetized cats, we find that interleaved sound sequences, in configurations that likely would be segregated perceptually, result in activation of discrete cortical neuron populations. The spatial acuity for this neuronal stream segregation is substantially sharper than has been observed in studies of cortical representation of single sound sources.

MONDAY MORNING, 18 MAY 2009 BROADWAY I/II, 10:25 A.M. TO 12:00 NOON

Session 1aAO

Acoustical Oceanography, Underwater Acoustics, and Signal Processing in Acoustics: Temporal and Spatial Field Coherence Applied to Ocean Sensing: Measurement, Theory, and Modeling I

Timothy F. Duda, Chair
Woods Hole Oceanographic Inst., Woods Hole, MA 02543-1053

Chair's Introduction—10:25

Invited Papers

10:30

1aAO1. On the use of broadband acoustic scattering techniques for observing physical and biological scattering features in the ocean. Andone Lavery (Woods Hole Oceanograph. Inst., Dept. of Appl. Ocean Phys. and Eng., MS11, Woods Hole, MA 02543)

Broadband acoustic scattering techniques, and associated signal processing techniques, are emerging as a powerful tool for the remote sensing of physical and biological scattering features in the ocean. These techniques allow (1) the frequency spectrum of oceanic scatterers to be measured, thereby resulting in enhanced classification and quantification potential relative to more traditional single-frequency scattering techniques, and (2) increased spatial resolution resulting in high-resolution images of oceanic scattering features, obtained through use of pulse compression signal processing techniques that exploit the broadband content of the signal. In this presentation, broadband processing techniques are discussed and results from laboratory and field measurements of broadband acoustic scattering from both physical and biological scatterers, spanning the frequency range from 100–600 kHz, are presented. Controlled laboratory broadband scattering measurements allow coherent ping-to-ping analysis to be performed as well as individual scattering features to be determined. The laboratory approaches are then expanded into the field to investigate physical and biological scattering features in the ocean on relevant temporal and spatial scales. Examples of broadband field measurements are presented from recent field experiments, and include scattering from nonlinear internal waves, zooplankton, turbulent oceanic microstructure, salinity microstructure in rivers, and bubbles.

10:50

1aAO2. Coherence in long-range deep ocean propagation. Michael G. Brown, Francisco J. Beron-Vera (RSMAS, Univ. of Miami, 4600 Rickenbacker Cswy., Miami, FL 33149), Ilya A. Udovydchenkov, and Irina I. Rypina (Woods Hole Oceanograph. Inst., Woods Hole, MA 02543)

Acoustic wavefield structure, stability, and coherence properties in deep ocean environments will be discussed. Properties of transient wavefields will be emphasized. Observations, simulations, and some theoretical considerations will be presented. Some important experimental milestones, beginning approximately in 1979, will be noted. It will be argued that the last 30 years have been characterized by a number of pleasant surprises relating to the stability and coherence of transient wavefields in deep ocean environments. [Work supported by ONR.]

11:10

1aAO3. Transverse coherence lengths, processing limits and implications. William M. Carey (Dept. of Mech. Eng., College of Eng., Boston Univ., Boston, MA 02215)

How does scattering in sound channels (deep and shallow waters) limit coherent array processing or what is the limitation of resolution in terms of the mutual coherence function and its temporal and spatial coherence lengths? The resolution of an array is limited by the mutual coherence function; but estimation in a partially coherent noise background with a multipath signal is difficult using the normalized cross power spectral density, magnitude squared coherence, because of the properties of both signals and noise. The measurement of magnitude-squared coherence is a poor statistical estimator since it is a function of the signal-to-noise ratio and multipath interference with large confidence bounds. Array gain measurements and a wave-theoretic coherence functional form can provide estimates of temporal and spatial coherence lengths defined as the 1/*e* value of this function. This paper reviews single path coherence results and those derived from array measurements over the low- to mid-frequency range in deep and shallow water. Representative coherence lengths are discussed in terms of boundary interactions, internal wave scattering, and coastal mesoscale features. The implications for arrays used to estimate geoacoustic properties, mammal locations, and scattering from the boundaries are presented.

Contributed Papers

11:30

1aAO4. Comparing the effects of internal waves, mode coupling, and change in bandwidth of a radiated signal on low mode energy propagation. Natalie Grigorieva, Gregory Fridman (Dept. of Appl. Math., St. Petersburg State Marine Tech. Univ., 3 Lotsmanskaya Str., St. Petersburg 190008, Russia, nsgrig@natalie.spb.su), James Mercer, Rex Andrew, Michael Wolfson (Univ. of Washington, Seattle, WA 98105), Bruce Howe (Univ. of Hawaii, Honolulu, HI 96822), and John Colosi (Naval Postgrad. School, Monterey, CA 93943)

Wideband signal propagation modeling is carried out for the actual space-time configuration realized during the long-range ocean acoustic propagation experiment (LOAPEX) conducted in the North Pacific in 2004. In particular, the experiment used the Kauai transmit station that was located at a range of 2432 km from a vertical line array (VLA), M-sequence at 75 Hz, and a source depth of 800 m close to the depth of a sound-channel axis. Two sound speed profiles were utilized to get the smooth two-dimensional sound speed field. The first one was the profile obtained from the conductivity-temperature-depth (CTD) measurements at the Kauai station. The second sound speed profile from the VLA location was based on a CTD cast taken at station T50 (50 km from the VLA). To take into account the acoustic fluctuations due to internal waves, the buoyancy frequency profile based on LOAPEX CTD measurements made at station T50 was used. For simulations, 512 values of the horizontal wave number were utilized with the maximal mode number equal to 50. The low modes intensity received by the hydrophones close to the sound channel axis at the VLA is compared with the results of simulations. [Work supported by ONR and ONR Global.]

11:45

1aAO5. On the time-mean state of ocean models and the properties of long-range acoustics. Brian D. Dushaw (Appl. Phys. Lab., Univ. of Washington, 1013 N.E. 40th St., Seattle, WA 98105, dushaw@apl.washington.edu), Matthew A. Dzieciuch, and Peter F. Worcester (Univ. of California at San Diego, La Jolla, CA 92093-0225)

Global and regional circulation models of the ocean have the horizontal and vertical resolution required for realistic simulation of long-range acoustic propagation. These simulations offer a novel test of model accuracy, because acoustic propagation is sensitive to the vertical gradients of sound speed in the models and acoustic arrival patterns are known to have stable and universal properties. The travel time and dispersal of an arrival pattern are examples of such properties. Acoustic receptions on three long vertical line arrays from basin-scale transmissions in the North Pacific in 1996 and 1998 are used to test the acoustical properties of the time-mean state of several ocean models. The NOAA World Ocean Atlas, a global representation of the "average" ocean, yields acoustic predictions whose patterns closely match the observations, but not all ocean models accurately represent oceanic sound speed properties. Acoustical tests of a model do not necessarily require actual data, because basic acoustical properties may be derived from the World Ocean Atlas.

MONDAY MORNING, 18 MAY 2009 PARLOR B/C, 8:00 A.M. TO 12:00 NOON

Session 1aNS

Noise and Architectural Acoustics: Prediction and Control of Noise Related to Buildings

James E. Phillips, Chair
Wilson Ihrig & Assoc Inc., 5776 Broadway, Oakland, CA 94618-1531

Contributed Papers

8:00

1aNS1. Research into the human response to vibration in residential environments. David C. Waddington, Andrew T. Moorhouse (Acoust. Res. Ctr., Univ. of Salford, Salford M5 4WT, UK, d.c.waddington@salford.ac.uk), and Phil A. Brown (Univ. of Salford, Salford M5 4WT, UK)

This paper describes work being carried out at the Univ. of Salford to develop a method by which human annoyance to vibration in residential environments can be assessed. The objective of this study is to yield a robust relationship between vibration exposure and human response; therefore, providing a reliable basis for the development of standards and guidance for the assessment of vibration in residential buildings. The vibration sources to be considered are those affecting residents that are outside their control, such as construction, road, and rail activities. The protocol involves the measurement of vibration outside and inside residences and a social study questionnaire based on face-to-face interviews with adults. The project is expected to capture a wide-scale study of 2000 case studies, and examples of early field measurements and results are presented. This work is likely to be of interest to planners and practitioners involved in the design of buildings and environmental health officers involved in the assessment of vibration complaints. [Work funded by the Dept. for Environment, Food, and Rural Affairs (Defra) UK.]

8:15

1aNS2. Comparison of tapping machines for impact insulation class (IIC) measurement accuracy. Aaron M. Farbo (Cavanaugh Tocci Assoc., 327F Boston Post Rd., Sudbury, MA 01776, afarbo@cavtocci.com)

This study compares tapping machines from different manufacturers and age for impact insulation class (IIC) measurement accuracy. In addition to the IIC measurement accuracy, the study also compares the impact frequency and effect of different impact surfaces (steel and rubber). The tapping machines were tested in a model condo unit in accordance with impact insulation class testing standards ASTM E1007-04 and ISO 140-6. These standards clearly state the impact frequency and type of impact surface to be used (hardened steel). Previous testing standards did not specify the material type for impact surfaces and many older tapping machines included both steel and rubber feet. The results of this study will be presented to show the effect of age, impact frequency, and impact surface on the resulting IIC measurement for the tapping machines.

8:30

1aNS3. Investigation of damping materials for heavy-weight floor impact sounds in concrete structures. Jin Yong Jeon and Seung Yup Yoo (Dept. of Architectural Eng., Hanyang Univ., Seoul 133-791, Korea)

Multi-layers of surface materials were investigated for the isolation of floor impact sounds in concrete structures. Both resilient and viscoelastic damping materials were tested according to the modified method of ANSI S2.22 for calculating elastic modulus and loss factor. The dynamic characteristics of the concrete structures subjected to heavy-weight floor impacts were modeled by the compressional damping theory, using sandwiched beams. Results show that the viscoelastic damping materials are more effective than the resilient materials. In field measurements, the insertion loss of the heavy-weight impact sounds, which was mainly affected by the modal characteristics at low frequencies, was higher in floor structures with the damping materials.

8:45

1aNS4. Ceiling details in wood frame construction. James Keene (Keene Bldg. Products, 5910 Landerbrook Dr., Mayfield Hts., OH 44124, jrk @keenebuilding.com)

In multifamily projects, the details of floor/ceiling installation have been a known and constant problem. Errors when installing noise control elements have caused lots of failures. Specifically, gypsum board ceiling and wall isolation with caulking is often short circuited when the gaps are incorrect. This paper details a new method of installing gypsum board ceilings and walls to provide a more easily inspected and detail assured isolation system. The system applies a similar system for ceiling installation as has become the standard in floor perimeter isolation. An "L" shaped fabric strip is installed around the perimeter of the room to create the gaps that today are created by the contractor in the field. The paper and presentation will review the methodolgy of the system and the laboratory test results that show performace compared to the present approach. The fire rated test performance will also be summarized. Since this is multifamily, the economic savings of this system are also presented.

9:00

1aNS5. Comparison of apparent field sound transmission class (AFSTC) and outdoor indoor transmission class (OITC) as applied to modular classroom design and field analysis. Norman H. Philipp and Lily M. Wang (Architectural Engr. Prog., Peter Kiewit Inst., Univ. of Nebraska-Lincoln, 1110 S. 67th St., Omaha, NE 68182-0681, nphilipp@mail.unomaha.edu)

Regarding the drafted addendum for ANSI S12.60-2002 Standard on Classroom Acoustics Guideline, it was recently proposed to utilize Outdoor Indoor Transmission Class as a means of rating modular classroom exterior envelope noise reduction performance. This paper provides a comparative analysis of Outdoor Indoor Transmission Class with the Apparent Field Sound Transmission Class. Measurements were conducted in modular classrooms currently in use by the Omaha Public School District in Omaha, NE, along with newer construction units prior to installation on-site. Limitations to both measurement methodologies are discussed in relation to their source noise spectrums.

9:15

1aNS6. Acoustical analysis of an indoor test facility for a 30-mm Gatling gun. Matthew D. Shaw and Kent L. Gee (Dept. of Phys., Brigham Young Univ., N283 ESC, Provo, UT 84602, mdshaw16@gmail.com)

The Air Force commissioned a construction and engineering company to build an indoor test facility for the GAU-8 Avenger at Hill Air Force Base in Layton, UT. The blast pressures from this 30-mm Gatling gun, however, are large enough to cause spalling of the concrete walls over time. The facility is being designed and constructed to last for 20 yrs, requiring several acoustical treatments. The pressures from the gun were measured outdoors, with maximum pressures exceeding 3000 Pa (163 dB) at a distance of 30 ft (9.1 m). A computer model of the room was designed using EASE, and impulse responses were generated at several positions. These impulse responses were convolved with an ideal blast wave pulse train to mimic the sound of the gun in the room. From these data and results collected from preliminary tests in the range, recommendations have been provided as to placement and types of necessary treatments. [Work supported by HHI Corporation.]

9:30

1aNS7. The effective bandwidth of the enhanced transmission loss across a wall with Helmholtz resonators. Joel Garrelick (Appl. Physical Sci., Inc., 49 Waltham St., Lexington, MA 02421, jgarrelick@aphysci.com)

It is long known that the introduction of Helmholtz resonators can enhance the transmission loss of a wall in the vicinity of their natural frequencies. Resonance is typically achieved by creating open-mouthed cavities in the wall. The effective bandwidth of the phenomenon is a key factor in its practical utility. It is dependent on the internal resonator resistance as well as the *in situ* external acoustic impedance at the cavity mouth. This paper is an analysis of the bandwidth for a variety of resonator configurations, including cavities with septa and differing volumes. The mathematical model, driven by a normally incident plane wave, will treat the wall as rigid but moveable, and the cavities will be infinitely periodic arrays. Influences of wall impedance, resonator resistance, cavity array spacing and natural frequencies will be described. In addition, the potential for minimizing the deleterious effects of diffraction between neighboring cavities by placing the cavity mouths on both sides of the wall will be assessed.

9:45

1aNS8. Vertical profile of community noise in high-rise built environment. Alam Sheikh Mahbub, Lee Siew Eang (Dept. of Bldg., Natl. Univ. of Singapore, Singapore), and W. L. H. Johnny (Housing and Development Board)

High-rise public housing apartments in Singapore are often subjected to different types of community noise sources in close proximity. Community noise in housing estates comprises noise sources such as food center noise, children playground noise, soccer court noise, basketball court noise, waste disposal truck noise, etc. It was found as one of the major noise source categories in residential environment. With the enhanced performance quality in many aspects of public housing in recent years, it is imperative that the acoustics performance of the apartments can be enhanced to match overall quality of public housing environment through appropriate characterization of these noise sources. A scientific and reliable approach for characterizing noise propagation from these sources is vital in the implementation of projects. Software modeling and simulation were carried out to study noise propagation characteristics of five types of community noise sources. Five 16-storey residential buildings were investigated with noise measurement from these sources, conducted on each level of the buildings. The predicted results were verified with field measured data. This study observes that noise level from these sources reduces between 3.7 and 12.8 dBA at a height between 43 and 46.5 m above the ground with the increase in building height.

10:00

1aNS9. Prediction of the reverberation times and sound pressure levels in town streets. Kai Ming Li (Ray W. Herrick Lab., Purdue Univ., 140 S. Martin Jischke Dr., W. Lafayette, IN 47907-2031) and Chenly Y. C. Lai (Hong Kong Polytech. Univ., Kowloon, Hong Kong)

The current study was devoted to investigate the propagation of noise along a town street where building façades and road surfaces form a street canyon. An energy based approach was used to estimate reverberation times and the sound pressure levels due to a point source located along the town street. An image source method was developed to compute sound energy in this semienclosed space. However, the image source model can be simplified leading to the numerical solutions expressed in an integral formulation. The absorption coefficients of the boundary surfaces and the air absorption factor can be included in the present analysis. The integral can further be evaluated analytically with the sound pressure levels expressed in a closed form solution. Both outdoor full scale measurements and indoor scale model experiments were conducted and compared with the computed results. The numerical results due to the proposed formula not only agree well with the experimental measurements but were also in accord with the published data.

10:15—10:30 Break

10:30

1aNS10. Modeling sound transmission through large rectangular openings in a damped enclosure. Buye Xu and Scott D. Sommerfeldt (Dept. of Phys. and Astron., Brigham Young Univ., Provo, UT 84602)

The understanding of sound transmission through large rectangular openings in a damped enclosure plays an important role in designing noise reduction solutions where enclosures and openings are often involved. A complete solution for this problem should consider diffraction, coupling between exterior and interior sound fields, and damped boundaries, each of which is a challenging task. A theoretical model will be introduced in this paper that deals with all these phenomena and, in addition, also considers the effects of the direct sound field in the solution. Numerical simulations and experimental results will be presented and analyzed.

10:45

1aNS11. Prediction of sound propagation in rectangular ducts by a ray-tracing model with phase. Behrooz Yousefzadeh and Murray Hodgson (Acoust. and Noise Res. Group, Univ. of British Columbia, 3rd floor, 2206 East Mall, Vancouver, BC V6T 1Z3, Canada, behroozy@interchange.ubc.ca)

A ray-tracing model which includes phase changes due to distance traveled and surface reflection has previously been developed for predicting sound field in rectangular rooms. Including phase has removed the limitation associated with energy-based models, that room dimensions must be larger than the sound wavelength. The model has been applied to a rectangular room with dimensions not large compared to the wavelength. The room dimensions are that of a real-world ventilation duct, with the boundary conditions idealized as a point source at one end and anechoic termination at the other end. Predictions have been made of the pressure field inside the duct and the results have been compared with analytical solutions. The model is found to be capable of predicting the modal characteristics of the duct.

11:00

1aNS12. Assessing the origin of tonal sound emanating from refrigerant piping in the historic Mattress Factory project at the University of Washington Tacoma Campus. Elizabeth Bogomolov and Julie A. Wiebusch (The Greenbusch Group, Inc., 1900 West Nickerson St., Ste. 201, Seattle, Washington 98119)

UW Tacoma campus is located in the Union Depot Warehouse National Historic District. Siting the campus in this area and renovating rather than demolishing some of the remaining historic, industrial buildings as campus classrooms and offices has stimulated the city's urban renewal. Mattress Factory was one of these historic structures, regulated by the Tacoma Landmarks Commission. The new chilled water system in Mattress Factory consists of two 70-ton chillers located in a mechanical room on the lowest level of the four-story building. Air-cooled condensing units, located on the roof, connect to the chiller compressors. The refrigerant piping was required to be incorporated within the building envelope so as not to alter the appearance of the historic exterior. The piping path between the compressors and condensers was somewhat circuitous due to existing structural elements. A discernable, tonal whine was exhibited in spaces on the second and third floors when the chiller was running. The sound corresponded with the location of the refrigerant piping within the wall cavity, with the dominant sound level at the second floor in the area of an existing beam. This paper outlines the process of identifying the cause of the sound and the ultimate mitigation treatment.

11:15

1aNS13. Noise from rooftop mechanical equipment in residential buildings. Anthony Nash (Charles M. Salter Assoc., 130 Sutter St., Ste. 500, San Francisco, CA 94104, anthony.nash@cmsalter.com)

Community planning codes restrict building heights with the intent that the building is in general conformance with the character of the neighborhood's built environment. Although these codes prescribe the maximum height, they do permit a limited amount of rooftop space to be used for elevator penthouses, exhaust fans, cooling towers, and miscellaneous other rotating mechanical equipment. Thus, the stage is set for a potential acoustical conflict between the need for building mechanical services on the rooftop above and the expectations of people occupying residences below (ironically, the most expensive space in the building). This paper discusses a number of design and retrofit projects involving rooftop equipment installations over noise-sensitive spaces. The cases include rooftop equipment such as small corridor exhaust fans, rotary chillers, traction elevators, etc. The paper describes both the noise generating mechanisms and transmission paths as well as techniques for mitigating mechanical equipment noise into occupied spaces.

11:30

1aNS14. Active feedback control of broadband noise from a small axial fan. Cole V. Duke, Scott D. Sommerfeldt, and Kent L. Gee (Dept. of Phys. and Astron., Brigham Young Univ., N283 ESC, Provo, UT 84602)

In the past, significant progress has been made in actively controlling the tonal noise of axial cooling fans. Current work focuses on the control of broadband fan noise, which presents different challenges. Because the broadband noise is not deterministic, a feedback system will be used, and the total system delay must be reduced as much as possible. To ensure stability of the closed-loop control system, the phase and magnitude characteristics of the filter must be chosen to produce attenuation in the target frequency band without augmenting the noise outside the band. A practical analog feedback controller will be presented, and results will be shown for a single input single output control system, as well as a configuration using multiple independent control systems.

11:45

1aNS15. The prediction of airflow generated noise in ventilation system bends. David C. Waddington (Acoust. Res. Ctr., Univ. of Salford, Salford M5 4WT, UK, d.c.waddington@salford.ac.uk) and David J. Oldham (Univ. of Liverpool, Liverpool L69 3BX, UK)

This paper describes recent work on the prediction of airflow generated noise in ducts based upon pressure loss characteristics. Strategies employed to reduce turbulence by smoothing the flow around a bend include the use of turning vanes, and a large radius of curvature with the associated use of splitters. The effect of these strategies on the regenerated noise is examined for a range of common bends including mitred bends, with and without turning vanes, and radiussed bends, with and without splitters. The effect of bends with different aspect ratios is also examined. An excellent collapse of measured data onto a single characteristic curve is reported for each configuration based upon standard values of pressure loss coefficients. It is suggested that that these characteristic curves might form the basis for a practical prediction method.

Session 1aSCa

Speech Communication: Articulatory Speech Synthesis and Robotic Speech

Michael C. Brady, Cochair
Dept. of Cognitive Science, Univ. of Indiana, Bloomington, IN 47406

Frank H. Guenther, Cochair
Dept. of Cognitive and Neural Systems, Boston Univ., Boston, MA 02215

Invited Papers

8:30

1aSCa1. Three-dimensional mechanical tongue and vocal cord models for human speech production. Kotaro Fukui, Eiji Shintaku, Yuma Ishikawa, Nana Sakakibara, Yoshikazu Mukaeda, Atsuo Takanishi (Dept. of Modern Mech. Eng., School of Creative Sci. and Eng., Waseda Univ., 3-4-1, Ookubo, Shinjuku-ku Tokyo, 196-8555, Japan, kotaro@toki.waseda.jp), and Masaaki Honda (Waseda Univ., Saitama, Japan)

We have developed talking robot Waseda Talker series, which mimicked human speech production mechanism. This robot is consisted of mechanical vocal cords, tongue, lips, and nasal cavity. The current version, WT-7R (Waseda Talker No. 7 Refined), has 16 DOF (7 DOF in tongue, 1 DOF in jaw, 4 DOF in lips, 1 DOF in velum, 2 DOF in vocal cords, and 1 DOF in lung), having same vocal tract length as average adult male. The mechanical vocal cord model is made of styrene based thermoplastic elastomer "Septon" and the shape is mimicking vocal folds of human with three dimensions. The vocal code model has pitch control mechanism by adjusting the cord tension and glottal opening-closing mechanism, and it could reproduce the vocal cord vibration of the vocal fry (double pitched) and the breathy voice (unclose in one cycle) as well as the modal voice with variable pitch. The three-dimensional tongue model is constructed by the rigid link mechanism covered by the Septon soft rubber. The tongue model could be controlled so as to configure the tongue shape in a three-dimensional way. We will describe the details of the mechanism of talking robot by showing some demonstration videos.

8:50

1aSCa2. Gesture controlled synthetic speech and song. Sidney Fels (Elec. and Comput. Eng., Univ. of British Columbia, 2332 Main Mall, Vancouver, BC, Canada, V6T 1Z4, ssfels@ece.ubc.ca), Bob Pritchard (Univ. of British Columbia, Vancouver, BC, Canada V6T 1Z2), Eric Vatikiotis-Bateson (Univ of British Columbia, Vancouver BC, V6T 1Z4 Canada, and The Visual Voice Team)The Visual Voice Team

We describe progress on creating digital ventriloquized actors (DIVAs). DIVAs use hand gestures to synthesize audiovisual speech and song by means of an intermediate conversion of hand gestures to articulator (e.g., tongue, jaw, lip, and vocal chords) parameters of a computational three-dimensional vocal tract model. Our parallel-formant speech synthesizer is modified to fit within the MAX/MSP visual programming language. We added spatial sound and various voice excitation parameters in an easy-to-use environment suitable for musicians. The musician's gesture style is learned from examples. DIVAs will be used in three composed stage works of increasing complexity performed internationally, starting with one performer initially and culminating in three performers simultaneously using their natural voices as well as the hand-based synthesizer. Training performances will be used to study the processes associated with skill acquisition, the coordination of multiple "voices" within and among performers, and the intelligibility and realism of this new form of audio/visual speech production. We are also building a robotic face and computer graphics face that will be gesture controlled and synchronized with the speech and song. [This project is funded by the Canada Council for the Arts and Natural Sciences and Engineering Research Council, Canada. More information is at: www.magic.ubc.ca/VisualVoice.htm]

9:10

1aSCa3. Biomimetic vocal tract modeling: Synthesis of speech articulation. Robin Hofe and Roger K. Moore (Univ. of Sheffield, Dept. of Comp. Sci. Regent Court, 211 Portobello, S1 4DP, Sheffield, UK, R.Hofe@sheffield.ac.uk)

In recent years, a number of researchers have started to investigate cognitive aspects of speech production using "speaking robots." This paper introduces AnTon–the Animatronic Tongue and vocal tract model developed at the University of Sheffield, UK. AnTon's distinctive feature is its design principle. Whereas other speaking robots are motivated by functionality—to produce a good imitation of human speech—AnTon mimics human anatomy, including tissue properties and muscle (actuator) configurations. Any ability to produce speech sounds emerges solely as a consequence from that anatomy. Such imitation of a biological system is called "biomimetic." The reason to choose a biomimetic approach lies in the need for an experimental tool that makes speech articulation processes transparent. The use of realistic muscle configurations means that the model's speech gestures can be mapped directly to human behavior, facilitating the evaluation of both the quality of AnTon's speech gesture reproductions as well as any predictions about human behavior inferred from experiments with AnTon. AnTon will be used to clarify speech articulation processes, specifically in relation to energy efficiency of articulatory gestures and the importance of an auditory feedback loop. The project's progress is continually documented on http://www.dcs.shef.ac.uk/~robin/anton/anton.html.

9:30

1aSCa4. Synthesizing vowel transitions with an analog vocal tract. Michael C. Brady (Dept. of Cognit. Sci. Indiana Univ., 1910 E. 10th St., Bloomington, IN 47406, mcbrady@indiana.edu)

A mechanical vocal tract with a pneumatic sound source, silicone tongue, and lip rounding mechanism is introduced. The tract is designed to make controlled transitions between static articulatory vowel configurations. The focus on transitions is important because many argue that it is the change between steady-state sounds that the nervous system is tuned for in extracting information from the speech signal. I draw on examples and experimental results as I review this steady state versus transition distinction. The notion of articulatory transitions as the motor control targets of robotic speech production is then discussed. Video demonstrations of the mechanical tract helps to illustrate how transitions as control targets may be implemented. In conclusion, I argue for why an analog vocal tract with its true aerodynamics (as opposed to synthesis using a cascade of digital filters) is called for in generating articulator-transition-based speech sounds.

9:50

1aSCa5. A talking robot and the interactive speech training for hearing-impaired. Hideyuki Sawada (Dept. of Intelligent Mech. Syst. Eng., Faculty of Eng., Kagawa Univ., 2217-20 Hayashi-cho, Takamatsu, Kagawa, 761-0396, Japan, sawada@eng.kagawa-u.ac.jp)

A speech trainer employing a talking robot for auditory-impaired people will be introduced. The talking robot consists of mechanically-designed vocal organs such as a vocal tract, a nasal cavity, artificial vocal cords, and a sound analyzer with a microphone system, and the mechanical parts are fully controlled by ten servomotors for generating humanlike voices. The robot autonomously learns the relation between motor control parameters and the generated vocal sounds by an auditory feedback control, in which a Self-organizing neural network is employed for the adaptive learning. By employing the robot and its properties, the authors have constructed an interactive speech training system. The training is divided into two approaches; one is to use the robot for showing the shape and the motion of the vocal organs, and the other is to use a topological map for presenting the difference of phonetic features of a traine's voices. While referring to the robot motions and the phonetic characteristics, a trainee is able to interactively practice vocalization for acquiring clear speech with appropriate speech articulations. To assess the validity of the training system, a practical experiment was conducted by 19 subjects in a school for the deaf children. The talking robot is expected to intensively teach auditory-impaired patients the clear speech articulations.

10:10

1aSCa6. Real-time speech synthesis for neural prosthesis. Frank H. Guenther (Dept. of Cognit. & Neural Syst., Boston Univ., 677 Beacon St., Boston, MA 02215)

The idea of building an artificial vocal tract that could serve as a prosthetic device for mute individuals has been pursued since the pioneering work of Wolfgang von Kempelen and contemporaries in the 18th century. These early speaking machines were built out of wood, metal, rubber, and other materials fashioned into a mechanical vocal tract whose shape was manipulated manually by the user while air was forced through it. In the mid-20th century, Gunnar Fant and colleagues constructed electrical resonating circuits capable of producing speechlike sounds over loudspeakers, and shortly thereafter fully computer-based speech synthesizers were developed, including formant synthesizers and articulatory synthesizers. The recent development of brain-computer interfaces driven by neuronal activity raises the possibility of a speech prosthesis that would allow mute paralyzed individuals to directly control a speech synthesizer to create speech output. We have implemented such a brain-computer interface in a 25-year-old volunteer suffering from locked-in syndrome, characterized by complete loss of movement but intact perception and consciousness. To date the volunteer has successfully used the prosthesis to produce vowel sounds. Future developments in speech synthesis necessary for allowing full conversational speech via brain-computer interface will be outlined. [Work supported by NIH/NIDCD and NSF.]

Session 1aSCb

Speech Communication: Speech Production and Articulatory Models (Poster Session)

Michael C. Brady, Cochair
Dept. of Cognitive Science, Univ. of Indiana, Bloomington, IN 47406

Frank H. Guenther, Cochair
Dept. of Cognitive and Neural Systems, Boston Univ., Boston, MA 02215

Contributed Papers

All posters will be on display from 10:30 a.m. to 11:30 a.m. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 10:30 a.m. to 11:00 a.m. and contributors of even-numbered papers will be at their posters from 11:00 a.m. to 11:30 a.m.

1aSCb1. Response of synthetic vocal fold models with geometry based on visible human project data. Brian A. Pickup and Scott L. Thomson (Dept. of Mech. Eng., Brigham Young Univ., 435 CTB, Provo, UT 84602, thomson@byu.edu)

Numerous synthetic and computational models have been and are currently being used in research studies of human vocal fold vibration. Model geometry plays an integral role in governing dynamic response. However, the various model geometry definitions have typically been idealized and often exhibit wide variability with each other. The present research compares the response of synthetic vocal fold models of different geometries. One model is based on idealized geometry, while the other is based on geometry obtained from the National Library of Medicine's visible human project (VHP). The process for image extraction, model definition, and model fabrication is described, including: (1) image conversion from 2-D VHP image sequences to 3-D stereolithography (STL) format, (2) conversion to 3-D computer model format in which model geometric manipulations can be performed, and (3) fabrication of synthetic models using rapid prototyping. Results of measurements to characterize the dynamic response of self-oscillating synthetic vocal fold models, including onset and offset pressure, instantaneous glottal width using high-speed imaging, and glottal jet velocity profiles using particle image velocimetry (PIV), are presented for models based on both VHP data and idealized geometries. The sensitivity of the models to geometry changes is also reported.

1aSCb2. Analysis of recent experimental studies of phonation threshold pressure done with a physical model of the vocal fold mucosa. Lewis P. Fulcher (Dept. of Phys. and Astronomy, Bowling Green State Univ., Bowling Green, OH 43403), Ronald C. Scherer, Chen Liu, and Marco Nardone (Bowling Green State Univ., Bowling Green, OH 43402)

The pioneering work on the physics of small amplitude oscillations of the vocal folds by Titze resulted in a simple analytic formula that predicted the phonation threshold pressure of the uniform glottis to be a linear function of the glottal half-width, to be directly proportional to energy dissipation properties of the vocal fold, and to be reduced by the presence of a vocal tract. These relationships have been examined in a series of experiments [R. Chan and I. Titze, "Dependence of phonation threshold pressure on vocal tract acoustics and vocal fold tissue mechanics," J. Acoust. Soc. Am. **119**, 2351–2362 (2006)], and the linear connection with glottal half-width has been observed in several cases. However, the analysis described above did not consider the role of viscosity of the air, which may be important at small widths, as suggested by Lucero. The present study examines these experiments in the context of the surface wave model developed by Titze using intraglottal pressure distributions taken with the static physical model of the glottis M5. The M5 data include effects of viscosity and of flow separation. Preliminary results indicate that the threshold pressure should decrease as glottal width decreases, over the entire range of glottal widths examined by Chan and Titze. [Work supported by NIH R01DC03577.]

1aSCb3. Nonlinear viscoelastic response of the vocal fold lamina propria under large-amplitude oscillatory shear. Roger W. Chan (Otolaryngol., Biomed. Eng., Univ. of Texas Southwestern Med. Ctr., Dallas, TX 75390) and Miwako Kimura (Univ. of Texas Southwestern Med. Ctr., Dallas, TX 75390)

Viscoelastic properties of the vocal fold lamina propria have been reasonably quantified beyond the linear range for describing tissue behavior under uniaxial stretch or tensile deformation, but the same cannot be said for oscillatory shear deformation, which is more relevant for understanding the mechanics of large-amplitude vocal fold vibration. Previous studies reporting the viscoelastic shear properties of vocal fold tissues have been limited to characterization of the small-amplitude viscoelastic response in the linear viscoelastic region, partly due to the fact that derivation of the elastic and viscous shear moduli is based on the assumption of linearity, and also because of the lack of a coherent framework for describing any nonlinearities beyond the linear viscoelastic region. Based on a recently proposed rheological framework for quantifying such nonlinearities, this study examined the viscoelastic response of human vocal fold cover specimens subjected to large-amplitude oscillatory shear, up to a shear strain amplitude of around 1.0. Results indicated that the linear viscoelastic moduli (G' and G'') cannot adequately describe the tissue response under large-strain shear deformation, whereas geometrical interpretations of Lissajous–Bowditch curves could unveil nonlinearities that are obscured by the use of G' and G'', such as the phenomenon of strain stiffening.

1aSCb4. Effects of the airway surface liquid on vocal fold vibration. David Daily, Dan Lo Forte, and Scott L. Thomson (Dept. of Mech. Eng., Brigham Young Univ., 435 CTB, Provo, UT 84602)

A two-layer airway surface liquid (ASL), comprised of a Newtonian sublayer and a non-Newtonian surface layer, lines the laryngeal lumen. During phonation the layers on opposing vocal folds experience merging, squeezing, stretching, and rupture. Various aspects of the ASL have been studied, but its role in vocal fold vibration remains largely understudied. This presentation describes the results of experiments and computational simulations aimed at improving our understanding of the role of the ASL, in particular the non-Newtonian layer, in phonation. The experimental setup for measuring the liquid properties is described, and results are presented. The measured liquid properties are input into a finite element model of the vocal folds, which includes a simulated non-Newtonian layer, using the software package ADINA. The non-Newtonian liquid is also incorporated into a two-mass vocal fold model, which includes the effects of the ASL during colli-

sion and separation. The finite element and two-mass models are compared with each other and with experimental results to assess the influence of the non-Newtonian component of the ASL on vocal fold vibration.

1aSCb5. Rapid three-dimensional magnetic resonance imaging of vocal tract shaping using compressed sensing. Yoon-Chul Kim, Shrikanth Narayanan, and Krishna S. Nayak (Ming Hsieh Dept. of Elec. Eng., Univ. of Southern California, 3740 McClintock Ave., Los Angeles, CA 90089, yoonckim@usc.edu)

Conventional 3D magnetic resonance imaging (MRI) of vocal tract during sustained sound production takes prohibitively long and requires repetitions of the same articulation with pauses of the scan. This can produce image misregistration artifacts possibly due to different positioning of the jaw, tongue, and lips at each repetition. Within a scan time limit of 6–10 seconds, a duration practical for sustained sound production, it is desirable to accelerate high-resolution 3D imaging by undersampling 3D *k*-space. In this study, compressed sensing theory was adopted to achieve the acceleration by incoherent undersampling of 3D *k*-space and minimum L1-norm reconstruction. The increase in transform sparsity was exploited by constraining image phase in which rapid phase variation is expected in air-tissue boundaries due to large magnetic susceptibility difference. Vocal tract volume of interest was imaged during sustained sound production of English consonants /s/, /sh/, /r/, and /l/. Vocal tract area functions were measured using OsiriX software. The visualization of 3D tongue shape was demonstrated via image segmentation and volume rendering process. MR acquisition for each consonant took 6 sec without any repetitions of the articulation. The reconstruction of a whole 3D vocal tract was achievable with a 1.33 $\times 1.33 \times 1.33$ mm^3 isotropic resolution. [Work supported by NIH.]

1aSCb6. An analysis-by-synthesis approach to modeling real-time MRI articulatory data using the task dynamic application framework. Erik Bresch (Dept. of Elec. Eng., Univ. of So. California, Los Angeles, CA 90089, bresch@usc.edu), Louis Goldstein, and Shrikanth Narayanan (Univ. of So. California, Los Angeles, CA 90089)

We report on a method of modeling real-time MRI articulatory speech data using the Haskins task dynamic application (TaDA) framework. TaDA models speech using a set of discrete dynamical regimes that control the formation of vocal tract constrictions (gestures). An utterance can be specified by a gestural score: the pattern of activation of these regimes in time. Individual model articulator degrees of freedom are automatically coordinated according the concurrent demands of the unfolding constrictions. Our modeling procedure consists of two stages: (1) After determining the outline of the midsagittal upper airway, time series of constriction measurements are derived which allow the estimation of the subject-specific parameters relating the articulator and constriction domains. (2) Gradient descent is utilized to adjust the activation intervals of the gestural score generated by TaDA for that utterance so that the synthesized vocal tract constriction evolution matches the observed MRI time series. Additionally, information related to glottal control, which cannot be derived from the MR images, is extracted from the simultaneously recorded audio, and it is used to optimize the timing of glottal abduction gestures in the score. Adequacy of fit is evaluated at articulatory, constriction, and perceptual levels. [Work supported by NIH.]

1aSCb7. Vowel recognition from articulatory position time-series data. Jun Wang (Dept. of Comput. Sci. & Engr., Univ. of Nebraska-Lincoln, Lincoln, NE 68508, junwang@cse.unl.edu), Ashok Samal , Jordan Green , and Tom Carrell (Univ. of Nebraska-Lincoln, Lincoln, NE 68583)

The goal of this project is to determine the accuracy and processing speed of different approaches for mapping time-varying articulatory positional data to vowels. Three widely used classifiers were compared on two datasets: one single speaker and one multiple speaker. The single-speaker dataset was acquired using the Articulograph AG500. The multiple-speaker dataset was obtained from seven speakers in the Xray Microbeam Speech Production Database (Westbury, 1994). The recognition rate for single speaker dataset (eight English vowels) ranged from 94.25% to 98.1%, and from 62.38% to 99.35% for the multiple-speaker dataset. For the single-speaker dataset, recognition accuracy was comparable across classifiers. For the multiple-speaker dataset, recognition accuracy was better for the Support Vector Machine and C4.5 than for the neural networks. The decision tree generated by C4.5 was consistent with articulatory features commonly used to descriptively distinguish vowels. Moreover, the Support Vector Machine and C4.5 performed much faster than did the neural network. The high recognition rates observed suggest that static vowels can be accurately recognized from articulatory position time-series data. The findings are intended to improve the accuracy and response time of a real-time articulatory-movement based speech synthesizer.

1aSCb8. Recovering area functions from acoustics and midsagittal profiles. Richard S. McGowan and Rebecca W. McGowan (CReSS LLC, 1 Seaborn Pl. Lexington, MA 02420)

Simultaneously recorded midsagittal pellet positions and acoustic output for several subjects comprise the Wisconsin X-Ray Microbeam Speech Production Database (XRMB-SPD). An articulatory synthesizer, EASY, was used to help recover area function parameters corresponding to the production of vowels by subjects in the XRMB-SPD by means of analysis-by-synthesis. Specifically, for each subject, an initial personalized midsagittal profile was created based on the pellet positions, pellet trace, and pharyngeal wall trace during neutral vowel production. Based on the scales of this initial profile for a subject, subsequent midsagittal profiles were generated from the pellet data during various vowel productions. Each midsagittal profile and the corresponding first three formant frequencies were used to optimize the parameters of an area function in the EASY for closest acoustic match. The optimization was performed with a genetic algorithm. The results are examined in terms of consistency within a subject for different vowels and consistency within vowel for different subjects. Synthesized examples of vowel production will also be presented. [Work partially supported by Grant DC-001247 to CReSS LLC.]

1aSCb9. Korean articulatory speech synthesis using physical vocal tract model. Huynh V. Luong (Univ. of Ulsan, Korea), Jong-Myon Kim (Univ. of Ulsan, Korea), and Cheol Hong Kim (Chonnam Natl. Univ., Korea)

Artificial vocal tract models provide the support of learning a second language and the therapy of speech disorders. Moreover, phonetic education and research can benefit from articulatory speech synthesis. Articulatory speech synthesis models are constructed by the source-filter model of the human vocal tract. In this study, we generated a Korean articulatory speech synthesis model using Artisynth [Fels *et al.*, ISSP, 419–426 (2006)], which is a 3-D biomechanical open-source simulation platform. As the origin of the Korean language, it has 10 basic vowel phonemes and 11 complicated vowels in which some vowels can be rounded and unrounded such as /eu/, /yeo/, /wae/, etc. To synthesize these specific vowels, we created a new physical vocal tract model, which interconnects to form a complete integrated biomechanical system. The created model efficiently supports recording the Korean vowel sounds and linguistic analysis based on the linear prediction model. As a result, parameters of the glottis and controllable vocal tract filter are automatically evaluated. The acoustic quality of the synthesizer for Korean vowels is comparable with that of the existing commercial speech synthesis systems such as concatenation synthesizers [Donovan (1996)] and [Hamza (2000)]. [Work supported by the MKE, Korea, under the ITRC supervised by IITA (IITA-2008-(C1090-0801-0039)).]

1aSCb10. Vocal and postural coupling in singing and speech. Eric Vatikiotis-Bateson, Martin Oberg, Adriano Barbosa (Linguist., Univ. of British Columbia, 2613 West Mall, Vancouver, BC, Canada, V6T 1Z4), Nancy Hermiston, and Richard Kurth (Univ. of British Columbia, Vancouver, BC, Canada, V6T 1Z1)

This study examines behavioral measures during speech and singing to assess the interaction of the postural control system and the production of expressive vocal behavior at different levels of vocal effort. We know that head motion is essential to both expressive speech and to balanced posture, suggesting that the head must be entrained to some extent by both subsystems simultaneously. This study shows that increased vocal effort (defined here as loudness) affects rigid body (6D) head motion and other measures of postural behavior acquired from force plates (6D) and video-

based motion analysis (2D). Specifically, as vocalizations becomes louder, the postural system becomes more simply coupled spatially and temporally to the vocal system.

1aSCb11. Optimization of articulator trajectories in producing learned nonsense words. Mark Tiede (Haskins Lab., 300 George St., New Haven, CT 06511 & Speech Comm. Group, MIT-R.L.E., tiede@haskins.yale.edu), Christine Mooshammer (Haskins Laborities, New Haven, CT 06511), Louis Goldstein (U.S.C., Los Angeles, CA 90089), Stefanie Shattuck-Hufnagel, Joseph Perkell, and Melanie Matthies (MIT-R.L.E., Cambridge, MA 02139)

Previous work assessing motor learning in novel speech utterances has shown that kinematic duration and variability are reduced as a function of practice [Schulz *et al.* (2001)]. In this study EMA was used to observe articulator movements during production of a variety of polysyllabic nonsense words conforming to English phonotactics (e.g., "thraimpoframodis"). Target words were elicited eight times in block-random order, with each block separated by ten minutes of an unrelated task serving as a distractor. Analysis of EMA sensor trajectories between initial and final repetitions shows in general reduction of overall duration, lower STI [Smith *et al.* (1995)], reduced stiffness, and fewer acceleration peaks, with intermediate productions indicating that these trends occur asymptotically. Comparison of consonant closure timings delimited using velocity extrema and aligned through nonlinear time normalization [Lucero *et al.* (1997)] suggests that as fluency increases, the overlap between adjacent consonantal gestures also increases. The patterns of articulatory optimization illustrated by these results will be contrasted with similar analyses of real but difficult-to-pronounce sequences (tongue twisters) collected from the same subjects, with the goal of identifying similarities between pre-fluent and dysfluent utterances. [Research supported by NIH.]

1aSCb12. Measuring articulatory similarity with algorithmically reweighted principal component analysis. Jeff Mielke and Joseph Roy (Arts 401, 70 Laurier Ave. E., Ottawa, ON Canada, K1N6N5, jmielke @uottawa.ca)

Articulatory similarity was assessed for a corpus of 2700 ultrasound images of 75 cross-linguistically frequent speech sounds produced by four subjects in three segmental contexts (a_a, i_i, u_u). Cross-distances were generated for the entire length of the vocal tract using the Palatron algorithm and realistic estimates of the location of the pharyngeal wall and teeth, resulting, after interpolation, in 60 cross-distances per token. A standard principal component analysis of these data is overwhelmed by the coarticulatory effects of the context vowels. Algorithmically reweighted principal component analysis was devised in order to use coarticulatory variation to isolate the most distinctive cross-distances for each target segment. The reweighting algorithm considers the variance across repetitions in each of the 60 cross-distances, as well as variance in the slope of the cross-distance function, in order to identify areas of stability across tokens. For each target sound, cross-distances with the greatest variance are reset to the mean for all target segments, while cross-distances with the least variance maintain their original values. This has the effect of defining each target sound by the cross-distances, which are most stable across contexts.

1aSCb13. The influence of dynamic flow conditions on sound generation in fricative consonants. Gordon Ramsay (Haskins Lab., 300 George St., New Haven, CT 06511)

Studies of speech production have often remarked that frication noise in voiced fricatives is weaker than that produced in homorganic voiceless fricatives. This is traditionally attributed to the reduction in mean flow (and, hence, Reynolds Number) produced by the glottal constriction needed to induce voicing. However, research in fluid mechanics has demonstrated that turbulence may develop quite differently in steady and unsteady background flows. Strong flow gradients may inhibit the transition to turbulence, or even relaminarize an already-developed turbulent flow. This paper describes a series of mechanical modelling experiments designed to investigate the effect of dynamic flow conditions on sound generation in fricatives. A feedback-regulated Kolvenbach flow valve, controlled by computer, is used to create carefully-shaped dynamic modulations of the air flow through a stereolithographic replica of the vocal tract, about a series of static flow conditions. A multitaper spectral analysis technique is then used in conjunction with sweeptone measurements to separate out the contribution of source and filter components to the far-field sound spectra recorded with a microphone, and source properties are compared across flow conditions. The observed variations cannot be explained by differences in mean flow alone, suggesting that dynamic effects may be important.

1aSCb14. Effects of sleep deprivation on nasalization in speech. Xinhui Zhou (Dept. of Elect. and Comp. Eng., Univ. of Maryland, College Park, MD 20742, zxinhui@glue.umd.edu), Suzanne Boyce (Univ. of Cincinnati, Cincinnati, OH 45267), Joel MacAuslan (Speech Technol. and Appl. Res. Corp., Bedford, MA 01730), Walter Carr, Balkin Thomas, Picchioni Dante (Walter Reed Army Inst. of Medicine, Silver Spring, MD 20910), and Carol Espy-Wilson (Univ. of Maryland, College Park, MD, 20742)

Various aspects of speech appear to be subtly affected by sleep deprivation [Boyce *et al.*, "Landmark-based analysis of sleep-deprived speech," J. Acoust. Soc. Am. **123**(5), 3887 (2008)]. An automatic procedure for detection of nasalization in the acoustic spectrum of vowels was applied to a database of speech recordings from 15 subjects in rested and sleep-deprived conditions. Results showed acoustic differences consistent with differences in sleep condition, suggesting that lack of sleep affects the motor organization/execution of speech articulation in general, and for oral/nasal coupling during vowels in particular. Significant differences in male vs female patterns were also observed. Correspondences between these patterns and those reported in the clinical and linguistic literature will be discussed. [Work supported by NIH.]

1aSCb15. Gestural intrusions in Taiwanese and English. Man Gao (Haskins Labs., 300 George St., Suite 900, New Haven, CT 06511), Yueh-chin Chang, Feng-fan Hsieh (Grad. Inst. of Linguist., Natl. Tsing Hua Univ., Taiwan), Hosung Nam, Mark Tiede (Haskins Labs., New Haven, CT 06511), and Louis Goldstein (U.S.C, Los Angeles, CA 90089; & Haskins Labs., New Haven, CT 06511)

Kinematic studies of repetitive speech production tasks involving English sequences such as "cop top" have provided abundant evidence for gestural intrusion errors: errorful constrictions that are co-produced with intended target constrictions (Pouplier and Goldstein, 2005; Goldstein *et al.*, 2007). These intrusions have been analyzed as transitions to more dynamically stable coordination modes. This work investigates Taiwanese speakers' production of similar repetitive sequences such as [kap tap]. If the speech sequences are planned and executed in Taiwanese in the same way as in English, similar patterns of intrusive speech errors should be observed. However, analysis of the kinematic data from three speakers shows that these effects in Taiwanese differ from those in English in a number of ways. A systematic comparison of Taiwanese and English intrusions is presented in terms of frequency of occurrence (much higher in Taiwanese), standard deviation of magnitude (smaller in Taiwanese), and temporal relationships to intended gestures. The results are discussed with respect to the language-specific regularities of gestural coordination and their implications for the mode-locking account of speech errors. [Work supported by NIH-NIDCD DC008780.]

Session 1aUW

Underwater Acoustics: Propagation Modeling

Eric I. Thorsos, Chair
Applied Physics Lab., Univ. of Washington, Seattle, WA 98105-6606

Chair's Introduction—7:55

Contributed Papers

8:00

1aUW1. Transport theory for shallow water propagation with rough boundaries. Eric I. Thorsos, Frank S. Henyey, W. T. Elam, T. Brian Hefner, Stephen A. Reynolds, and Jie Yang (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105)

At frequencies of about 1 kHz and higher, forward scattering from a rough sea surface (and/or a rough bottom) can strongly affect shallow water propagation and reverberation. The need exists for a fast, yet accurate method for modeling such propagation where multiple forward scattering occurs. A transport theory method based on mode coupling will be described that yields the first and second moments of the field. This approach shows promise for accurately treating multiple forward scattering in one-way propagation. The method is presently formulated in two space dimensions, and Monte Carlo rough surface PE simulations are used for assessing the accuracy of transport theory results. [Work supported by ONR Ocean Acoustics.]

8:15

1aUW2. Wave refraction at an interface: Snell's law versus Chapman's law. Oleg A. Godin (CIRES, Univ. of Colorado and NOAA/Earth System Res. Lab., Boulder, CO 80305, Oleg.Godin@noaa.gov)

Wave energy streamlines determine the paths along which energy is transported from a wave source. The energy streamlines provide insights into mechanisms of wave propagation and scattering and are often utilized to visualize wave fields. In contrast to rays, which are essentially an asymptotic, short-wave concept, energy streamlines adequately represent arbitrary wave fields. However, usefulness of energy streamlines in studies of wave fields is limited by the fact that, unlike rays, no general laws governing energy streamline refraction are known. Here a simple refraction law is derived for energy streamlines of acoustic and linearly polarized electromagnetic waves. The refraction law is an extension of a result recently established by D. M. F. Chapman for sound refraction at plane interfaces of homogeneous media. While Snell's law of ray refraction at an interface is formulated in terms of the ratio of sound speeds in media in contact in the case of acoustic waves, Chapman's law implies that refraction of acoustic energy streamlines is controlled by the ratio of mass densities. Similarities and differences between Snell's and Chapman's refraction laws are discussed. It is shown that analysis of energy streamlines provides a new, helpful perspective on wave transmission through an interface.

8:30

1aUW3. Parabolic equation modeling of anisotropic waves in heterogeneous media. Adam M. Metzler, William L. Siegmann (Rensselaer Polytech. Inst., Troy, NY 12180), Ralph N. Baer, and Michael D. Collins (Naval Res. Lab., Washington, DC 20375)

Wave propagation in heterogeneous media can be modeled efficiently with the parabolic equation method, which has been extended to problems with homogeneous anisotropic layers [A. J. Fredricks *et al.*, Wave Motion **31**, 139–146 (2000)]. This approach is currently being extended to the heterogeneous case, including piecewise continuous vertical dependence and horizontal dependence that is relatively gradual but which may be large over sufficient distances. Vertical dependence is included by applying appropriate heterogeneous depth operators in the equations of motion. Horizontal dependence is included by extending the single-scattering solution [E. T. Kusel *et al.*, J. Acoust. Soc. Am. **121**, 808–813 (2007)] to the anisotropic case. [Work supported by the Office of Naval Research.]

8:45

1aUW4. A two-dimensional coupled mode model for acoustic propagation in a range-dependent waveguide. Wenyu Luo and Henrik Schmidt (Dept. of Mech. Eng., MIT, 77 Massachusetts Ave., Cambridge, MA 02139, lwy@mit.edu)

A two-dimensional, two-way coupled mode model is presented in this work. By combining a forward marching and a backward marching, this model is valid for problems where backscattering is significant. This two-way solution provides great accuracy, and it also features in efficiency, which is achieved by applying the single-scatter approximation. In addition, this model is stable with appropriately normalized range solutions in the formulation. This model is capable of handling two-dimensional problems with either a point source in cylindrical geometry or a line source in plane geometry. The computer code CSNAP is adopted to provide the fundamental model solutions and their associate coupling matrices in this model. This two-dimensional model can be easily extended to a three-dimensional model with the feasibility to solve large-scale realistic three-dimensional problems. Numerical examples are presented to demonstrate the accuracy, efficiency, and stability of this two-dimensional, two-way coupled mode model. [Work supported by the Office of Naval Research.]

9:00

1aUW5. Shallow water tomography: Ray theory vs travel-time sensitivity kernels. I. Iturbe (GIPSA-lab, 961 Rue de la Houille Blanche, BP 46, 38402 Saint Martin d'Heres, France, ion.iturbe@gipsa-lab.inpg.fr), P. Roux (Univ. Joseph Fourier, 38042 Grenoble, France), B. Nicolas (GIPSA-lab, 38402 Saint Martin d'Heres, France), J. Virieux (Univ. Joseph Fourier, 38042 Grenoble, France), and J. Mars (GIPSA-lab, 38402 Saint Martin d'Heres, France)

In underwater acoustics, ray tomography is the classical method used to estimate velocity variations, but recently. Travel-Time Sensitivity Kernels (TSK) approaches have been developed. In this paper, we deal with TSK for two source-receive arrays in an acoustic waveguide for shallow water tomography. As a first step, we show that separation of the different raypaths is improved by using a recently proposed new array processing [time-delay double beamforming (DBF) algorithm]. DBF consists of changing the 3-D data space from source depth, receiver depth and time into a new 3-D space related to ray propagation expressed by the beamformed variables, source angle, receive angle and time. As a consequence, each eigenray of the multipath propagation for a source-receiver couple can be identified and separated through DBF. In this context of DBF, the TSK is no longer point-to-point as usual, but relies on all source-receiver time series. Kernels are computed using the fact that the processed signal is a linear combination of time-delayed signals between all sources and receivers. Results in simulated data and in real datasets recorded in an ultrasonic tank prove that combination of TSK and DBF increases the resolution and robustness performance of shallow-water acoustic tomography.

9:15

1aUW6. Computationally efficient parabolic equation solutions to seismoacoustic problems. Jon M. Collis (Math and Comput. Sci., Colorado Sch. of Mines, 1500 Illinois St., Golden, CO 80401, jcollis@mines.edu)

Improving the capabilities and accuracy of the elastic parabolic equation method has been an active area of research. Variable rotation [D. A. Outing *et al.*, J. Acoust. Soc. Am. **120**, 3534–3538 (2006)] is an effective solution for treating variable bathymetry and has recently been extended to handle variable sediment thicknesses [J. M. Collis *et al.*, J. Acoust. Soc. Am. **123**, 51–55 (2008)] and beach interactions. To accurately resolve the field in elastic sediments, much finer grid spacing may be needed than is necessary to propagate the acoustic field in the water. In particular, as the shear wave speed in the elastic medium approaches zero, the governing system becomes singular, and very fine grid spacing is necessary to capture subtle effects of the field. In this presentation we show how elastic parabolic equation solutions can be computed with greater efficiency, through the use of multiple resolution grids. Interface conditions and derivative approximations on a nonuniform grid will be described. Finally, applications of multiple grid solutions will be discussed for rough surface scattering and treating near-singular low-shear speed sediments.

9:30

1aUW7. Acoustic wave scattering in turbulence and in internal waves. Tokuo Yamamoto, Clair Paris (AMP, RSMAS, Univ. of Miami, 4600 Rickenbacker Cswy, Miami, FL 33149, tyamamoto@rsmas.miami.edu), and Mohsen Badiey (Univ. of Delaware, Newark, DE 19716)

Analytical solutions of acoustic wave scattering have been obtained for turbulent currents and for shallow water internal waves. The scattering characteristics of the two are quit different. The most striking difference is that the maximum scattering is in only one direction by turbulence. Whereas there are two directions of maximum scattering in internal waves. In turbulence, acoustic waves scatter quite strongly while internal waves acoustic waves scatter very weakly. Therefore, turbulence changes acoustic wave (coherent) energy rapidly into (incoherent) scattered acoustic energy. The scattering of acoustic energy by internal waves is weak, on the order of 0.001 of that of turbulence. The theoretical results are verified with transmission loss measured in Kanmon Strait and at New Jersey Shelf with good agreements. [Work supported by ONR Code 322 OA.]

9:45

1aUW8. Resonant forward scattering of sound in the deep ocean by internal tides. Irina I. Rypina (Dept. of Physical Oceanogr., Woods Hole Oceanograph. Inst., 266 Woods Hole Rd., Woods Hole, MA 02543), Ilya A. Udovydchenkov, Timothy F. Duda (Woods Hole Oceanograph. Inst., Woods Hole, MA 02543), and Michael G. Brown (Univ. of Miami, Miami, FL 33149)

The importance of resonant forward scattering of sound by highly directional large amplitude nonlinear internal waves in shallow water environments has been well established. In contrast, most deep ocean environments are characterized by the presence of a sound speed perturbation (due mostly to internal waves) whose horizontal wave number spectrum is broad and approximately horizontally isotropic; under such conditions, the importance of resonant forward scattering is less clear. Exceptional deep ocean conditions are found in the vicinity of submarine ridges, which serve to generate internal tides that are both highly directional and have a narrow horizontal wave number spectrum. In this talk, ray, mode, and PE simulations are presented to explore the importance of resonant forward scattering by such structures under fairly realistic oceanographic conditions. [Work supported by ONR.]

10:00—10:15 Break

10:15

1aUW9. The second order resolution operator of a nonlinear ocean acoustics inverse problem. Robert I. Odom and Andrew A. Ganse (Appl. Phys. Lab, Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, odom@apl.washington.edu)

The resolution operator for a linear inverse problem indicates how much smearing exists in the map between the true model and the estimated model. The trace of the resolution operator provides an estimate of the number of model parameters model that are resolved. In a series representation of the resolution operator for a nonlinear problem, the higher-order terms indicates how much spurious nonlinear leakage there is from the true model to the estimated model. In previous work, the solution of a simple nonlinear ocean acoustic inverse problem as a perturbation series in the horizontal wavenumber was constructed, and the linear data kernels were presented. This linear problem permitted the quantification of the magnitude of the perturbation and indicated when nonlinear effects must be taken into consideration. In this extension of previous work, the second order resolution kernel is constructed, which describes how effectively nonlinear effects can be removed from the reconstructed model. [Work supported by ONR.]

10:30

1aUW10. A finite element model for acoustic propagation in shallow water waveguides. Marcia Isakson (Appl. Res. Lab., The Univ. of Texas at Austin, Austin, TX 78713, misakson@arlut.utexas.edu)

A finite element model is developed using a commercial code to calculate the backscattering of acoustic waves in a two-dimensional shallow water waveguide with rough interfaces. Finite elements approach an exact solution to the Helmholtz equation as the discretization density increases. A time series for the backscattering is computed from time harmonic computations using Fourier synthesis. A parametric study of the roughness parameters for the sediment/water interface and the air/water interface will be presented in order to assess the range of validity of approximate methods. [Work sponsored by Internal Research and Development, Applied Research Laboratories.]

10:45

1aUW11. A comparison of acoustic uncertainty approximations in a Pekeris waveguide. David R. Dowling and Kevin R. James (Dept. of Mech. Eng., Univ. of Michigan, 2212 G. G. Brown, Ann Arbor, MI 48109, drd@engin.umich.edu)

Uncertainty in environmental parameters is often the dominant source of error in underwater acoustic field predictions. This presentation provides comparisons of three techniques for assessing uncertainty in a predicted acoustic field caused by uncertainty in environmental parameters: field shifting, polynomial chaos expansion, and coarse uniformly sampled direct simulations. The uncertainty assessments are performed in a 100-m deep Pekeris waveguide with an uncertain water-column sound speed for frequencies of 100 Hz to 1 kHz at ranges of 1 to 10 km with a variety of common bottom types. The accuracy and computational efficiency of each approximation are quantified in terms of an absolute-difference error norm for the probability density function (PDF) of acoustic field amplitude and the number of acoustic field calculations necessary to predict this PDF, respectively. In all cases, the true field-amplitude PDF is determined from numerically converged direct numerical simulations. The strengths and limitations of each technique are highlighted and the applicability of these results to other sound channels and uncertain parameters is discussed. In particular, polynomial chaos expansion produces the highest accuracy but does so at the highest computational cost, while field shifting produces acceptable accuracy for the least computational cost. [Work sponsored by the Office of Naval Research.]

11:00

1aUW12. Modeling gradients in ocean acoustics. Robert I. Odom (Appl. Phys. Lab, Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, odom@apl.washington.edu) and Karsten James (California Polytechnic State Univ., San Luis Obispo, CA 93407)

Modeling acoustic propagation in the ocean requires a representation of the ocean sound speed. For a 1-D, range independent ocean, probably the most common representation comprises constant sound speed layers and/or gradient layers. Gradients of the form $1/c^2(z)$, admitting an exact solution for the acoustic pressure field in terms of Airy functions, are commonly employed. A disadvantage of using the Airy functions is that they are spe-

cific to that profile. For more general profiles, the best representation may be a stack of constant sound speed layers for which the pressure field is expressible in terms of simple exponentials. As an example, computing the backscattering strength from a stack of layers containing volume heterogeneities requires evaluation of an integral proportional to the 4th power of the pressure field. Integrating the simple exponential solutions for constant sound speed layers is trivial. Integrating a product of four Airy functions, either analytically or numerically, is not. If the medium is represented only by constant sound sound speed layers, what is the best way to choose the layers, and how many layers are needed? These questions are investigated by employing an algorithm originally due to Fuchs (1968). [Work partially supported by ONR.]

MONDAY AFTERNOON, 18 MAY 2009 GRAND BALLROOM II, 1:00 TO 4:50 P.M.

Session 1pAA

Architectural Acoustics and Noise: Acoustics of Green Buildings: A 360 Degree Perspective

David M. Sykes, Cochair
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Chair's Introduction—1:00

Invited Papers

1:05

1pAA1. National Academy of Engineering "Technology for a quieter America" project—impact on "green acoustics"? Patricia Davies (Sch. of Mech. Eng., Ray W. Herrick Labs., Purdue Univ., 140 Martin Jischke Dr., West Lafayette, IN 47907-2031, daviesp@ecn.purdue.edu) George C. Maling, Jr. (Inst. of Noise Control Eng., Harpswell, ME 04079) and Gregory C. Tocci (Cavanaugh Tocci Assoc., Inc., Sudbury, MA 01776)

In 2006, the National Academy of Engineering (NAE) initiated an investigation of the potential benefits of a focused effort to control noise in the workplace, communities, and homes that may be of interest to professionals engaged in "green" acoustics. It covered three initiatives: (1) applications of current technology; (2) research and development initiatives for noise control technology; and (3) intragovernmental and public relations programs. Several workshops were held in the 2005–2008 period with representatives from industry, academia, professional societies, private consultants, and government agencies who examined existing and potential solutions along with policies that would help to develop and deploy solutions. Members of the committee were George C. Maling, Jr., Robert J. Bernhard, Robert Bruce, Elizabeth A. Cooper, Patricia Davies, Carl Hanson, Robert Hellweg, Gerald Lauchle, Richard Lyon, Gregory Tocci, Ian Waitz, and others. In mid-2009, a consensus report will be issued that: (a) summarizes the state of current noise control engineering practice; (b) recommends how existing knowledge can address current challenges; (c) presents a research and education agenda that promotes the generation of new knowledge that can provide the greatest benefit to society; and (d) recommends policies that government agencies can develop to further improve the American soundscape.

1:20

1pAA2. LEED (Leadership in Energy and Environmental Design) and indoor environmental quality—What about office acoustics? Kevin M. Powell (Office of Appl. Sci. GSA Public Bldg.s Service, 555 Battery St., Rm. 518, San Francisco, CA 94111, kevin.powell@gsa.gov)

Recent research conducted by the US General Services Administration and the Center for the Built Environment disclosed an anomaly about "green building" interiors, that is, an apparently unintended consequence of the current LEED rating system. The anomaly is this: while occupants of offices that have been upgraded to achieve LEED certification express high levels of satisfaction with the "indoor environmental quality" of these spaces on most dimensions, there is one exception where scores actually decrease. This exception is the only dimension where occupant DISsatisfaction grows—it is acoustics. What is it about LEED-rated offices that cause this anomaly to occur? Why are occupants dissatisfied with the "acoustical comfort" of the new LEED-certified offices? Is it because there are more reflective, hard surfaces causing noise to reverberate? Or because the HVAC system has become quieter? Or because partition heights have been lowered to allow better airflow and more natural light, resulting in less privacy? Since increased occupant dissatisfaction may result in reduced productivity or increased absenteeism, the GSA's office of research has been examining this issue and will report on its findings and conclusions.