Session 1aAA

Architectural Acoustics and Noise: Soundscape Methods and Case Studies in Architectural Projects: Applications of Soundscape Techniques to the Realization of Perceived High Quality Sonic Environments in Architectural Projects

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

Juergen Bauer, Cochair
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Chair's Introduction—8:05

Invited Papers

8:10

1aAA1. Soundscape and architecture—What is your vision? Bennett M. Brooks (Brooks Acoustics Corporation, 30 Lafayette Square-Ste. 103, Vernon, CT 06066, bbrooks@brooks-acoustics.com) and Dennis Paoletti (Paoletti Consulting, San Mateo, CA)

The soundscape technique combines physical acoustical parameters with user perceptions of the sonic environment in a context of meaning. This method can be a powerful analysis and design tool for a wide range of projects, including building interior and exterior spaces, site planning, urban and transportation planning, environmental noise control, public parks, etc. The soundscape method can also be a useful marketing and project management tool, with the goal to address acoustical concerns as early as possible in the architectural design process, even in the inspiration phase. Sonic perceptions of the built environment are often a vital part of the vision for a project, and must be expressed at the outset to be fully incorporated in the design. Innovative project delivery methods and contract structures such as Integrated Project Delivery (IPD), unlike design-bid-build, assign shared risk and reward among the design, construction, and management teams. Design inputs are solicited from all stakeholders and design team members very early, before programming. This and similar delivery methods offer great opportunities for practitioners, through soundscaping, to include acoustics in the initial project discussions, and to advance the implementation of quality sonic environments. Project case study examples are discussed.

8:30

1aAA2. Architectural acoustics and sense of place. Michael A. Stocker (Ocean Conservation Res., P.O. Box 559, Lagunitas, CA 94938, mstocker@OCR.org)

A large part of the work in architectural acoustics is focused on noise control, tailoring reverberation time, improving speech intelligibility, and reduction of specular reflections. But there are other factors that play into the suitability of an acoustic environment for a specific purpose that are not easily expressed on numerical indices. In this presentation, the author reviews his own work in the US National Holocaust Museum "Daniel's House" exhibit, the Museo Papalote del Niño Rainforest Tree Exhibit in Mexico City, and various other settings in public exhibition spaces. The presentation will also review some historic successes and failures in public enclosures and soundscapes that hinge on how the user/visitor experiences their relationship with the acoustical surroundings.

8:50

1aAA3. Traffic design for soundscape improvements. Klaus Genuit and André Fiebig (HEAD Acoust. GmbH, Ebertstr. 30a, Herzogenrath 52134, Germany, klaus.genuit@head-acoustics.de)

Different kinds of traffic contribute to urban acoustic environments and therefore have a major impact on urban soundscapes. The noise of traffic depends on several aspects such as traffic management, traffic routing, traffic composition, and infrastructure. In the past, any optimization of these aspects targets only on sound pressure level reduction, neglecting perceptional relevant phenomena. It is well known that the human hearing does not work like a simple sound level meter. Besides loudness, humans perceive psychoacoustic properties of noise and notice certain sound events and sources. Thus, any improvement of traffic noise must be guided by knowledge from psychoacoustics and cognition. To sustainably improve the appraisal of a soundscape, traffic must be deliberately designed. In different research projects, the psychoacoustic potential of traffic design was systematically investigated. For example, the perceptual difference between roundabouts and intersections with and without traffic lights was investigated, psychoacoustic requirements for the layout of road markings were studied, and the required penetration level of electric cars for a substantial noise reduction beyond sound pressure level considerations was an object of investigation. Options and possibilities of traffic design from a psychoacoustic perspective and their implications for urban planning will be presented.

9:10

1aAA4. Perceived space, an essential soundscape variable. Alexander U. Case (Sound Recording Technol., Univ. of Massachusetts Lowell, 35 Wilder St., Ste. 3, Lowell, MA 01854, alex@fermata.biz)

The perceived space within a soundscape is free from the constraints of place and time. The space defined by the architecture need not define the perceived space within a soundscape. Space is a variable, determined by the combined influences of the architecture, sounds system, source signals, and signal processing applied in the creation of the soundscape. The perceived space is also free to change over time through dynamic signal processing and any other variable acoustics.

9:30

1aAA5. Production techniques for perceptually realistic soundscape auralization. Matthew Azevedo (Acentech Inc., 33 Moulton St., Cambridge, MA 02138, mazevedo@acentech.com)

It is now possible to build complex, parametrically accurate auralizations with many sound sources. However, parametric accuracy does not necessarily translate to perceptual accuracy. In order to create soundscapes that listeners will perceive as "real," audio production techniques are required, which generally lie outside of the expertise of an acoustician. This paper provides an overview of audio recording, mixing, and postproduction techniques that are commonly employed by music producers, sound designers, and audio engineers in creating soundtracks for television, film, and video games, which provide listeners with perceptually realistic sound experiences. These techniques are presented in the context of auralization projects in which they were employed to successfully bridge the gap between parametric and experiential accuracy, with an emphasis on methods which satisfy both. Also discussed are modeling and convolution techniques which lay the groundwork on which these recording and mixing techniques can be deployed successfully.

9:50–10:05 Break

10:05

1aAA6. Clues from the brief, from the site and from the users Architectural design and soundscape. Juergen Bauer (Dept. of Architecture, Waterford Inst. of Technol., Granary, Hanover St., Waterford, Co Waterford 00000, Ireland, jbauer@wit.ie)

This paper investigates how the Soundscape approach and the architectural design process can inform each other. Designing is an intuitive process and is therefore not bound by strict rules. However, most architects will agree that a successful design concept is guided by three factors: First, a design proposal needs to meet its purpose, i.e., to respond to the demands of a design brief. Second, a design proposal should contribute to its location and the surrounding neighborhood, whether it is blending into this context or emerging from it. Lastly, the actual idea for a design proposal is informed by the clues from the brief and the clues from the site. Supported by case studies, it is argued that the Soundscape approach can greatly contribute to the design process and its focus on the brief, the location, and the conceptual idea. It is further discussed how the public debate on architecture in the context of town planning and design competitions can be beneficial to the Soundscape approach and its interest in the contribution of the users and the local stakeholders.

10:25

1aAA7. The challenge of interdisciplinarity in soundscape. Brigitte Schulte-Fortkamp (Inst. of Fluid Mech. and Eng. Acoust., TU Berlin, Einsteinufer 25, Berlin 10587, Germany, b.schulte-fortkamp@tu-berlin.de)

Interdisciplinarity is one of the most stressed player in Soundscape. Over time when starting to collaborate in and with different disciplines, it became obvious that it is necessary to understand the needs in soundscapes as well as the use of soundscape techniques in the respective disciplines. One of the most important outcomes in the COST project on Soundscape TD 0804 with respect to... that a common language is needed to guarantee the collaboration. Especially in the field of Architecture, there is the need to understand why Soundscape and landscape have some similarities but also that landscape and soundscape are different in their focus. The paper will discuss understandings and misunderstandings in the respective fields and provide an orientation guideline.

Contributed Papers

10:45

1aAA8. Acoustic measurements based on a soundscape analysis in an open-plan classroom in a primary school. Sang Bong Shin (Architecture, Univ. of Florida, 2330 SW Williston Rd., APT534, Gainesville, FL 32608, archisangbong@gmail.com)

It is critical to fully understand the acoustic environment in open plan classrooms because there is a current increasing popularity of a new type of classrooms and because it has been reported that the open plan classroom has a serious problem of noise. Since the architectural features of the open plan classrooms are different from those of traditional classrooms, traditional measurement methods are not sufficient to investigate the new type of classrooms. In this study, a new type of open-plan classroom combined with small classrooms is examined with soundscape approaches. Acoustical events occurring in the open-plan classroom in a primary school were analyzed, and the activities that created the specific acoustical events were observed using methods of soundwalks, focus group discussions, and narrative interviews. Also acoustic measurements were conducted with measurement sets from soundscape analysis and from traditional methods in the building. The results of measurements were compared to determine the differences in the effects of measurement methods on the acoustical events in open plan classrooms. The study found that the results of the acoustical measurements based on soundscape analysis are different from those of traditional measurement methods. The differences among the measurement sets demonstrate that it is useful to use soundscape analyze to understand open plan classrooms.

11:00

1aAA9. Soundscape evaluation on Mississippi State University campus. Yalcin Yildirim (Mississippi State Univ., 103 Eudora Wwelty Dr., Avalon Apartments, D/11, Starkville, MS 39759, yy214@msstate.edu)

The term soundscape, used first time at the end of 1970s, refers to the sum of the sounds which can be heard and perceived by people in a specific

environment (Schafer). The concept of soundscape has recently paid attention to planning and design disciplines where the focus point is commonly placed on the visual, rather than the acoustic aspect. The perception of an outdoor environment does not only depend on the physical features of a site, but also relies on the characteristics of the users. Thus, this research will examine how objective measurement of soundscape might be different from subjective perceptions of users in the Mississippi State University Campus as a public open space due to demographic and climatic variations. Stage one, as a pilot or a preliminary study, was a soundscape walk with a small group in four selected sites. Stage two will include more detailed interviews in these sites with a much larger sample size from the general public. At the end of the study, the research findings will help to characterize soundscapes of different types of urban open spaces and to understand how a person perceives and evaluates the sound qualities in these areas.

11:15

1aAA10. "Illuminating" reflection orders in architectural acoustics using SketchUp and light rendering. J. Parkman Carter (Architectural Acoust., Rensselaer Polytechnic Inst., 32204 Waters View Circle, Cohoes, NY 12047, cartej8@rpi.edu)

The conventional architecture workflow tends to—quite literally—"overlook" matters of sound, given that the modeling tools of architectural design are almost exclusively visual in nature. The modeling tools used by architectural acousticians, however, produce visual representations, which are, frankly, less than inspirational for the design process. This project develops a simple scheme to visualize acoustic reflection orders using light rendering in the freely available and widely used Trimble SketchUp 3D modeling software. In addition to allowing architectural designers to visualize acoustic reflections in a familiar modeling environment, this scheme also works easily with highly complex geometry. The technique and examples will be presented.

11:30–12:00 Panel Discussion

MONDAY MORNING, 5 MAY 2014 — BALLROOM B, 8:00 A.M. TO 11:55 A.M.

Session 1aAB

Animal Bioacoustics and Psychological, and Physiological Acoustics: Comparative Perspectives on the Cocktail Party Problem I

Mark Bee, Cochair
Dept. of Ecology and Evolutionary Biology, Univ. of Minnesota, 100 Ecology, 1987 Upper Buford Circle, St. Paul, MN 55108

Micheal L. Dent, Cochair
Psychology, Univ. at Buffalo, SUNY, B76 Park Hall, Buffalo, NY 14260

Invited Papers

8:00

1aAB1. Directional cues for sound source segregation in birds, crocodilians, and lizards. Catherine E. Carr (Biology, Univ. Maryland, Stadium Dr., 20742, College Park, MD 20742-4415, cecarr@umd.edu) and Jakob Christensen-Dalsgaard (Dept. of Biology, Univ. Southern Denmark, Odense, Denmark)

Sound source segregation depends on neural mechanisms that enhance directionality. The main directional cues are interaural time difference (ITD) and interaural level difference (ILD). Birds, crocodilians, and lizards have a brainstem circuit used for detection of ITDs. In birds and crocodilians, this circuit forms a map of ITD by delay lines and coincidence detection. The physical range of ITDs for these maps are small in animals with small heads, which should make detection of ITDs difficult. Both birds and crocodilians have coupled ears, however, which extend the range of ITDs available as well as enhancing ILD. Lizards have even more strongly coupled ears, extending the ITD range by a factor 3, but ITD and ILD covary, and a large part of the ITD is a constant delay created by filtering by interaural cavities, chiefly producing enhanced lateralization. All lizard auditory nerve fibers show strongly directional responses, and effectively every neuron in the lizard auditory pathway is directional, enhancing the already strong lateralization by simple EI-type neural processing, but with no clear maps of auditory space. Thus, the processing of sound direction in the bird, alligator, and lizard CNS is different, but all three groups have mechanisms for enhancing sound source directionality.

8:20

1aAB2. Spatial stream segregation by cats, rats, and humans. John C. Middlebrooks, Peter Bremen (Otolaryngol., Univ. of California at Irvine, Rm. 116 Medical Sci. E, Irvine, CA 92697-5310, j.midd@uci.edu), Lauren K. Javier, and Justin D. Yao (Neurobiology and Behavior, Univ. of California at Irvine, Irvine, CA)

Spatial hearing aids a listener in disentangling multiple competing sound sequences. We find that separation of around 10° between target and masker sound sources permits humans and cats to hear interleaved sound sequences as segregated streams, thus enabling a "rhythmic masking release" task requiring recognition of target rhythms. In cats and rats, neurons in primary auditory cortex (A1)

exhibit spatial stream segregation in that they synchronize selectively to one of two interleaved sequences of noise burst originating from spatially separated sources. Cortical spatial selectivity is markedly sharper under competing-sound conditions compared to that observed with single sound sources. Cortical responses are predicted well by a model that incorporates moderate spatial selectivity inherited from the brainstem sharpened by forward suppression at the level of thalamocortical synapses. Consistent with that model, spatial stream segregation in rats is stronger in cortical area A1 than in the ventral division of the medial geniculate body, its principal source of thalamic input. In cats, psychophysical performance was better for high-frequency sounds, and cortical stream segregation was stronger for neurons having high characteristic frequencies. In contrast, human psychophysics was better for low-frequency sounds, suggesting that the larger heads of humans provide them with greater interaural time differences.

8:40

1aAB3. Solutions to cocktail-party-like problems in acoustic insects. Heiner Römer (Zoology, Karl-Franzens-Univ., Universitätsplatz 2, Graz 8010, Austria, heinrich.roemer@uni-graz.at)

Insects often communicate by sound in mixed species choruses; like humans and many vertebrates, in crowded social environments, they thus have to solve cocktail-party-like problems in order to ensure successful communication. This is a particular problem in species-rich environments like tropical rainforests with background noise levels of up to 60 dB SPL. I describe three "bottom-up" mechanisms in cricket receivers, which contribute to an excellent neuronal representation of conspecific signals under such conditions. First, more sharply tuned frequency selectivity of the receiver reduces the amount of masking energy around the species-specific calling song frequency, resulting in a signal-to-noise ratio (SNR) of −-8 dB, when masker and signal were broadcast from the same side. Second, spatial release from masking improved the SNR by further 6 to 9 dB. Neurophysiological experiments carried out in the nocturnal rainforest yielded a further improvement of SNRs by 8 dB compared to the laboratory. Finally, a neuronal gain control mechanism enhances the contrast between the responses to signals and the masker, by inhibition of neuronal activity in inter-stimulus intervals. The results indicate that without knowledge of receiver properties and the spatial release mechanisms the detrimental effect of noise may be strongly overestimated.

9:00

1aAB4. Cross-modal integration and non-linear relationships: What can frogs tell us about solving cocktail party problems? Ryan C. Taylor (Biology, Salisbury Univ., 1101 Camden Ave., Salisbury, MD 21801, rctaylor@salisbury.edu)

Courtship in most anuran amphibians occurs in noisy environments, analogous to human communication at cocktail parties. Female frogs express strong mating preferences for particular properties of male vocalizations, but how they identify individual callers within the noisy chorus environment remains unclear. One possible mechanism is cross-modal integration, whereby females attend to both acoustic and visual cues (male vocal sac inflation). In choice experiments, we used a robotic frog with an inflating vocal sac, combined with acoustic playbacks, to test the role of cross-modal integration in female túngara frogs. In nature, male túngara frogs produce a two-note courtship call and the vocal sac inflates synchronously during production of both notes. We tested female mating preferences when we artificially varied the temporal synchrony of the vocal sac inflation relative to the two call notes. Some combinations elicited a strong preference from females, some combinations generated a strong aversive response, and other combinations were neutral. These data show that females conduct cross-modal assessments of male callers. The temporal combinations that elicited positive, negative, or neutral responses were not predictive in a linear fashion, however, suggesting that the integration of visual cues may strongly modulate auditory perception in females.

9:20

1aAB5. Comparative perception of temporally overlapping sounds. Erikson G. Neilans and Micheal L. Dent (Psych., Univ. at Buffalo, SUNY, B76 Park Hall, Buffalo, NY 14260, mdent@buffalo.edu)

Parsing the auditory scene is a problem faced by humans and animals alike. Characteristics such as frequency, intensity, and location all help organisms assign concurrent sounds to specific auditory objects. The timing of sounds is also important for object perception. When sounds totally overlap in time, ascribing separate sounds to individual objects is difficult. However, slight temporal separations make this task slightly easier. In humans, synchronous streams of high and low frequency tones are heard as a single auditory stream. When the tones are slightly offset in time from one another, a second stream emerges. Here, we compared the perception of simultaneous, asynchronous, and partially overlapping streams of tones, human speech sounds, and budgerigar (*Melopsittacus undulatus*) contact calls in budgerigars and humans using operant conditioning methods. Human and bird subjects identified the partially overlapping stimuli differentially. Both species required less temporal separation to identify the sounds as "asynchronous" for the complex stimuli than for the pure tones. Interestingly, the psychometric functions differed between the two species. These results suggest that both humans and nonhumans are capable of using temporal offsets for assigning auditory objects, and that the ability to do this depends on the spectrotemporal characteristics of the sounds.

9:40

1aAB6. Echolocating bats face a cocktail party nightmare when they fly together in cluttered environments. Cynthia F. Moss (Dept. of Psych. and ISR, Univ. of Maryland, Biology-Psych. Bldg. 2123M, College Park, MD 20742, cynthia.moss@gmail.com), Clement Cechetto (AGROSUP, Inst. Nationale Superieur des Sci. Agronomique, Dijon, Ce, DC France), Michaela Warnecke (Psychol. and Brain Sci., Johns Hopkins Univ., Baltimore, MD), Chen Chiu, Wei Xian, and Benjamin Falk (Dept. of Psych. and ISR, Univ. of Maryland, College Park, MD)

Echolocating bats often operate in the presence of conspecifics and in cluttered environments, which can be characterized as a "cocktail party nightmare." Each bat's sonar vocalization can result in an echo cascade from objects distributed in direction and range. Adding to the acoustic clutter are the signals from neighboring bats. Past studies demonstrate that bats adapt their echolocation to avoid signal jamming from conspecifics by adjusting the frequencies of their vocalizations, as well as going silent. When bats fly alone in densely cluttered environments, they adjust the frequencies of call pairs to disambiguate overlapping echo streams. How do echolocating

bats adapt to both conspecific signals and clutter? We sought to answer this question by flying big brown bats in a large room equipped with high-speed video and audio recording equipment. In baseline trials, bats flew alone in an empty room and were later introduced to an artificial forest, first individually and later in pairs. The echolocation behavior and flight paths are analyzed to evaluate the spectro-temporal adjustments of bat calls and silent behavior as animals progressed from open room, to forest, to forest with conspecifics. The results shed light on how echolocating bats adapt to a "cocktail party nightmare."

10:00

1aAB7. Neural representations of the cocktail party in human auditory cortex. Jonathan Z. Simon (Biology, Univ. of Maryland, Dept. of Elec. & Comput. Eng., Univ. of Maryland, College Park, MD 20742, jzsimon@umd.edu)

An auditory scene is perceived in terms of its constituent auditory objects. Here, we investigate how auditory objects are individually represented in human auditory cortex, using magnetoencephalography (MEG) to record the neural responses of listeners. In a series of experiments, subjects selectively listen to one of two competing streams, in a variety of auditory scenes. In the acoustically richest example, subjects selectively listen to one of two competing speakers mixed into a single channel. Individual neural representations of the speech of each speaker are observed, with each being selectively phase locked to the rhythm of the corresponding speech stream, and from which can be exclusively reconstructed the temporal envelope of that speech stream. The neural representation of the attended speech, originating in posterior auditory cortex, dominates the responses. Critically, when the intensities of the attended and background speakers are separately varied over a wide intensity range, the neural representation of the attended speech adapts only to the intensity of that speaker, but not to the intensity of the background speaker. Overall, these results indicate that concurrent auditory objects, even if spectrally overlapping and not resolvable at the auditory periphery, are indeed neurally encoded individually as objects, in auditory cortex.

10:20–10:35 Break

10:35

1aAB8. Temporal and spatial coherence as cues for across-frequency grouping in treefrogs. Mark Bee (Ecology, Evolution and Behavior, Univ. of Minnesota, 100 Ecology, 1987 Upper Buford Circle, St. Paul, MN 55108, mbee@umn.edu)

Humans exploit environmental regularities in sounds to perceptually bind acoustic energy occurring simultaneously at different frequencies. Such abilities influence vowel perception in speech and timbre perception in music. Other animals solve similar binding problems in the recognition of species-specific acoustic signals. Moreover, they commonly do so using auditory systems that differ in notable ways from that of mammals. This study of two treefrog species investigated temporal and spatial coherence as cues that promote grouping of two spectral bands emphasized in their acoustic signals. In two-alternative choice tests, females preferred temporally and spatially coherent calls over alternatives in which the onsets/offsets of the two bands were time-shifted by more than 25 ms or in which the two bands were spatially separated by 7.5° or more. These results, which suggest temporal coherence and spatial coherence promote across-frequency auditory grouping, are notable given differences in how the two spectral bands are processed by the anuran auditory system. Sound energy in the high- and low-frequency bands primarily enters the auditory system via different pathways (tympanum and body wall, respectively) and is encoded primarily by different papillae in the inner ear (basilar papilla and amphibian papilla, respectively).

10:55

1aAB9. Release from auditory masking with complex signals and complex noise in the bottlenose dolphin (*Tursiops truncatus*). Brian K. Branstetter (National Marine Mammal Foundation, 2240 Shelter Island Dr. #200, San Diego, CA 92106, brian.branstetter@nmmpfoundation.org), Jennifer S. Trickey, Kimberly L. Bakhtiari, Amy Black (G2 Software Systems Inc., San Diego, CA), and James J. Finneran (US Navy Marine Mammal Program, San Diego, CA)

Dolphins are social animals that rely heavily on passive and active acoustics for communication, navigation, foraging, and detecting predators. Auditory masking, from both natural and anthropogenic noise sources, may adversely affect these fitness-related capabilities. The dolphin's ability to detect a variety of complex signals (both dolphin phonations and tonal signals) masked by Gaussian, comodulated, snapping shrimp, and ice squeaks noise was tested. Detection thresholds were measured using a go/no-go adaptive staircase procedure. Masking patterns were similar for all signals (whistles, burst-pulse, and pure tones) except for click signals. Masking from ice squeaks resulted in the largest masked thresholds, while snapping shrimp and comodulated noise resulted in a release from masking relative to thresholds from Gaussian noise. Click signals were most difficult to detect when masked by snapping shrimp. Recognition thresholds were estimated for whistle-like signals using a cross-modal, matching-to-sample procedure. Recognition thresholds were on average 4 dB greater than detection thresholds for all noise types. The auditory mechanisms governing the results are discussed. [Work supported by the ONR.]

11:15

1aAB10. Neural correlates of hearing in noise in macaque auditory cortex. Yale Cohen and Sharath Bennur (Univ. Pennsylvania, 3400 Spruce St., 5 Ravdin, Philadelphia, PA 19104, ycohen@mail.med.upenn.edu)

The perception of sound in a noisy environment is a critical function of the auditory system. Here, we describe results from our study into the link between neural activity in the auditory cortex and the hearing-in-noise tasks described above. We recorded neural activity from single neurons in the core auditory cortex (i.e., A1) while monkeys were participating in these tasks. Neural recordings were conducted with tetrodes, and the frequency of the target matched the best frequency of the recorded auditory neuron. We found that the relative intensity of the target tone in the presence of the noise masker significantly modulated the response of A1 neurons. In contrast, the presentation of the target sound alone did not elicit a significant response from A1 neurons. This suggests a task-relevant contextual modulation of A1 responses during hearing in noise. Additionally we found no correlation between the monkey's behavioral choices—as assessed by their responses on choice trials—and A1 activity. Our results suggest that the encoding of a sound of interest in the presence of a noise masker is an active process, providing new insights into the neural basis for hearing in noise in the auditory system.

11:35

1aAB11. Communicating in a cacophony: Possible solutions to the cocktail party problem in treefrog choruses. Joshua J. Schwartz (Biology and Health Sci., Pace Univ., 861 Bedford Rd., Pleasantville, NY 10570, jschwartz2@pace.edu)

Male treefrogs advertise for mates in dense assemblages characterized by high levels of noise and acoustic clutter. Non-mutually exclusive approaches to ameliorating the "cocktail party" problem in frog choruses could involve signal production or perception. Male neotropical *Dendropsophus microcephalus* employ multi-note calls and can rapidly alter inter-note timing to reduce call overlap. Adjustments are made selectively such that interference is most effectively reduced among closest neighbors. There is evidence that intensity and perhaps spatial cues contribute to this selectivity. Male gray treefrogs, *Hyla versicolor*, do not seem to exhibit selective attention in a way that reduces call interference among nearest neighbors, and changes made in call duration and rate that occur with increasing noise levels do not aid in signal detection by females. Moreover, auditory induction, by which the auditory system might perceptually restore masked or missing elements of pulsatile calls, does not seem to occur. Although, under some circumstances, differences in call frequency may help females distinguish among neighboring males, naturalistic spectral differences do not seem to help females perceptually separate the overlapping calls of such males. There is evidence, however, that spatial separation of males can contribute to signal segregation by listening females during acoustic interference.

MONDAY MORNING, 5 MAY 2014 552 A, 7:55 A.M. TO 11:50 A.M.

Session 1aAO

Acoustical Oceanography and Signal Processing in Acoustics: Using Acoustics to Study Fish Distribution and Behavior I

Kelly J. Benoit-Bird, Chair
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Timothy K. Stanton, Chair
Woods Hole Oceanogr. Inst., MS #11, Woods Hole, MA 02543-1053

Chair's Introduction—7:55

Invited Papers

8:00

1aAO1. The benefits and challenges of passive acoustic monitoring of fish. Carrie C. Wall (Cooperative Inst. for Res. in Environ. Sci., Univ. of Colorado at Boulder, 216 UCB, Boulder, CO 80309, carrie.bell@colorado.edu)

While is it widely known that numerous families of fish produce sound for communication, discerning when, where, and who is more difficult. Recent developments in passive acoustic technologies have facilitated marine bioacoustic studies to effectively monitor soniferous fishes. Because acoustic data can be collected over a wide range of habitats and depth for long periods of time, passive acoustic monitoring can map and monitor marine species to efficiently provide year-round information on distribution. This presentation reviews data recorded using moored passive acoustic arrays and hydrophone-integrated gliders. Low frequency (50–6000 Hz) sounds recorded by these methods provide a better understanding of the diurnal and spatial distribution of known fish calls (e.g., red grouper). However, this is seemingly overwhelmed by the vast number of sounds produced by unknown species. Instrument and anthropogenic noise, managing the large of amounts of data collected, and identifying the source of previously undocumented sounds, are just some of the challenges passive acoustic monitoring presents. The connection between sound and important behavior, including courtship and spawning, the application for fisheries management, and the potential impacts of aquatic noise on critical behaviors that affect populations exemplifies the need to overcome these issues.

8:20

1aAO2. Broadband classification and statistics of long-range, mid-frequency sonar measurements of aggregations of fish. Benjamin Jones (Oceanogr., Naval Postgrad. School, Monterey, CA), Timothy K. Stanton (Dept. of Appl. Ocean Phys. and Eng., Woods Hole Oceanogr. Inst., MS #11, Woods Hole, MA 02543, tstanton@whoi.edu), John A. Colosi (Oceanogr., Naval Postgrad. School, Monterey, CA), Roger C. Gauss (Acoust. Div., Naval Res. Lab., Washington, DC), Joseph M. Fialkowski (Acoust. Div., Naval Res. Lab., Washington, California), and J. M. Jech (NOAA Northeast Fisheries Sci. Ctr., Woods Hole, MA)

Scattering from fish can constitute a significant portion of the high-amplitude echoes in the case of a horizontal-looking sonar system operating at mid-frequencies (1–10 kHz). In littoral environments, reverberation from fish with resonant gas-filled swimbladders can dominate bottom and surface reverberation and add spatio-temporal variability to an already complex acoustic record. Measurements of sparsely distributed, spatially compact fish aggregations have been conducted in the Gulf of Maine using a long-range, broadband sonar with continuous coverage over the frequency band of 1.5–5 kHz. Concurrent downward-looking, multi-frequency echosounder measurements (18, 38, and 120 kHz), and net samples of fish are used in conjunction with physics-based acoustic models to classify and statistically characterize the long-range fish echoes. A significant number of echoes, which are at least 15 dB above background levels, were observed in the long-range data and classified as due to mixed assemblages of swimbladder-bearing fish. These aggregations of fish produce highly non-Rayleigh distributions of echo magnitudes. The probability density functions of the echoes are accurately predicted by a computationally efficient, physics-based model that accounts for beam-pattern and waveguide effects as well as the scattering response of aggregations of fish. [Work supported by the U.S. Office of Naval Research.]

Contributed Papers

8:40

1aAO3. Getting more for less: Increasing the accessibility of water column sonar data for fisheries management. Carrie C. Wall, Charles Anderson (Cooperative Inst. for Res. in Environ. Sci., Univ. of Colorado at Boulder, 216 UCB, Boulder, CO 80309, carrie.bell@colorado.edu), and Susan J. McLean (National Geophysical Data Ctr., NOAA, Boulder, CO)

Active acoustic technology is of increasing importance for studies examining fish populations and biological abundance in the water column. Multibeam echosounders are employed routinely on NOAA fishery vessels to estimate biomass, conduct trophic- and species-level identification, measure school morphology and behavior, and characterize habitat for commercially important species. These surveys deliver valuable information for ecosystem-based fisheries management but they also produce massive amounts of data that are costly and difficult to maintain. With its ability to store and preserve large datasets, NOAA's National Geophysical Data Center is acquiring and archiving acoustic data collected from NOAA and academic fleets. Through these efforts, an accessible archive of acoustic water column data will be made available to researchers and the public around the world. A web-based search engine will allow anyone to identify where data were collected, what instrument was used, and access the raw data and associated products. Years of decreasing funding for the sciences have necessitated our ability to get more information and more users out of data currently collected. This globally accessible archive is a large step in that direction. Of most importance is identifying how best to tap the archive to benefit current and future fisheries research and management.

8:55

1aAO4. Detection of fish near the bottom with a hull-mounted multibeam echosounder. Christian de Moustier (2535 Midway Dr. #81777, San Diego, CA 92138, cpm@ieee.org)

In single-beam or split-beam fisheries echosounding, bottom echoes often mask echoes from fish hovering near the bottom. Likewise, in down-looking multibeam echosounding bottom echoes received in the main lobe of near normal incidence beams appear in the sidelobes of the other beams and obscure weaker echoes received at the same range in the mainlobe of these other beams. Some multibeam sonars use frequency division multiplexing to avoid crosstalk between beams. However, it was shown recently [de Moustier, *Proc. MTS/IEEE OCEANS'13*, San Diego, September 23–27, 2013] that detection of fish near the seafloor is possible with a multibeam echo-sounder operating at a single acoustic frequency. This is achieved with a signal processing technique based on the ordered statistic constant false alarm rate (OS-CFAR) detection method used in radar. Here, the OS-CFAR operator is applied in the angle domain, rather than the customary range domain, and it estimates the signal-to-clutter ratio across all angles at a given time slice or range increment at the output of the beamformer. In this context, clutter encompasses both clutter and noise against which the desired signal is to be detected. Data collected with a 160 kHz multibeam echosounder are used to demonstrate the technique.

9:10

1aAO5. Observations of fission/fusion processes in fish aggregations using a multibeam echosounder. Thomas C. Weber (Ctr. for Coastal and Ocean Mapping, Univ. of New Hampshire, 24 Colovos Rd., Durham, NH 03824, weber@ccom.unh.edu), Daniel Grunbaum (School of Oceanogr., Univ. of Washington, Seattle, WA), and Timothy K. Stanton (Dept. of Appl. Ocean Phys. and Eng., Woods Hole Oceanogr. Inst., Woods Hole, MA)

Models of fish behavior within aggregations typically incorporate a fish's ability to sense near neighbors. Some newer models also include a cognitive functionality that allows the fish to understand not only the action of the near neighbors but also their intent. This cognitive function might be manifested, for example, in some stochastic estimate of the movement of a local population rather than at the individual level. Inherent in such a function are the temporal and spatial scales at which a fish's cognitive function operates. To help constrain these scales we have analyzed acoustic backscatter from walleye pollock collected with a multibeam echosounder as part of the NOAA Alaska Fisheries Science Center survey. During this survey, repeat passes that were approximately 1 nmi long were collected at approximately 15 mi intervals over an aggregation of fish. In at least one case, we have been able to acoustically observe an initial group of fish that undergoes both fission (splitting) and fusion (recombining) behaviors. In doing so, we are able to track the net movement, speed, and size of various parts of the group, thereby providing some ground truth for cognitive functionality models. [Research supported by the U.S. Office of Naval Research.]

9:25

1aAO6. Phased-array Doppler sonar measurements at the equator: Currents or swimming? Jerry A. Smith (SIO, UCSD, 9500 Gilman Dr., M.S. 0213, La Jolla, CA 92093-0213, jasmith@ucsd.edu)

A 64-channel 200 kHz phased array Doppler sonar (PADS) was deployed on the Equator at 140 W, sampling a vertical slice of the ocean, from 9 m to 200 m depth by 100 degrees, twice per second. The instrument was operated for two nearly continuous time-series, Oct. 10–20 and Oct.24 to Nov. 3, 2012. While the PADS was operated off the Starboard side of the R/V Revelle, a "Fast-CTD" (FCTD) was simultaneously operated off the port stern, sampling to 250 m every 2.5 min. Headway was maintained at less than 1 m/s relative to the surface flow (which was small, in contrast to the ~1.5 m/s undercurrent at 100 to 130 m depth). Because of this slow headway, motile scatterers were able to develop ship-centric swimming patterns, particularly at night. This, in turn, introduced some subtly non-physical characteristics in the estimated vertical velocities in particular. Animations of the backscatter intensity reveal a variety of scales and speeds of the scatterers. Visual nighttime inspection revealed a preponderance of small squid (< 30 cm) with the occasional large predator passing rapidly through (not identified). In spite of this biological interference, reasonable profiles of horizontal velocity were produced via a simple de-spiking selector.

9:40

1aAO7. Impacts of the warming of the continental shelf in spring 2012 on acoustic propagation conditions and fish distributions north of Cape Hatteras. Glen Gawarkiewicz, Ke Chen, James F. Lynch (Woods Hole Oceanogr. Inst., M.S. #21, Woods Hole, MA 02543, ggawarkiewicz@whoi.edu), Thomas M. Grothues (Rutgers Univ., NB, NJ), Arthur Newhall, and Ying-Tsong Lin (Woods Hole Oceanogr. Inst., Woods Hole, MA)

During May 2012, we conducted hydrographic surveys in conjunction with studies of acoustic scattering from fish schools north of Cape Hatteras. The waters of the continental shelf were greater than 4° Degrees C. warmer than prior observations during typical spring-time conditions in May 1996. In addition, the temperature gradients which normally exist across the shelfbreak front were absent, leading to intensification of the shelfbreak frontal jet. We relate the warming to large-scale atmospheric shifts and also report on the absence of cold water fish species, which were expected to be abundant in the study area.

9:55–10:10 Break

10:10

1aAO8. Deep-diving autonomous underwater vehicle provides insights into scattering layer dynamics. Kelly J. Benoit-Bird (College of Earth, Ocean & Atmos. Sci., Oregon State Univ., 104 CEOAS Admin Bldg., Corvallis, OR 97331-0000, kbenoit@coas.oregonstate.edu)

Organisms within deep scattering layers are often too densely packed to be ensonified as individuals using surface or seafloor based sensors, are too fast to be easily captured by research nets yet too small for most fishing gear, and are mixed with other individuals, making it difficult to interpret acoustic data from these ecologically important animals. To address this, we integrated a two-frequency, split-beam echosounder into an autonomous underwater vehicle (AUV) capable of flight at 600 m. As part of a study on whale foraging ecology off the Channel Islands, California, we flew the echosounders through scattering layers found at three depth ranges. Echoes were obtained from individual scatterers within layers being foraged upon by Risso's dolphins. Examining the echo statistics throughout layers revealed remarkable heterogeneity of echo strength and frequency response within layers that generally appeared homogeneous with respect to these same characteristics from ship-based echosounders. Some layers were internally layered but most features showed distinct, small patches of similar scatterers adjacent to those with different characteristics. The extensive horizontal coverage and near target sampling permitted by the AUV-based echosounders are providing a new understanding of the scales of biological organization within horizontally extensive scattering features.

10:25

1aAO9. Mapping the scatterscape of pelagic side scan sonar targets relative to oceanographic features. Thomas M. Grothues (Inst. of Marine and Coastal Sci., Rutgers Univ., Marine Field Station, 800 c/o 132 Great Bay Blvd., Tuckerton, NJ 08087, grothues@marine.rutgers.edu), Arthur E. Newhall, James F. Lynch, Glen G. Gawarkiewicz (Woods Hole Oceanogr. Inst., Woods Hole, MA), and Kaela S. Vogel (Dept. of Marine Biology, Univ. of North Carolina, Wilmington, Wilmington, NC)

Sonar reconnaissance of fishes for stock assessment and research has been an effective and minimally invasive method of gathering abundance and distribution data on scales of 10s to 100s of km since the 1950s. Yet, classification of fishes remains one of the greatest challenges of active sonar surveys. Many variables affect sonar reflection, including size, shape, orientation to the sonar source, the spatial relationship of individuals in a school to each other, and the number and distribution of individuals within a school. The long wavelengths of low frequency (typically $<$60 kHz) that allow depth penetration provide poor small scale resolution for identifying objects. High frequency side scan sonar (600—900 kHz), while imaging only over short ranges, can resolve individual fish and thus orientation and behavior relevant to understanding low frequency sonar returns and ecology. We demonstrate here that autonomous underwater vehicles (AUVs) offer a mechanism for putting side scan sonar transducers near potential targets together with telemetry, imaging, and oceanographic sensors, and can thus work together with low frequency sonar to develop holistic scatterscapes of oceanographic features, inclusive of information on species identity, orientation, behavior, abundance, individual size, and feature stability.

10:40

1aAO10. Scattering and reverberation from fish schools in the 500–1500 Hz band. Arthur Newhall, Ying-Tsong Lin, James F. Lynch (Appl. Ocean Phys. and Eng., Woods Hole Oceanogr. Inst., 210 Bigelow Lab, MS #11, Woods Hole, MA 02543, anewhall@whoi.edu), Glen G. Gawarkiewicz (Physical Oceanogr., Woods Hole Oceanogr., Woods Hole, MA), and Thomas M. Grothues (Rutgers Univ., Tuckerton, NJ)

We will report on the results from an experiment off Cape Hatteras, North Carolina, to look at scattering and reverberation from fish schools in the 500–1500 Hz band. The experiment, which was performed during the period May 12–29, 2012, was a joint acoustics, biology, and physical oceanography effort, with distinct, but coordinated, goals in each area. Acoustically, we wished to examine the scattering of sound from fish schools over a full range of azimuthal angles. To do this, we employed a source mounted on an autonomous vehicle and a moored, four element array receiver. The source traveled around the fish school and the receiver, giving the desired angular diversity. Video images, sidescan sonar, and direct sampling of the school allowed us to quantify the *in-situ* scattering field. Estimates for attenuation and scattering versus azimuthal angle will be presented. Directions for analysis and further research will be discussed.

10:55

1aAO11. Deep sea organisms density estimation using a Dual Frequency Identification Sonar (DIDSON). Giacomo Giorli (Oceanogr., Univ. of Hawaii at Manoa, 1000 Pope Rd., Honolulu, HI 96822, giacomog@hawaii.edu), Adrienne M. Copeland, Whitlow W. Au (Hawaii Inst. of Marine Biology, Univ. of Hawaii at Manoa, Kaneohe, HI), and Jeff Polovina (Pacific Island Fisheries Sci. Ctr., NOAA, Honolulu, HI)

Estimating the density of organism living in deep sea scatting layers is of key importance for understanding the biomass in the mesopelagic layers. Scientific echosounders are routinely used for this task; however, new imaging sonar technologies pose the opportunity for estimating density of organism, as well as identification at smaller scales. During the 2013 NOAA KONA Integrated Ecosystem, a Dual Frequency Identification Sonar (DIDSON) (SoundMetrics Inc.) was used to estimate the density, length of organisms in the deep sea scattering layers during nighttime and daytime along the KONA coast of the island of Hawaii. At each station, an EK60 38 kHz echosounder was used to find the depth of the deep sea scattering layers, and the DIDSON was lower into the layer (or layers if two were present) (about 500 and 600 m), and underneath the deeper layer (about 800 m). A total of 4621 organisms were counted and sized. We estimated densities ranging from 6 to 1 organism/m^3. Density shows some variation between locations and depth and organism as big as 3 m were sighted.

11:10

1aAO12. Bioacoustic absorption spectroscopy measurements at the shelf beak off Oregon. Orest Diachok (Appl. Phys. Lab., Johns Hopkins Univ., 11100 Johns Hopkins Rd., Laurel, MD 20723, orestdia@aol.com)

This paper describes the results of a Bioacoustic Absorption Spectroscopy (BAS) experiment, which was conducted at a biological hot spot at the shelf break off Oregon in August 2012. The location of the hot spot was identified by a NW Fisheries Science Center survey. This experiment included coincident measurements of transmission loss (TL), fish layer depths, fish length distributions, and continuous temperature profiles. The objective was to derive the bioacoustic parameters of hake (a physoclist), the dominant species in this region and other species from measurements of TL vs. range (0–10 km), frequency (0.3–5 kHz), depth (0–260 m), and time of day, and demonstrate consistency with trawl and echo sounder data. TL measurements were conducted between a moving, ship-deployed, broadband source and a 24 element, water column-spanning vertical array, which were provided by the Naval Research Laboratory. This was the first BAS experiment that targeted a physoclist. Previous BAS experiments targeted physostomes. The TL data exhibited absorption lines that were as high as 2 dB/km. Absorption lines at relatively high frequencies, which were observed on near-surface and mid-ocean hydrophones, are attributed to myctophids. Absorption lines, which were observed at relatively low frequencies on near-bottom hydrophones, are attributed to hake.

11:25–11:50 Panel Discussion

Session 1aBA

Biomedical Acoustics: Breast Ultrasound I

Koen W. A. van Dongen, Cochair
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Timothy E. Doyle, Cochair
Physics, Utah Valley Univ., MS 179, 800 W. University Parkway, Orem, UT 84058-5999

Chair's Introduction—7:55

Invited Papers

8:00

1aBA1. Three-dimensional ultrasound computer tomography at Karlsruhe Institute of Technology (KIT). Nicole V. Ruiter, Michael Zapf, Torsten Hopp, Ernst Kretzek, and Hartmut Gemmeke (Inst. for Data Processing and Electronics, Karlsruhe Inst. of Technol., Postfach 3640, Karlsruhe 76021, Germany, nicole.ruiter@kit.edu)

The KIT 3D USCT surrounds the breast with ultrasound transducers on a 3D aperture and emits and receives nearly spherical wave fronts for synthetic aperture focusing. This full 3D system achieves isotropic 3D resolution, has a nearly spatial invariant point spread function, and allows fast data acquisition. The 3D USCT device is equipped with 2041 ultrasound transducers. The acquisition is carried out by sequentially selecting a single emitter, sending a chirp at 2.5 MHz center frequency and recording the transmitted and reflected waves with all receivers. Rotational and translational movement of the aperture is applied to enhance the image contrast. Up to 40 GB of raw data is acquired with 480 parallel channels for digitization at 12 bit and 20 MHz sampling frequency. In a first pilot study, ten patients with different lesions were imaged. Speed of sound, attenuation, and reflection images of each patient were derived from the raw data. Overlaid volumes of the modalities show qualitative and quantitative information at a glance. The results are promising because the breasts' tissue structures and cancerous lesions could be identified in the USCT images.

8:20

1aBA2. Quantitative three dimensional nonlinear inverse scattering and reflection breast imaging: Initial clinical results. James Wiskin, David Borup, Elaine Iuanow, John Klock, and Mark Lenox (CVUS, LLC, Inc., 3216 Highland Dr., Ste. 100, Salt Lake City, UT 84106, jwiskin.cvus@gmail.com)

Water bath breast scanners utilize either ray based or inverse scattering techniques for the quantitative images. The 3D inverse scattering approach we use requires a 3D forward and back-propagation problem to be solved, resulting in ~1.2 mm resolution images. The resulting speed of sound map is then used in a 3D inhomogeneous, eikonal equation based reflection algorithm to account for refraction effects, and yield a 3D co-registered speckle free 360 degree compounded B-scan like volume. There is no harmful ionizing radiation, no compression, and no required contrast agents for proper utilization of our device. The quantitative transmission ultrasound (QTUS) images are independent of operator skill. The patient lies prone on a full length table with the breast pendant in the water bath. The breast is gently immobilized with the use of a breast retention pad that magnetically "attaches" to a magnetic retention rod for imaging of the breast. Scan time for each breast is approximately 10 min. We will show full 3D quantitative images obtained from consented, anonymous patients from several academic collaborating institutions. We will compare our QTUS images with mammographic, MRI, and hand held US images, and correlate biopsy results where appropriate.

8:40

1aBA3. Clinical breast imaging with ultrasound tomography: A description of the SoftVue system. Neb Duric, Peter Littrup (Oncology, Karmanos Cancer Inst., 4100 John R, Detroit, MI 48201, duric@karmanos.org), Olivier Roy, Cuiping Li, Steve Schmidt, Xiaoyang Cheng, and Roman Janer (Delphinus Medical Technologies, Plymouth, MI)

We describe the technical design and performance of SoftVue, a breast imaging device based on the principles of ultrasound tomography. SoftVue's imaging sensor is a ring shaped transducer operating at 3 MHz and consisting of 2048 elements. Data acquisition is achieved through 512 receive channels. The transducer encircles the breast, which is immersed in warm water while the patient lies in a prone position. The transducer is translated vertically to acquire data from the entire breast. The acquired data are used to reconstruct images using tomographic inversions. The reconstruction engine is based around a blade-server design that houses multiple CPUs and GPUs. Separate algorithms are used to reconstruct reflection, sound speed, and attenuation images. A patient scan generates a stack of each type of image. The system was designed with the clinical goal of detecting and characterizing breast masses based on their biomechanical and morphological properties. Ongoing clinical studies are being used to assess the performance of the system under realistic clinical settings. Results of the clinical assessment are presented. [The authors acknowledge research support from the National Cancer Institute (5 R44 CA165320-03). Neb Duric and Peter Littrup also acknowledge that they have financial interests in the SoftVue technology.]

9:00

1aBA4. High definition ultrasound of the breast. Robert Kruger, Richard Lam, Daniel Reinecke, and Stephen Del Rio (OptoSonics, Inc., 108 Straight Rd., Oriental, NC 28571, bobkruger@optosonics.com)

We propose to develop high definition ultrasound (HD-US) based on a hemispherical synthetic aperture and backscattered sonic waves. This configuration will produce direct three-dimensional maps of ultrasound reflectivity of breast tissues and will display higher signal-to-noise, reduced speckle, and greater spatial resolution compared to ultrasound arrays based on linear or curved planar sampling apertures. Our prototype hemispherical array consists of four interdigitated sub-arrays, each having 128 discrete elements. Currently, this array is used to capture 3D photoacoustic images of the whole breast using a spiral scanning strategy to increase the field of view sufficiently to perform whole-breast screening—photoacoustic mammography (PAM). We will add a multiplexed pulser, pulse sequencer, and T/R switches to capture ultrasound reflectivity data sufficient to form a 3D ultrasound image using the same reconstruction strategy used in PAM. Full 3D image acquisition will take 1.7 min, the same as currently used for PAM data collection. HD-US will be combined with PAM to produce 3D images of soft tissue and microcalcifications. The 3D images will be co-registered with the PAM images of hemoglobin distribution in the breast using a single hemispherical transducer array.

9:20

1aBA5. Low-contrast lesion detection enhancement using pulse compression technique. Serge Mensah, Julien Rouyer, Arnaud Ritou, and Philippe Lasaygues (Acoust. and Mech. Lab., National Ctr. for Sci. Res., 31 Chemin Joseph Aiguier, Marseille 13402, France, mensah@lma.cnrs-mrs.fr)

Pulse compression methods greatly improve the quality of medical images. In comparison with standard broadband pulse techniques, these methods enhance the contrast-to-noise ratio and increase the probing depth without any perceptible loss of spatial resolution. The Golay compression technique is analyzed here in the context of ultrasonic computed tomography, first on a one-dimensional target and second, on a very low-contrast phantom probed using a half-ring array tomograph. The imaging performances were assessed based on both the point spread function properties and the image contrast-to-noise ratio. The improvement obtained in the image contrast-to-noise ratio (up to 40%) depends, however, on the number of coherently associated diffraction projections. Beyond a certain number, few advantages were observed. Advances in ultrasound computed tomography suggest that pulse compression methods should provide a useful means of optimizing the trade-off between the image quality and the probing sampling density. It could also be used to accelerate the reconstruction process during the examination of patients. The results of this study also suggest that when it is proposed to search for very low contrast lesions (such as diffuse lobar carcinomas in breast cancer), once the number of projections has been set for a given tomographic set-up, the image contrast, i.e., the probability of detection, can been enhanced by using low-power, high-energy Golay sequences

9:40–10:00 Break

Contributed Papers

10:00

1aBA6. The effect of crosstalk in a circular transducer array on ultrasound transmission tomography of breast. Krzysztof J. Opielinski, Piotr Pruchnicki (Faculty of Electronics, Wroclaw Univ. of Technol., Wyb. Wyspianskiego 27, Wroclaw 50-370, Poland, krzysztof.opielinski@pwr.wroc.pl), Wlodzimierz Roguski (DRAMINSKI Medical Instruments, Olsztyn, Poland), Mateusz Celmer, Tadeusz Gudra (Faculty of Electronics, Wroclaw Univ. of Technol., Wroclaw, Poland), Jaroslaw Majewski (Faculty of Telecommunications, Comput. Sci. and Elec. Eng., Univ. of Technol. and Life Sci. in Bydgoszcz, Bydgoszcz, Poland), Mariusz Bulkowski, Tomasz Piotrowski, and Andrzej Wiktorowicz (DRAMINSKI Medical Instruments, Olsztyn, Poland)

Various types of undesirable crosstalk can occur in ultrasonic arrays consisting of a number of elementary piezoelectric transducers. Crosstalk is a result of deficiency in electrical or mechanical isolation between array elements. It is a function of proximity between piezoelectric transducers of the array or between connections. Small piezoceramic transducers tightly located inside of a circle (cylinder or sphere in 3-D systems) are mostly used for *in vivo* ultrasound tomography imaging of breast tissue. This means that proper switching of transducers makes it possible to quickly acquire tomographic transmission and reflection data for all directions around the object. A problem with such arrangements is the occurrence of crosstalk, which introduce specific errors to measurement data. Transmission and reflection tomographic images reconstructed based on ultrasonic measurements with crosstalk are distorted in a characteristic way. This work analyzes the effect of crosstalk in a circular transducer array on ultrasound transmission tomography imaging—especially, when used for examination of breast tissue.

10:15

1aBA7. Improving ultrasound-based estimates of vector displacement fields in elastography applications. Sanjay S. Yengul, Olalekan A. Babaniyi (Mech. Eng., Boston Univ., 69 Spyglass Hill Dr., Ashland, MA 01721, ysanjay@gmail.com), Faik C. Meral, Bruno Madore (Radiology, Brigham and Women's Hospital, Boston, MA), and Paul E. Barbone (Mech. Eng., Boston Univ., Boston, MA)

Ultrasound elastography is based on estimating tissue displacement using ultrasound imaging technology. The displacement field of interest may be the result of either an external quasistatic deformation (i.e., strain imaging), or a propagating shear wave (i.e., radiation force imaging), or some other excitation. Ultrasound image data with standard beamforming can provide very precise measurements of soft tissue displacement in the axial direction, i.e., in the direction of the ultrasound beam. Lateral (and elevational) displacement estimates are relatively noisy. This observation demonstrates that the quality of displacement estimates depends on the ultrasound image formation method. In this contribution, we evaluate the precision of axial and lateral displacement estimates obtained using different image formation methods. Ultrasound data from a Verasonics V-1 system (Redmond, WA) is used, which provides access to the raw signal from each element of the transducer array, thereby allowing beamforming to be done in software as a post-processing step. Using information from a mechanics-based model [Babaniyi *et al*., J. Acoust. Soc. Am. **134**, 4011 (2013)] further improves the precision of the displacement field estimate. We will present the results and discuss the next steps.

10:30

1aBA8. Boundary conditions in quantitative elastic modulus imaging.
Daniel T. Seidl, Olalekan A. Babaniyi (Mech. Eng., Boston Univ., 110 Cummington Mall, Boston, MA 02215, tseidl@bu.edu), Assad A. Oberai (Mech., Aerosp. and Nuclear Eng., Rensselaer Polytechnic Inst., Troy, NY), and Paul E. Barbone (Mech. Eng., Boston Univ., Boston, MA)

Quantitative elastic modulus imaging from quasistatic strain fields has several advantages over other approaches. It requires no specialized hardware, provides spatial resolution nearly commensurate with Bmode, and may be extended to quantitative nonlinear modulus imaging in a straightforward way. It has the drawback, however, of requiring the solution of a complex inverse problem with an ultrasound measured displacement field. The most common approach to solving the inverse problem is iterative optimization. To solve the inverse problem, the experimental configuration is simulated. Material property distributions in the simulated experiment are then varied until the simulated deformation field matches the observed deformation field. The weak link in this process is uncertainty in the experimental conditions. In the context of iterative inversion, this translates into uncertainty in the boundary conditions of the forward model. In this talk, we discuss how different choices of displacement and/or traction boundary conditions affect the inverse problem's solution using phantom and clinical breast data. We show that a Bayesian estimate of the displacement field in the face of uncertain boundary conditions can be implemented by spring finite elements on the domain boundary. [Authors gratefully acknowledge funding from NSF and NIH (NSF SI2 Grant No. 1148111; NIH NCI-R01CA140271).]

10:45

1aBA9. Boundary condition-free elastic modulus reconstructions from ultrasound measured quasi-static displacements.
Olalekan A. Babaniyi, Daniel T. Seidl (Mech. Eng., Boston Univ., 730 Commonwealth Ave., Boston, MA 02215, lekanb@bu.edu), Assad A. Oberai (Mech., Aerosp., and Nuclear Eng., Rensselear Polytechnic Inst., Troy, NY), Michael S. Richards (Dept. of Surgery, Univ. of Rochester, Rochester, NY), and Paul E. Barbone (Mech. Eng., Boston Univ., Boston, MA)

Quantitative elastic modulus imaging from quasi-static displacement data requires the solution of an inverse elasticity problem. The inverse problem formulation generally requires specification of either displacement or traction boundary conditions. Most current ultrasound devices are not capable of measuring traction data, and the measured displacement field is noisy. The incomplete and imprecise nature of the available boundary information often necessitates that educated guesses be made in order to have adequate knowledge of the boundary conditions to compute mechanical properties. These assumed boundary conditions, however, can to lead to errors in the reconstructions. This abstract proposes a method to perform reconstructions without knowing the boundary conditions *a priori*. This method relies on using the constrain imposed by the equilibrium equation and an optimization algorithm to estimate the modulus field. This method was verified with simulated displacement data, validated with phantom displacement data, and applied to *in-vivo* displacement data measured from patients with breast masses. [Authors gratefully acknowledge funding from NSF and NIH (NSF Grant No. 50201109; NIH NCI-R01CA140271).]

11:00

1aBA10. Comparative analysis of small versus large transducers for high-frequency ultrasonic testing of breast cancer.
Madison J. Peterson, Nathan M. Bliss (Biology, Utah Valley Univ., 904 N 960 W, Orem, UT 84057, madisonpr@aol.com), and Timothy E. Doyle (Phys., Utah Valley Univ., Orem, UT)

High-frequency ultrasound (20–80 MHz) has been found to be sensitive to margin pathology from breast cancer surgery. In order to improve the resolution and sensitivity of this method, transducers are needed that have smaller piezoelectric elements than those currently in use. This study's purpose was to determine if small-element transducers (Blatek, 50 MHz, diameter <2 mm) produce similar results as those from large-element transducers (Olympus NDT, 50 MHz, 6.35-mm diameter). Pulse-echo and through-transmission measurements were performed on bovine heart tissue and 10 phantom specimens containing chopped nylon fibers and polyethylene microspheres. The density of peaks in the ultrasonic spectra of the small or mini transducers (MT) paralleled those of the large transducers (LT) in the bovine tissue, with higher peak densities associated with connective tissue and lower peak densities with muscle tissue. The MT data from the phantoms showed greater variance than the LT data, indicating that the MT were more sensitive to the heterogeneous wavefields arising from microsphere scattering. Additional *in vivo* testing is currently being performed on breast tumors grown in mice treated with Avastin. Small-element transducers may ultimately provide *in vivo* cancer detection in margins, allowing more precise excision of cancerous tissue and thus eliminating follow-up surgeries.

Session 1aED

Education in Acoustics and Physical Acoustics: Tools for Teaching Advanced Acoustics

David T. Bradley, Cochair
Physics + Astronomy, Vassar College, 124 Raymond Ave., #745, Poughkeepsie, NY 12604

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

Invited Papers

8:30

1aED1. An information-rich learning environment for instruction in acoustics, part 3. Robert Celmer (Acoust. Prog. & Lab, Univ. of Hartford, 200 Bloomfield Ave., West Hartford, CT 06117, celmer@hartford.edu)

The static written word has always had its limits when it comes to learning about sound. Much of the subject matter is dynamic, multifaceted, and of course, aural. This presentation will describe additional multimedia materials developed for in-class presentation and self-paced review exercises for acoustics instruction at the University of Hartford. Some of the materials were developed using certain authoring applications, draw and animation programs, sound manipulation software, as well as 3D-CAD and spectral analysis applets. Audio equipment used in class as well as acoustic treatments of the classroom/listening environment will be described. This approach to acoustic pedagogy will be discussed in the context of a student-centered learning environment. New and updated demonstrations of the materials for the instruction of acoustical concepts as well as case studies will be presented.

8:50

1aED2. Fun with levitators. R. Glynn Holt (Mech. Eng., Boston Univ., Dept. of Mech. Eng., 110 Cummington Mall, Boston, MA 02215, rgholt@bu.edu)

The acoustic radiation force is enjoying something of a comeback on the celebrity circuit these days. Current applications are seen, for example, in elasticity imaging in biomedical ultrasound, and separation technologies in the biomedical and petroleum arenas. But of course the application of radiation force to acoustic levitation has a long history, and in this talk, we will explore levitation to illustrate the principles of radiation force in standing waves with sample inclusions. With a little luck, both large bubbles in water and liquid drops in air will be demonstrated.

9:10

1aED3. Animations illustrating the reflection of longitudinal sound wave pulses. Daniel A. Russell (Graduate Program in Acoust., Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, drussell@engr.psu.edu)

The reflection of transverse wave pulses from a fixed or free boundary may be demonstrated physically in the classroom using an elastic string under tension. Animations of transverse wave pulses reflecting from hard and soft boundaries clearly indicate the phase relationship between incident and reflected signals. However, demonstrating similar phenomena for longitudinal waves difficult. The reversal of longitudinal waves in a spring may be physically demonstrated, but not the phase relationship between incident and reflected wave pulses. A microphone and digital oscilloscope may be used to demonstrate the phase change in pressure for reflections from the open end of a pipe, and the lack of phase change for reflections from a rigid end, but such demonstrations do not illustrate the longitudinal behavior of the wave motion at the boundary during reflection. This paper will showcase animations of longitudinal wave pulses reflecting from fixed and free boundaries, with an emphasis the appropriate phase changes upon reflection, with clarification of relationships between particle displacement, particle velocity, and pressure for longitudinal waves traveling in positive and negative directions. Discussion will include ways these animations may be used to improve student understanding.

9:30

1aED4. Bringing MATLAB into the acoustics class. Joseph F. Vignola, Aldo A. Glean (Mech. Eng., The Catholic Univ. of America, 620 Michigan Ave., NE, Washington, DC 20064, vignola@cua.edu), Teresa J. Ryan (Eng., East Carolina Univ., Greanville, NC), and Diego Turo (BioEng., George Mason Univ., Fairfax, VA)

The Catholic University of America has a history of graduate education in acoustics that ranges back to the 1930s. Many of the students served by CUA have been and continue to be working professionals in careers related to acoustics. Over time, we have observed that many of our students come to us having developed an impressive depth of knowledge in their specialty. However, some of these students have little of the formal acoustics training needed to provide broader context. This presentation will discuss teaching practices designed to serve this cohort as well as more traditional graduate students. We create an experiential learning environment that capitalizes on the fact that many simple but powerfully instructive measurements can be made with now ubiquitous items. All modern laptops

have both speakers and a microphone. A laptop, coupled with a library of MATLAB code, gives students the opportunity to explore many important topics. This presentation includes examples from room acoustics, musical acoustics, and elastic wave propagation.

9:50

1aED5. Teaching acoustics in the time domain and frequency domain: Going back and forth. Karim G. Sabra (Mech. Eng., Georgia Inst. of Technol., 771 Ferst Dr., NW, Atlanta, GA 30332-0405, karim.sabra@me.gatech.edu)

Our common experience with acoustic signals typically occurs in the time domain: for instance when listening to a conversation or recording an audio signal. On the other hand, acoustics in the classroom is often taught in the frequency domain; for instance, doing so can offer clear simplification of the wave equation (e.g., when solving the Helmholtz equation). Exact time-domain solutions can ultimately be obtained back from these frequency solutions using Fourier synthesis; but it typically requires numerical methods or extra analytical work. However, based on classroom experiences, it can be valuable that students develop an intuitive interpretation of these frequency-domain solutions in the time-domain without using any computers or actual inverse Fourier transform evaluation. For instance, a question one may ask students is "how would you predict the overall shape and features of time-domain waveform corresponding to this specific frequency-domain solutions we just derived, i.e., as if you were to perform an actual experiment and just recorded a time-domain waveform." I will discuss various basic examples covered in the classroom such as modal propagation and dispersion effects in a waveguide.

10:10–10:20 Break

10:20

1aED6. Using Python to teach mathematics, physics, and acoustics. Derek C. Thomas and Benjamin Y. Christensen (Dept. of Phys. and Astronomy, Brigham Young Univ., N283 ESC, Provo, UT 84602, dthomas@byu.edu)

Advanced technical courses often suffer from a lack of interactive materials. Common tools to remedy this deficiency include MATLAB and MATHEMATICA, both of which can be prohibitively expensive to obtain outside of the university environment. Python is a scripted language that is easy to read and use and is rapidly emerging as a lingua franca for scientific computing due to the flexibility and facility of the language, the large and active community, and the large number of high quality scientific libraries that are available in Python. Python provides a free and open source tool to develop classroom materials that students can modify and extend. We discuss the use of Python in teaching advanced topics in mathematics, physics, and acoustics. Examples are drawn from courses in acoustics, mechanics, and mathematical and computational physics

10:40

1aED7. Resources for teaching near-field acoustical holography in advanced acoustics courses. Kent L. Gee, Tracianne B. Neilsen, and Scott D. Sommerfeldt (Dept. of Phys. and Astronomy, Brigham Young Univ., N243 ESC, Provo, UT 84602, kentgee@byu.edu)

The principles of planar near-field acoustical holography (NAH) can be used to motivate discussion of, or illustrate, physics of sound radiation or topics in signal processing. These include separable geometries, the Helmholtz equation, spatial Fourier transforms and the wavenumber domain, superposition of waves, the relationship between pressure and particle velocity, near and far fields, the radiation circle and evanescence, filtering, and signal-to-noise ratio. This paper describes the incorporation of NAH as part of a graduate-course unit on structural acoustics. Resources discussed, and which will be made available to educators, include a basic NAH processing script and example data collected by students using an automated positioning system in Brigham Young University's fully anechoic chamber.

Contributed Papers

11:00

1aED8. Software usage for synthesized sound in acoustics education. Jennifer K. Whiting, Katherine H. Fortney, Tracianne B. Neilsen, and Kent L. Gee (Dept. of Phys. and Astronomy, Brigham Young Univ., C110 ESC, Provo, UT 84606, lundjenny@comcast.net)

Musical acoustics engages students of many backgrounds when studying acoustics. Free software in the form of digital audio workstations can be used to provide students with a hands-on instruction to concepts of synthesis. Students can easily manipulate variables to add or detract from the realism of synthesized sound. This process of synthesizing sound provides a means for students to understand the concepts of additive and subtractive synthesis, attack and decay times, reverberation times, and filters. Application of free digital audio workstation software in an undergraduate class at Brigham Young University and in the recently revamped ASA Outreach workshop will be discussed.

11:15

1aED9. Architectural acoustics illustrated, animated, designed, and built. Michael Ermann, Nawazish Nanji (Architecture + Design, Virginia Tech, 201 Cowgill Hall (0205), Blacksburg, VA 24061-0205, mermann@vt.edu), Vinny Argentina, Matt Yourshaw (School of the Visual Arts, Virginia Tech, Blacksburg, VA), Marie Zawistowski, Keith Zawistowski, Lauren Duda, Megumi Ezure, Samantha Stephenson, Taylor Terrill, Ian Shelton, Samantha Yeh, Kyle Lee, Huy Duong, Brent Sikora, Leo Naegele, Tyler Atkins, Derek Ellison, Margaret Nelson, Leah Schafer, and Emarie Skelton (Architecture + Design, Virginia Tech, Blacksburg, VA)

In this line of pedagogy, architectural acoustics is filtered through the graphic and built language of architecture. First, a series of animations were created to explain room acoustics to architecture students. The impulse response as a concept is inherently spatial and dynamic. It can be explained with text, and it can be explained more clearly when illustration is included,

but because the path of sound and the loudness at a receiver fluctuates with time, it can be best explained with a narrated animation (available online). Second, the physics of sound, room acoustics, and noise control were illustrated as part of a book, *Architectural Acoustics Illustrated* (Wiley, 2015). Building material choices, spatial relationships, best-practices, and data were explored through drawing; non-obvious and counter-intuitive graphic findings are presented. Finally, 15 architecture students explored room acoustics through ray tracing and auralization software. Then they designed and built an amphitheater for the town of Clifton Forge, Virginia. The completed project was widely published, was the subject of a documentary film, and earned award recognition from the American Institute of Architects.

11:30

1aED10. Demonstrations and laboratory experiments in a senior level acoustics course. Murray S. Korman (Phys. Dept., U.S. Naval Acad., 572 C Holloway Rd., Chauvenet Hall Rm. 295, Annapolis, MD 21402, korman@usna.edu)

The Physics Department at the U.S. Naval Academy has a senior level four credit course (SP436) in acoustics (called "Acoustics") that features a well-equipped laboratory including an anechoic chamber. The course is populated by Physics Majors along with a few Engineering Majors at times. This presentation will show how MATHEMATICA 9 is used in the laboratory portion of the course to enhance lecture topics and student computational assignments. It is "hands-on" which helps motivate and enhance learning. Students learn the fundamentals of Mathematica as part of the laboratory experience. Tasks after data collection, including analysis and write-up, are done with this software. Workstations include laptop personal computers, spectrum analyzers, oscilloscopes, laser Doppler vibrometers, and various accelerometers, mics, hydrophones, and ultrasonic transducers. A demonstration of standing waves in a cylindrical cavity will be presented as a representation of some of our featured lab experiments—including: Helmholtz resonators, linear and nonlinear vibration of a circular membrane, or circular elastic plate, flexural waves on a thin bar, Chladni plates, the hanging oscillating chain, spectral analysis (Fourier series, Fourier integral), sound speed vs. temperature and salinity, acoustic landmine detection, moving coil loud speaker, waves on strings, transmitting arrays, and wave guide studies.

11:45

1aED11. YouTube EDU: Inspiring interest in acoustics through online video. Michael B. Wilson (Acoust., Penn State, 1649 Highlandon Ct, State College, PA 16801, wilsomb@gmail.com)

The Internet is changing the face of education in the world today. More people have access to more information than ever before. The Khan Academy, iTunes U, YouTube EDU, and other programs are providing educational content for free to millions of Internet users worldwide. This content ranges from interesting facts that introduce a topic to entire undergraduate courses. And just as acoustics is an underrepresented science at the secondary and undergraduate levels, acoustics is an underrepresented science in the world of online education. But that is changing. Online content is being created focusing on clarifying misconceptions and sparking an interest in the field of acoustics. These videos are available without charge to everyone in the world.

MONDAY MORNING, 5 MAY 2014 BALLROOM A, 8:00 A.M. TO 12:00 NOON

Session 1aPP

Psychological and Physiological Acoustics: Temporal Processing; Pitch and Timbre Perception; Speech Perception (Poster Session)

Agnès C. Léger, Chair
Res. Lab. of Electronics, Massachusetts Inst. of Technol., Rm. 36-757, 77 Massachusetts Ave., Cambridge, MA 02139

All posters will be on display from 8:00 a.m. to 12:00 noon, To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 8:00 a.m. to 10.00 a.m. and contributors of even-numbered papers will be at their posters from 10:00 a.m. to 12:00 noon.

Contributed Papers

1aPP1. Does fundamental frequency-segregation interact with dip listening or spatial unmasking? Thibaud Leclére, Mathieu Lavandier (LGCB, ENTPE-Université de Lyon, rue Maurice Audin, Vaulx-en-Velin, Rḫnes 69518, France, thibaud.leclere@entpe.fr), and Mickael L. Deroche (Dept. of Otolaryngol., Johns Hopkins Univ. School of Medicine, Baltimore, MD)

Differences in fundamental frequency (F0) and location between a speech target and a masker as well as amplitude modulations in the masker are helpful cues to improve speech intelligibility in cocktail party situations. Each cue has been thoroughly investigated independently in many studies, but it remains unclear whether they interact with each other. Experiment 1 examined potential interactions between F0-segregation and dip listening while experiment 2 examined interactions between F0-segregation and spatial unmasking. Speech reception thresholds were measured for a monotonized or an intonated voice against eight types of harmonic complex interferers. In experiment 1, the eight interferers varied in F0 contour (monotonized or intonated), mean F0 (0 or 3 semitones above the target) and broadband temporal envelope (stationary or 1-voice modulated). In experiment 2, interferers varied in F0 contour, mean F0 and spatial location (colocated or separated from the target). Thirty-two listeners participated in each experiment. The results will be presented and discussed.

1aPP2. Speech intelligibility and masking release using temporal fine structure and recovered envelope cues for normal-hearing and hearing-impaired listeners. Agnes C. Leger, Charlotte M. Reed, Joseph G. Desloge, Jayaganesh Swaminathan, and Louis D. Braida (Res. Lab. of Electronics, Massachusetts Inst. of Technol., Rm. 36-757, 77 Massachusetts Ave., Cambridge, MA 02139, aleger@mit.edu)

Consonant-identification ability was examined in normal-hearing (NH) and hearing-impaired (HI) listeners in the presence of continuous and 10-Hz square-wave interrupted speech-shaped noise. The speech stimuli (16 consonants in a-C-a syllables) were processed to present envelope (ENV) cues, temporal fine-structure cues (TFS), or envelope cues recovered from TFS speech (RENV). ENV- and TFS-speech was generated by extracting the ENV or TFS component of the unprocessed speech in N adjacent bands (N = 40 for ENV-speech, N = 1 or 4 for TFS-speech; range = 80–8020 Hz). RENV-speech was generated by extracting the ENV component of both types of TFS-speech in 40 adjacent bands. NH listeners were tested at an SNR of −-10 dB and individual HI listeners were tested with SNR in the range of −-6 to + 5 dB. These values of SNR yielded consonant-identification scores of roughly 50%-correct for intact speech in continuous noise for each listener. HI listeners had poorer speech scores than NH listeners. For both groups, scores with TFS- and RENV-speech were very similar. Scores were higher in interrupted noise than in continuous noise (indicating substantial release from masking), except for unprocessed- and ENV-speech for HI listeners. Audibility, frequency selectivity, and forward masking were estimated for each listener and compared with speech identification.

1aPP3. Aging and lexical neighborhood effects in competing speech perception. Karen S. Helfer, Angela Costanzi, and Sarah Laakso (Commun. Disord., Univ. of Massachusetts Amherst, 358 N. Pleasant St., Amherst, MA 01002, khelfer@comdis.umass.edu)

Little is known about the extent to which lexical neighborhood effects are influenced by the presence of to-be-ignored messages. When the competing signal is understandable speech, words in the masker may activate their own lexical neighbors, causing increased competition for word identification. Older adults may have particular difficulty inhibiting lexical activation. This poster will present results of an examination of lexical neighborhood influences on competing speech perception. Sentences were developed in which we manipulated the neighborhood density and frequency of key words to create lexically easy (low density or high frequency of usage) and lexically difficult (high density or low frequency) stimuli lists. Pairs of sentences were created in which one sentence was lexically easy (in terms of density or frequency) and one was difficult. The target sentence was always spoken by the same talker and could be either lexically easy or difficult. Participants were younger (18–23 years), middle-aged (45–59 years), and older (>60 years) adults; they also completed a battery of cognitive tests. This poster will show results of analyses comparing lexical neighborhood effects among the participant groups, as well as how these effects are influenced by hearing loss and cognitive abilities. [Work supported by NIH DC012057.]

1aPP4. Relationship between pitch and rhythm perception with tonal sequences. Sandra J. Guzman, Robert Almeida, Karson Glass, Cody Elston (Audio Arts & Acoust., Columbia College Chicago, 3734 Kenilworth, Berwyn, IL 60402, sguzman@colum.edu), Valeriy Shafiro, and Stanley Sheft (Commun. Disord. and Sci., Rush Univ. Medical Ctr., Chicago, IL)

Past work has shown varying degrees of relationship between pitch and rhythm perception. Current work investigated the relationship between pitch and rhythm processing in a four-tone sequence-reconstruction task which places additional demands on short-term memory. Sequence tones either had a fixed duration (212 ms) with frequency randomly selected from a logarithmically scaled distribution (400–1750 Hz), a fixed frequency (837 Hz) with a randomly selected log scaled duration (75–600 ms), or a random frequency and duration. In initial conditions, the task was to assemble sequence elements to recreate the target sequence for each of the three sequence types. To evaluate effect of extraneous randomization, both frequency and duration were randomized in the final two conditions with only one of the two attributes defining the target sequence. When only one stimulus attribute was randomized, performance was significantly better with sequences defined by pitch rather than rhythmic variation. Combining pitch and rhythmic variations led to a slight improvement in performance, while the introduction of extraneous variation had little to no effect when either pitch or rhythm defined the target sequence. Overall, there was a wide performance range across listeners with listeners clustered primarily by ability to use pitch information. [Work supported by NIH.]

1aPP5. Examining the influence of forward, backward, and simultaneous notched noise on the mid-level hump in intensity discrimination. Elin Roverud and Elizabeth A. Strickland (Speech, Lang., and Hearing Sci., Purdue Univ., 712 Cincinnati St., Lafayette, IN 47901, eroverud@purdue.edu)

Psychoacoustical intensity discrimination limens (IDLs) for high frequency, short duration pedestals are poorer at mid levels than at lower and higher levels. It has been theorized that this so-called midlevel hump (MLH) may reflect mid-level cochlear compression, whereas the improvement at higher levels may be due to spread of excitation cues. To characterize intensity discrimination within a single frequency channel, researchers (e.g., Plack, 1998) have used simultaneous notched noise (NN) to limit cues from the spread of excitation to other channels. However, additional effects of the NN on the pedestal remain a matter of debate. The NN may provide a reference against which intensity judgments are made. Additionally, the NN may produce excitatory masking in the pedestal frequency channel, suppress the pedestal, and evoke cochlear gain reduction via the medial olivocochlear reflex. These latter two mechanisms change the basilar membrane compression slope, but operate over different time courses. In the present study, we examine the MLH with different durations of forward, backward, and simultaneous NN and pure tone maskers to isolate these potential mechanisms. Results will be interpreted using a computational model of the auditory system. [Research support provided by grants from NIH (NIDCD): R01-DC008327 and the Purdue Research Foundation.]

1aPP6. Influence of context on the relative pitch dominance of individual harmonics. Hedwig E. Gockel, Sami Alsindi, Charles Hardy, and Robert P. Carlyon (MRC-Cognition and Brain Sci. Unit, 15 Chaucer Rd., Cambridge CB2 7EF, United Kingdom, hedwig.gockel@mrc-cbu.cam.ac.uk)

There is evidence that the contribution of a given harmonic in a complex tone to residue pitch is influenced by the accuracy with which the frequency of that harmonic is encoded. We investigated whether listeners adjust the weights assigned to individual harmonics based on acquired knowledge of the reliability of the frequency estimates of those harmonics. In a two-interval forced-choice task, seven listeners indicated which of two 12-harmonic complex tones had the higher overall pitch. In context trials (60% of all trials), the fundamental frequency (F0) was 200 Hz in one interval and 200+ΔF0 Hz in the other. In different blocks, either the third or the fourth harmonic, plus (always) the seventh, ninth, and 12th harmonics were replaced by narrowband noises that were identical in the two intervals. Feedback was provided. In test trials (40% of all trials), the fundamental frequency was 200+ΔF0/2 Hz in both intervals and either the third or the fourth harmonic was shifted slightly up or down in frequency. There were no narrowband noises. Feedback was not provided. The results showed that substitution of a harmonic by noise in context trials significantly reduced the contribution of that harmonic to pitch judgments in the test trials.

1aPP7. How is periodicity pitch encoded on basilar membrane of the cochlea? Takeshi Morimoto (Sensory and Cognit. Neural System Lab., Graduate School of Life and Medical Sci., Doshisha Univ., 8-16-10, Kanou, Higashiosaka-shi 578-0901, Japan, master1020scncl@gmail.com), Kohta I. Kobayasi, and Hiroshi Riquimaroux (Sensory and Cognit. Neural System Lab., Graduate School of Life and Medical Sci., Doshisha Univ., Kyotanabe, Japan)

Pitch is the perceptual correlate of periodicity of sound. Even if all the energy at the fundamental frequency (F0) is removed, a periodic complex tone retains the same pitch of missing fundamental frequency. The periodicity pitch has been mostly discussed in the central auditory system while less reported in the peripheral system. The purpose of this study was to investigate how the periodicity pitch is encoded in the cochlea. Cochlear microphonics (CM) was recorded from the round window for confirming frequency characteristics of vibration of the basilar membrane (BM) to periodic complex tones in order to generate F0. The experiments were carried out in the condition which a frequency component corresponding to F0 or frequency components of the sound stimuli were masked by low or high frequency band pass noises respectively. The forward masking did not have temporal overlap of the sound stimuli on the BM. The results showed that the frequency component of F0 existed in the CM. The frequency component of F0 decreased for high frequency band pass masking but little for low frequency band pass masking. The findings suggest that the periodicity pitch is encoded using rates of amplitude modulation rather than the location on the BM.

1aPP8. Pitch discrimination with harmonic and inharmonic tone complexes. Lars Bramsløw and Niels H. Pontoppidan (Eriksholm Res. Ctr., Rørtangvej 20, Snekkersten 3070, Denmark, lab@eriksholm.com)

The measurement of sensitivity to temporal fine structure (TFS) in listeners can be measured by using a linear frequency-shift of a harmonic tone complex. However, the linear frequency shift of the harmonic tone complex breaks the harmonic structure, introducing a harmonic-inharmonic cue in addition to the pitch shift. In the present study, we investigated the relative contributions of frequency shift and harmonicity in normal-hearing listeners, using harmonic-harmonic, harmonic-inharmonic, and inharmonic-inharmonic shifts for unresolved tone complexes. A two-down-one-up adaptive method was used to measure the frequency shift threshold. Our results show that both inharmonic variants of the frequency shift have lower detection thresholds than the harmonic-only shift. The two inharmonic conditions are not different, indicating that the linear shift threshold is not driven by a harmonic-inharmonic cue. The effect is the same for all frequencies tested here. The results are discussed in relation to both excitation pattern and temporal fine structure models.

1aPP9. Masked speech recognition in school-age children and adults: Effects of age, masker type, and context. Lori Leibold (Allied Health Sci., The Univ. of North Carolina at Chapel Hill, 3122 Bondurant Hall, CB#7190, Chapel Hill, NC 27599, leibold@med.unc.edu), Emily Buss, and Joseph W. Hall (Otolaryngology/Head and Neck Surgery, The Univ. of North Carolina at Chapel Hill, Chapel Hill, NC)

This study evaluated the influences of age, masker type, and context on masked speech recognition. A repeated-measures design compared the speech recognition thresholds of two age groups of children (5–7 and 9–13 years) and a group of adults (19–33 years) in a continuous speech-shaped noise or a two-talker speech masker. Target stimuli were disyllabic words that were familiar to young children. Masker level was fixed at 60 dB SPL, and signal level was adapted to estimate the SNR required for 70.7% correct performance. Each listener completed testing in each masker condition using both an open-set task requiring a verbal response, and a 4AFC closed-set task requiring a picture-pointing response. Consistent with previous studies, and regardless of response context, larger and more prolonged child-adult differences were observed in the two-talker compared to the speech-shaped noise masker. Poorer performance was observed for all three age groups using the open-set compared to the closed-set context. This performance gap was similar across the three age groups in the speech-shaped noise masker, but a developmental effect was observed in the two-talker masker. Specifically, the decrement in performance using the open-set compared to the closed-set procedure increased with age in the two-talker masker.

1aPP10. Gap detection in school-age children and adults: Effects of marker center frequency and ramp duration. Heather Porter, Emily Buss, Joseph W. Hall, and John H. Grose (Otolaryngol., Univ. of North Carolina at Chapel Hill, 170 Manning Dr., Chapel Hill, NC 27599-7070, heather_porter@med.unc.edu)

Data on the development of auditory temporal processing varies widely across studies, depending on the particular stimulus (e.g., marker center frequency) and the methods used to quantify performance. Recently, gap detection thresholds were observed to be adult-like later in childhood for narrowband low-fluctuation noise (with 40-ms ramps) than wideband Gaussian noise (with 4-ms ramps; Buss *et al.*, 2013). These results could reflect relatively protracted development of the ability to detect gaps in spectrally narrow stimuli compared to broader stimuli, irrespective of inherent stimulus fluctuation. That is, the use of across-channel cues that benefit gap detection for wideband stimuli could develop early in childhood. Alternatively, age effects in gap detection may depend on center frequency or ramp duration. The present study measured gap detection for low-fluctuation narrowband noise with either a low (500 Hz) or a high (5000 Hz) center frequency, and for wideband noise with short (4 ms) or long (40 ms) ramps. Listeners were normal-hearing 4- to 16-year-old children and adults. Results will be discussed in terms of the effect of center frequency, ramp duration, and across-frequency cues on gap detection for children and adults.

1aPP11. Priming in speech perception through captions and sign language. Richard L. Freyman, Gwyneth C. Rost, Derina S. Boothroyd, Charlotte Morse-Fortier, Amanda M. Griffin, and Sarah F. Poissant (Commun. Disord., Univ. of Massachusetts, 358 N. Pleasant St., Amherst, MA 01003, rlf@comdis.umass.edu)

When listeners know the content of the message they are about to hear, distorted or partially masked speech appears dramatically more intelligible compared to when the content is unknown. In the current research this priming phenomenon was investigated quantitatively using a same-different task where the prime and auditory message match on only 50% of trials. The first experiment was concerned with the effect of the timing of typed captions relative to the auditory message. For nonsense sentences subjected to four different types of distortion and masking, optimum performance was achieved when the initiation of the text preceded the acoustic speech signal by 800 ms. Performance was slightly poorer with simultaneous delivery and much poorer when the auditory signal preceded the caption. The second experiment investigated whether priming effects could be observed when the prime was delivered by sign language rather than captions. Preliminary results with normal-hearing signers indicate that some benefits in same-different performance can be triggered by sign language primes. Confirmation of this result and extension to listeners with hearing impairment who use sign and speech together in simultaneous communication (SimCom) may inform best practices for the use of SimCom in educational and other settings. [Work supported by NIDCD 01625 and The Hearing Health Foundation.]

1aPP12. The role of bilingualism and musicianship on performance in a masked speech detection task. Charlotte Morse-Fortier, Giang Pham, and Richard L. Freyman (Commun. Disord., Univ. of Massachusetts Amherst, 1147A North Pleasant St., Amherst, MA 01002, cmorsefo@comdis.umass.edu)

Listening to an individual talker in the presence of multiple speakers is one of the most difficult auditory situations listeners encounter. Past research has shown that musicians and bilinguals have more robust neural responses to a target in a competing speech background compared to non-musicians and monolinguals, perhaps due to enhanced processing of fundamental frequency. In this study, a simple speech detection paradigm was used to minimize the effect of language comprehension while studying behavioral performance on a speech-masking task. Four groups of listeners detected English target words in the presence of two-talker babble, where all speakers were female. The task was conducted in spatial and non-spatial configurations to address conditions with low and high informational masking, respectively. The four groups tested were musicians, non-musician Asian tonal bilinguals, non-musician Spanish bilinguals, and non-musician English monolinguals. Results show that all subject groups experienced a significant release from masking in the spatial condition relative to the non-spatial condition. Musicians outperformed non-musicians in the non-spatial condition (high informational masking), but not in the spatial condition where energetic masking was presumed to dominate. Differences between the two bilingual groups and the monolingual group will be discussed. [Work supported by NIDCD 01625.]

1aPP13. The effects of amplitude envelope and context on auditory duration perception. Lorraine Chuen (Dept. of Psych., Neurosci. and Behaviour, McMaster Inst. for Music and the Mind, Psych. Bldg. (PC), Rm. 102, McMaster Univ., 1280 Main St. West, Hamilton, ON L8S 4K1, Canada, chuenll@mcmaster.ca) and Michael Schutz (School of the Arts, McMaster Inst. for Music and the Mind, Hamilton, ON, Canada)

Here, we extend research indicating that a sound's amplitude envelope (changes in intensity over time) affects subjective duration (Grassi and Darwin, 2006). Specifically, we explore whether amplitude envelope affects the strategies underlying duration perception by comparing "flat" (abrupt onset, sustain, abrupt offset) and "percussive" (abrupt onset followed by exponential decay) envelopes. Participants performed a two alternative forced choice task: judging which of two tones sounded longer. Trials were divided into blocks organized by envelope: (a1) uniform flat-flat, (a2) uniform percussive-percussive, and (b) mixed (uniform, percussive-flat, and flat-percussive). This was designed to either permit (a) or prohibit (b) envelope-specific listening strategies. Block order was counterbalanced across participants. An analysis of the first block (a between-subjects comparison of uniform and mixed block performance) indicated that performance on flat-flat trials was not significantly different for the two block types, whereas percussive-percussive trial performance was significantly worse for the mixed block. This suggests that participants optimally employ an envelope-specific strategy for the uniform block (a) when able to predict envelope, and a generalized strategy in the mixed block (b) when unable. Interestingly, there was no performance advantage for the uniform block when presented second, suggesting that contextual order effects may affect auditory duration perception.

1aPP14. Real-time implementation of a polyphonic pitch perception model. Nikhil Deshpande and Jonas Braasch (Dept. of Architecture, Rensselaer Polytechnic Inst., 220 3rd St., Troy, NY 12180, deshpn@rpi.edu)

This algorithm is a real-time implementation of a polyphonic pitch perception model previously described in Braasch *et al.* [POMA **19**, 015027 (2013)]. The model simulates the rate code for pitch detection by taking advantage of phase locking techniques. Complex input tones are processed through a filter bank, and the output of each filter is run through its own separate autocorrelation following the Licklider model. After conversion to polar form, analysis is done on the phase output of the autocorrelation, where the algorithm computes the time delay between localized peaks. This gives the fundamental period of the tone within a given filter; these values are then normalized by the magnitude output of the autocorrelation and combined with output data from the other filters to give full spectral information. The algorithm uses an adjustable running frame window to trade off between frequency resolution and rapid changes in pitch. The model can accurately extract missing or implied fundamental frequencies.

1aPP15. Infants ability to perceive changes in timbre. Bonnie K. Lau and Lynne A. Werner (Univ. of Washington, 523 Broadway East Unit 217, Seattle, WA 98102, bonniekwlau@gmail.com)

Recent studies have demonstrated that infants can ignore spectral changes in sequentially presented complex tones and categorize them on the basis of missing fundamental pitch. However, infants' ability to discriminate these spectral changes is unknown. This study investigated the ability of adults, 7- and 3-month-olds to perceive changes in the spectral centroid of harmonic complexes using an observer-based method. Stimuli were 500-ms complex tones, with 20-ms raised cosine onset/offset ramps presented at 70 dB SPL. All harmonics were generated up to 10 000 Hz then bandpass filtered with a -24 dB/octave slope around the center frequency (CF). The experiment consisted of five phases that presented complexes with a 15, 10, 5, 2, or 0.5% ΔCF. To demonstrate timbre discrimination, participants were required to ignore changes in the fundamental frequency of complexes which randomly varied between 170, 180, 190, 200, 210, and 220 Hz, and to respond only when the CF of the spectral centroid changed. Infants were tested on three conditions (15, 10, 5%, or 15, 2, 0.5%). Adult participants were tested on all phases until they reached a phase they could not discriminate. Preliminary results indicate that infant performance is comparable to adults, suggesting discrimination of timbre at 3 months.

1aPP16. Pitch shifts in scale alternated wavelet sequences and the prediction by auditory image model and spectral temporal receptive field. Minoru Tsuzaki (Faculty of Music, Kyoto City Univ. of Arts, 13-6 Kutsukake-cho, Oe, Nishikyo-ku, Kyoto 610-1197, Japan, minoru.tsuzaki@kcua.ac.jp), Toshio Irino (Faculty of Systems Eng., Wakayama Univ., Wakayama, Japan), Chihiro Takeshima (J. F. Oberlin Univ., Machida-shi, Japan), and Tohie Matsui (Tsukiba Adv. Res. Alliance, Univ. of Tsukiba, Tsukuba, Japan)

SAWS's are acoustic stimuli in which an impulse response of vocal tract and its scaled version are alternately placed in the time domain at a constant periodic rate. When the scale factor is close to unity (1.0), the perceived pitch corresponded to the original periodicity. As the difference in the scaling became large, the pitch tended to be matched to what corresponds to be lower than the original by an octave. One of the characteristics of this pitch shift was that the pitch chroma did not change. This sort of pitch continuum could not be realized by changing the fundamental frequency of harmonic complex tones, but could be realized by attenuating the odd harmonics of them. Two auditory models were used to predict this pitch shift phenomenon, i.e., AIM by Patterson's group; STRF by Shmma's group. Both models could predict the pitch shift by an octave, but AIM predicted the pitch ambiguity better than STRF. While it is easy to find the secondary local peak of periodicity besides the primary peak, the peak activity in STRF was singular in most cases. The results suggested that AIM could preserve the temporal fine structure better than STRF.

1aPP17. Discrete and continuous auditory feedback based on pitch and spatial lateralization for human-machine-interface control. Sylvain Favrot, Carolyn M. Michener, and Cara E. Stepp (Speech, Lang., and Hearing Sci., Boston Univ., 635 Commonwealth Ave., Boston, MA 02215, cmich@bu.edu)

The purpose of this study was to investigate auditory-motor learning via discrete and continuous auditory feedback using pitch and spatial lateralization. Sixteen subjects used facial surface electromyography (sEMG) to control a human-machine-interface (HMI). The output of the HMI was a lateralized harmonic tone. The fundamental frequency and lateralization (left-right ear) changed with the sum and the difference of the bilateral muscle signals. For eight participants, these changes were continuous, whereas the other eight participants received discrete feedback, in which the frequency of the tone was one of nine possible semitones (from midi note #64 to #75) and the lateralization was either left, center or right. Participants trained over three days. A mixed-models analysis of variance showed a significant effect of learning over sessions and a trend for increased performance for the group utilizing discrete feedback. Overall, information transfer rates using this purely auditory feedback averaged 38.5 bit/min by day 3, which is similar to results from similar systems utilizing visual feedback. These results show that with minimal training, auditory feedback can provide usable HMI control.

1aPP18. Speech, spatial, and qualities of hearing scale (SSQ): Normative data from young, normal-hearing listeners. Pavel Zahorik and Ann M. Rothpletz (Heuser Hearing Inst. and Div. of Communicative Disord., Dept. of Surgery, Univ. of Louisville School of Medicine, Univ. of Louisville, Louisville, KY 40292, pavel.zahorik@louisville.edu)

The Speech, Spatial, and Qualities of Hearing Scale (SSQ) was developed to assess listeners' subjective sense of listening ability and listening experience in everyday complex situations that often involve spatial hearing. The SSQ is one of the very few instruments designed to measure such quantities and has been used extensively to assess functional hearing impairment and benefit resulting from hearing remediation strategy. Although a recent study examined the psychometric properties of the SSQ with a large sample of hearing-impaired listeners, little published data exist from normal-hearing listeners. The data that have been published suggest that even young normal-hearing listeners do not rate their subjective listening abilities at the most-proficient end of the scale in all the listening situations probed by the SSQ. The goal of this study was to examine this issue more fully, using a sample of 233 young (median age 21.2 years, 3.1-year IQR), normal-hearing listeners (pure tone thresholds $\leq$ 25 dB HL from 250—8000 Hz). Results provide normative data on each of the (self-administered) SSQ

items, and describe the psychometric properties of the SSQ for a young normal-hearing population. These data are intended to aid in the interpretation of SSQ results from other populations.

1aPP19. Ear effect and gender difference of spontaneous otoacoustic emissions in children with auditory processing disorder. Kimberly Zwissler (Univ. of Delaware, 1600 Rockland Rd., Wilmington, DE 19803, zwissler@udel.edu), Kyoko Nagao, L. Ashleigh Greenwood (Ctr. for Pediatric Auditory and Speech Sci., Nemours/Alfred I. duPont Hospital for Children, Wilmington, DE), Rebecca G. Gaffney (Univ. of Delaware, Wilmington, DE), R. Matthew Cardinale (College of Osteopathic Medicine, New York Inst. of Technol., Wilmington, Delaware), and Thierry Morlet (Ctr. for Pediatric Auditory and Speech Sci., Nemours/Alfred I. duPont Hospital for Children, Wilmington, DE)

Spontaneous otoacoustic emissions (SOAEs) are found in most healthy ears and can be used to measure the health of the cochlear structures and feedback mechanism. According to existing literature, right ears tend to exhibit greater numbers of SOAEs than left ears (Bilger *et al*., 1990) and females tend to show higher incidence of SOAEs than males (Moulin *et al*., 1993). The SOAE prevalence has not been extensively studied in children with auditory processing disorder (APD), a disorder with unknown etiology that reduces one's ability to process auditory information. This study examined the prevalence and ear advantage of SOAEs between genders in children diagnosed with APD. SOAEs were investigated in 19 children (7 girls and 12 boys) with APD and 24 typically developing children (14 girls and 10 boys) aged 7–12. Right ear advantage was more prevalent in control (71%) than APD subjects (42%). However, over 30% more females exhibited a right ear advantage than males in each group. Although the results are not significant, our findings indicate that the lack of right ear advantage for SOAE is more prevalent in children with APD, particularly in males, suggesting that cochlear mechanisms or their control might be somehow affected in APD.

1aPP20. Testing a nonlinear computational channel model for masker phase effects. Yonghee Oh, Evelyn M. Hoglund, and Lawrence L. Feth (Speech and Hearing Sci., The Ohio State Univ., 110 Pressey Hall, 1070 Carmack Rd., Columbus, OH 43210, oh.172@osu.edu)

The masked threshold differences produced by Schroeder-phase maskers are most often attributed to the non-linear response of the normal cochlea (Summers *et al*., 2003). The nonlinear properties of the basilar membrane (BM) cause different response to the positive and negative Schroeder-phase maskers (i.e., + SCHR and -SCHR maskers, respectively) based on signal level, temporal synchrony, and on- and off-frequency harmonic components (Kohlrausch and Sander, 1995, Carlyon and Datta, 1997a,b, and Summers, 2000). In this study, manipulation of harmonic components of the maskers was used to explore nonlinear aspects of BM motion produced by the two different maskers. Specifically, masking period patterns (MPPs) for the + SCHR and -SCHR maskers were measured to show the influence of the phase relationships, and thus, the spectrotemporal characteristics of harmonic complexes on masking effectiveness. An enhanced channel model (Oh, 2013) provides a quantitative explanation for masking differences between + SCHR and -SCHR maskers by introducing nonlinear channel correlation across both frequency and time and a nonlinear decision criterion. [Research supported by a grant from the Office of Naval Research #N000140911017.]

1aPP21. Dual-carrier vocoder: Evidence of a primary role of temporal fine structure in streaming. Frederic Apoux, Carla L. Youngdahl, Sarah E. Yoho, and Eric W. Healy (Speech & Hearing Sci., The Ohio State Univ., 1070 Carmack Rd., Columbus, OH 43210, fred.apoux@gmail.com)

Thus far, two possible roles of temporal fine structure (TFS) have been suggested for speech recognition. A first role is to provide acoustic speech information. A second role is to assist in identifying which auditory channels are dominated by the target signal so that the output of these channels can be combined at a later stage to reconstruct the internal representation of that target. Our most recent work has been largely in contradiction with the speech-information hypothesis, as we generally observe that normal-hearing (NH) listeners do not rely on the TFS of the target speech signal to obtain speech information. However, direct evidence that NH listeners do rely on the TFS to extract the target speech signal from the background is still lacking. The present study was designed to provide such evidence. A dual-carrier vocoder was implemented to assess the role of TFS cues in streaming. To our knowledge, this is the only strategy allowing TFS cues to be provided without transmitting speech information. Results showed that NH listeners can achieve sentence recognition scores comparable to that obtain with the original TFS (i.e., unprocessed), suggesting a primary role of TFS cues in streaming. Implications for cochlear implants are discussed.

1aPP22. Modeling temporal effects in two-tone suppression. Erica L. Hegland and Elizabeth A. Strickland (Speech, Lang., and Hearing Sci., Purdue Univ., Heavilon Hall, 500 Oval Dr., West Lafayette, IN 47907, ehegland@purdue.edu)

The medial olivocochlear reflex (MOCR) reduces the gain of the active process in the cochlea. Both physiological and psychoacoustical data support the hypothesis that the MOCR may improve the response to a tone in noise. This is generally interpreted as resulting from a decrease in the response to the noise. However, elicitation of the MOCR may also reduce suppression, an instantaneous cochlear gain reduction that is a by-product of the cochlear active process. There is only limited physiological and psychoacoustic data on the effect of the MOCR on suppression. Recently, the timecourse of the MOCR has been integrated into a well-established computational auditory nerve model [Smalt *et al*., J. Assoc. Res. Otolaryngol. (2013)]. The purpose of this study is to use the auditory nerve model to systematically examine suppression and the effects of the MOCR at the level of the basilar membrane and the auditory nerve. These results will provide more understanding on the interaction of these two types of gain reduction and how they relate to speech understanding in noise. [Research supported by NIH(NIDCD) R01 DC008327 and T32 DC00030.]

1aPP23. Modeling speech intelligibility in distortion conditions using a speech-based speech transmission index. Gongqiang Yu, Anthony J. Brammer, and Eric R. Bernstein (Ergonomic Technol. Ctr., Univ. of Connecticut Health Ctr., 263 Farmington Ave., MC 2017, Farmington, CT 06030-2017, gyu@uchc.edu)

A modified objective model expanding on the speech-based speech transmission index is proposed to predict speech intelligibility under various conditions of nonlinear distortion. The proposed model computes values over a time window by analyzing the signal to noise ratio and the modulation transfer function between the input and output of a transmission channel. The channel is divided into frequency bands with ranges compatible with critical bands. The cross covariance among adjacent frequency bands is also considered. The index is obtained by averaging the calculated values of these time windows. The proposed model is evaluated with subjective measurement of word intelligibility scores using the modified rhythm test with non-linear distortions of phase jitter and clipping introduced into the speech material presented to subjects. The results demonstrate that high correlations between the indices of the proposed model and the intelligibility scores are maintained for these distortions.

1aPP24. Defining essential characteristics of reliable spectral properties that elicit spectral contrast effects in vowel identification. Paul W. Anderson and Christian Stilp (Univ. of Louisville, 2329 Mount Claire Ave., Apt. 4, Louisville, KY 40217, paul.anderson@louisville.edu)

Auditory perception excels at extracting reliable spectral properties from the listening environment. Preceding acoustic contexts filtered to emphasize a narrow frequency region (Kiefte and Kluender, 2008 *JASA*), broad differences between two vowel spectra (Watkins, 1991 *JASA*), or resynthesized to shift wide ranges of frequencies (Ladefoged and Broadbent, 1957 *JASA*) all influence identification of a subsequent vowel sound. Spectral differences between filtered contexts and subsequent vowel targets were perceptually enhanced, resulting in contrast effects (*e.g*., emphasizing spectral properties of [I] in the context produced more [ɛ] responses). Historically, this phenomenon has been studied using filters whose gain was broadband and/or high-amplitude, providing very strong evidence for these reliable spectral properties. Essential characteristics of these filters that are necessary and/or sufficient to elicit spectral contrast effects are unknown. A series of experiments examined relative contributions of filter frequency, amplitude, and bandwidth to reliable spectral properties that elicit contrast effects in vowel identification. Preceding sentence contexts were processed by narrowband, broadband, or spectral envelope difference filters derived from endpoints of a vowel series differing in one frequency region (*e.g.*, F_1 in [I] and [ε]). Preliminary results suggest complex dependencies on filter amplitude and bandwidth for vowel identification; further results will be discussed.

1aPP25. Establishing a clinical measure of spectral-ripple discrimination. Michelle R. Molis (National Ctr. for Rehabilitative Auditory Res., Portland VA Medical Ctr., 3710 SW US Veterans Hospital Rd., Portland, OR 97239, michelle.molis@va.gov), Rachael Gilbert (Dept. of Linguist, Univ. of Texas at Austin, Austin, TX), and Frederick J. Gallun (National Ctr. for Rehabilitative Auditory Res., Portland VA Medical Ctr., Portland, OR)

Spectral-ripple discrimination thresholds have been used effectively to assess frequency-resolving power in cochlear implant users. To improve potential clinical utility as a reliable and time-efficient measure of auditory bandwidth for listeners with acoustic hearing, possible confounds and limitations of the method must be addressed. This study examined frequency specificity and the possibility of edge-listening with narrowband stimuli. An adaptive 4-IFC procedure was used to determine ripple discrimination thresholds for normally hearing (NH) and hearing-impaired (HI) listeners. Stimuli were broadband (100–5000 Hz), high pass (1000–5000 Hz), or low pass (100–1000 Hz) logarithmically scaled, sinusoidally modulated Gaussian noises. In some conditions, Gaussian flanking noise was introduced to eliminate potential edge-listening cues. As expected, discrimination thresholds were significantly reduced for the HI listeners. Additionally, results indicate that both NH and HI listeners are able to use edge cues to improve discrimination thresholds. The introduction of flanking noise significantly reduced thresholds, and this effect was largest for the high pass stimuli. These results can be used to evaluate the usefulness of this method as a rapid and efficient means of assessing the effective frequency bandwidth of listeners with acoustical hearing. [Work supported by NIH.]

1aPP26. Sleeping position can reduce the effect of snoring on sleeping partners. Jen-Fang Yu, Yen-sheng Chen (Graduate Inst. of Medical Mechatronics, Chang Gung Univ., 259 Wen-Hwa 1st Rd., Kwei-Shan Tao-Yuan,-Taoyuan, Taiwan333, Taiwan, yhn888888@gmail.com), Hsueh-Yu Li (Otolaryngol., Chang Gung Memorial Hospital, Tao-yuan, Taiwan), and Li-Ang Li (Otolaryngol., Chang Gung Memorial Hospital, Taoyuan, Taiwan)

During sleep, the sleep partner is usually the first and direct victim of snoring. When snorers make mild snoring sounds, the majority of sleep partners will tolerate the sounds. In this study, the sound stimulus of snoring was the average snoring sound of ten snorers obtained using Adobe Audition. The sleeping position of sleep partners is either to the left or right of the snorers. Therefore, in this study, the measurements were performed in a semi-anechoic room, and positions of 0 degree to 180 degrees were used to represent the sound sources of snoring. This study performed measurements of the sound field of sleep partner's sleeping positions at three distances and at angles between 0 degree and 180 degree. The results demonstrate that the difference in volume received by the two ears was the largest at a separation distance of 30 cm and an angle of 90 degree. As mentioned in previous studies, this result occurred due to the head shadow effect being more pronounced at high frequency, while the snoring in this study was a complex sound and the frequencies were mostly low.

1aPP27. Frequency shifts in distortion product otoacoustic emission evoked by fast sweeping primaries in adults and newborns. Hammam A. AlMakadma, Beth A. Prieve (Commun. Sci. and Disord., Syracuse Univ., 261 Skytop Rd., Ste. 1200, Syracuse, NY 13210, haalmaka@syr.edu), Walid M. Dyab (L.C. Smith College of Eng. and Comput. Sci., Syracuse Univ., Syracuse, NY), Glenis R. Long, and Simon Henin (Speech-Language-Hearing Sci., Graduate School and Univ. Ctr., New York, NY)

Distortion product otoacoustic emissions (DPOAEs) are a vector sum of two components, generator and reflection, which produce overall DPOAE levels with a pattern of minima and maxima across frequency referred to as fine structure. The pattern of maxima and minima shifts higher or lower in frequency dependent on sweep direction (Henin *et al.*, 2011), consistent with cochlear scaling invariance. A break from scaling invariance occurs between 1 and 1.4 kHz in human adults. DPOAE phase at frequencies below the "break" are steeper in newborns than adults. We probed frequency shifts to up- and down-swept primaries of 1 octave/s in adults and newborns. Frequency shifts were examined for DPOAEs evoked by up-sweeps and down-sweeps using a covariate correlation function across the entire DPOAE frequency range, and above and below 2 kHz. Newborns had significantly greater frequency shifts in the reflection component below 2 kHz than adults. There was a significantly different fine structure frequency shift between the low- and high-frequency ranges in adults. In newborns, this difference in the frequency shifts for the low- and high- frequencies was significant for the reflection component. These preliminary findings suggest that frequency shifts can be used to assess potential maturational differences in cochlear mechanics.

1aPP28. The sharp frequency selectivity of low- and medium- spontaneous rate auditory nerve fibers might allow for rate-place coding up to 5 kilohertz. Marcos A. Cantu (Ctr. for Computational Neurosci. and Neural Technol. (CompNet), Boston Univ., 677 Beacon St., Boston, MA 02215, cantu@bu.edu)

One feature that might differentiate the three types of spontaneous rate (SR) auditory nerve fibers (ANFs) is the sharpness of frequency tuning at comparable sound pressure levels. My hypothesis, prior to modeling the tuning curves for each fiber type, was that low-SR fibers have a higher threshold for stimulation and thus have sharper frequency selectivity than medium-SR fibers, which in turn have sharper frequency selectivity than high-SR fibers. The results of the simulation support this framework. I used the Zilany *et al.* (2014) model of the auditory periphery and the cochlear tuning parameters from Shera *et al.* (2002) to generate tuning curves and neurogram raster plots for each of the three fiber types. The results of the simulation suggest that the sharp frequency selectivity of low-SR and medium-SR ANFs might allow for resolved rate-place coding up to 5 kHz. At frequencies below 1500 Hz, high-SR fibers were seen to have very different response properties than low-SR and medium-SR fibers. It is conceivable that the different fiber types constitute parallel pathways and mediate two different coding schemes. We should consider whether these sharply tuned low-SR and medium-SR ANFs, while fewer in number than High-SR ANFs, might be especially important for rate-place frequency coding.

1aPP29. Neural discrimination of degraded speech. Mark Steadman and Christian J. Sumner (MRC Inst. of Hearing Res., Sci. Rd., University Park, Nottingham NG14 5GL, United Kingdom, mark.steadman@ihr.mrc.ac.uk)

Cochlear implants provide a degraded input to the auditory system. Despite this, cochlear implant users are able to discriminate speech sounds in quiet with a degree of accuracy comparable to that of normal hearing listeners. The neural bases of this phenomenon is not well understood. A set of vowel-consonant-vowel phoneme sequences, each produced by multiple talkers, were parametrically degraded using a noise vocoder. Neural responses were recorded in the guinea pig midbrain and cortex, and auditory nerve responses were generated using a computational model. The discriminability of these responses was quantified using a nearest neighbor classifier. When envelope modulations were limited to 16 Hz, classifier performance was qualitatively similar to that of human listeners for all brain regions. However, in the auditory nerve and the midbrain, the preservation of high rate envelope cues enabled the near perfect discrimination of speech tokens even for heavily spectrally degraded speech. High rate envelope cues do not appear to increase discriminability of auditory cortex responses. High rate envelope cues, represented up to the midbrain, are useful for discriminating speech tokens. However, qualitatively more consistent with perception, high rate envelope cues do not contribute to the discriminability of cortical neural responses.

1aPP30. Pitch shift of the residue and its brainstem electrophysiological correlates are explained by nonlinear oscillation. Karl D. Lerud, Ji Chul Kim, and Edward W. Large (Psych., Univ. of Connecticut, 77 Forest Rd., Apt. B, Storrs, CT 06268, karl.lerud@uconn.edu)

Data show that amplitude modulation is an important factor in the neural representation and perceived pitch of sound. However, sounds with identical Hilbert envelopes can elicit different pitches. Sounds with consecutive harmonics elicit a pitch at the difference frequency of the harmonics. If this complex is shifted up or down, the amplitude envelope stays the same, but the perceived pitch moves in the direction of the shift; thus the fine structure contributes to its perceived pitch. Physiologically, auditory nerve spike timing data have shown that fine structure is preserved during mechanical to neural transduction, and brainstem EEG experiments in humans show a unique and consistent frequency-following response (FFR) associated with pitch-shifted stimuli. Here, we model the FFR with canonical networks of nonlinear oscillators representing the cochlea, cochlear nucleus, and lateral lemniscus. Dynamical analysis reveals a resonance at the frequency of the perceived pitch that is rooted in the fine structure-locked portion of the response. We show that these responses are natural outcomes of an intrinsically nonlinear system. Our results provide good evidence for highly nonlinear auditory brainstem processing, and suggest that these nonlinearities are essential for the perception of pitch.

1aPP31. Degraded temporal processing after traumatic brain injury. Eric Hoover, Pamela E. Souza (Northwestern Univ., 2240 Campus Dr., Evanston, IL 60201, EricHoover2014@u.northwestern.edu), and Frederick J. Gallun (National Ctr. for Rehabilitative Auditory Res., Portland, OR)

Hearing complaints are common following traumatic brain injury (TBI) even in the absence of peripheral auditory impairment. Study goals were to explore the mechanisms which underlie this complaint. Adult listeners with a history of mild TBI or concussion were compared to young and age-matched controls. All listeners had normal or near-normal audiometric thresholds. Listeners with TBI reported difficulty understanding speech in noise. We hypothesized that hearing complaints after TBI are related to damage to neural pathways resulting in degraded temporal resolution. A battery of psychophysical tasks were used to test this hypothesis, specifically: monaural temporal fine structure, to evaluate temporal coding of unresolved harmonics; binaural temporal fine structure, to evaluate temporal coding of a low-frequency sinusoid; interaural coherence, to evaluate sensitivity to the similarity of a correlated broadband noise; and masking level difference, to evaluate the ability to take advantage of interaural phase for tone detection in noise. Results showed individual differences in temporal processing ability among listeners with a history of TBI. Test scores varied from normal to significantly impaired, despite normal audiometry. These results suggest that auditory temporal processing may be related to difficulty understanding speech in noise after TBI. [Work supported by NIH.]

1aPP32. Optically generated responses in the cochlear nerve. Suguru Matsui (Sensory and Cognit. Neural System Lab., Graduate School of Life and Medical Sci., Doshisha Univ., 12-3 Oyanagi, Miyaketyo, shikigun, Nara 636-0216, Japan, dmn1018@mail4.doshisha.ac.jp), Kota I. Kobayasi (Sensory and Cognit. Neural System Lab., Faculty of Life and Medical Sci., Doshisha Univ., Kyotanabe-shi, Japan), and Hiroshi Riquimaroux (Neurosensing and Bionavigation Res. Ctr., Doshisha Univ., Kyotanabe-shi, Japan)

Optical stimulation has been shown to substitute the electrical stimulation because the optical method can be more spatially selective, less invasive, and generate no electrical artifact, comparing to electrical stimulation. Previous study researched that pulsed near infrared laser irradiation to cochlear nerves in Mongolian Gerbil evoked the compound action potentials (CAP) in the cochlear nerves. Purpose of this study is to examine the differences between optically and acoustically induced CAP. Wavelength of the laser was 1.871 μm. The neural response was evaluated by using following parameters, pulse width (10–1000 μs), radiant exposure (0–2.017 mW), and repetition rate (1–1000 Hz). Click sounds used for comparison had similar parameters with the laser, click sound duration (10–1000 μs), sound pressure level (35–80 peak equivalent dB SPL), and repetition rate (1–1000 Hz). A silver wire electrode was set on the bony rim of the round window. Optically induced neural activity was confirmed at 1–1000 Hz repetition rate and was synchronized to stimulation cycles of 1–1000 Hz. Increasing radiant exposure induced larger CAP amplitude. Laser stimulation could reproduce CAP amplitude in the range of acoustically induced one. It is concluded that near infrared laser irradiation generates similar neural activities as click sound does.

1aPP33. Effect of basilar and tectorial membrane properties and gradients on cochlear response. John Cormack, Yanju Liu, Jong-Hoon Nam, and Sheryl Gracewski (Mech. Eng., Univ. of Rochester, 217 Hopeman, Rochester, NY 14627, sheryl.gracewski@rochester.edu)

The cochlea is a spiral-shaped, fluid-filled organ in the inner ear that converts sound with high resolution over a large frequency range to neurological signals that can then be interpreted by the brain. The organ of Corti, supported below by the basilar membrane and attached above to the tectorial membrane, plays a major role in the amplification of small signals. In early fluid-structure interaction models of the cochlea, the mechanical properties of the organ of Corti were neglected and only the basilar membrane was considered, approximated by a series of springs. Recent experiments suggest that the mechanical properties and property gradients of the tectorial membrane may also be important for frequency response of the organ of Corti and that separate waves may propagate along the basilar and tectorial membranes. Therefore, a two-dimensional two-chamber finite difference model of the cochlea was developed to investigate the independent responses of the basilar and tectorial membranes. Responses are compared for models using one-, two-, or three-degree-of-freedom approximations for the organ of Corti, with parameters derived from a physiologically based finite element model. The effects of independent coupling of the fluid to the tectorial and basilar membranes and longitudinal coupling along the membranes are investigated.

Session 1aUW

Underwater Acoustics: Acoustic Tomography and Interferometry

John R. Buck, Chair
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Contributed Papers

10:00

1aUW1. Acoustic noise interferometry in the Straits of Florida at 100 m depth: A ray-based interpretation. Oleg A. Godin, Nikolay A. Zabotin, Liudmila Zabotina, Justin S. Ball (CIRES, Univ. of Colorado and NOAA Earth System Res. Lab., Physical Sci. Div., Mail Code R/PSD99, 325 Broadway, Boulder, CO 80305-3328, oleg.godin@noaa.gov), Michael G. Brown, and Neil J. Williams (RSMAS, Univ. of Miami, Miami, FL)

Cross-correlation function of ambient and shipping noise recorded simultaneously by two hydrophones provides an estimate of the acoustic Green's function, which describes deterministic sound propagation between the hydrophones and can be used to estimate physical parameters of the water column and seafloor. This paper presents results of an experimental investigation of acoustic noise interferometry in 100 m-deep water in the Straits of Florida. Acoustic noise was recorded continuously for about six days at three points near the seafloor. Coherent acoustic arrivals are successfully identified in the 20–70 Hz frequency band for pairs of hydrophones separated by ranges of 5.0 and 9.8 km. The measured noise cross-correlation functions are compared to ray-based simulations of Green's functions, with generally good agreement between correlation functions and simulations. Ray-based simulations are shown to reproduce multipath features of the measured correlation functions, which are due to multiple surface and bottom reflections. The feasibility of passive acoustic remote sensing using a few hydrophones in a shallow-water waveguide will be discussed. [Work supported by NSF, ONR, and NAVAIR.]

10:15

1aUW2. Acoustic noise interferometry in the Straits of Florida at 100 m depth: A mode-based interpretation. Michael G. Brown, Neil J. Williams, Geoffrey Banker (RSMAS, Univ. of Miami, 4600 Rickenbacker Cswy, Miami, FL 33149, mbrown@rsmas.miami.edu), Oleg A. Godin, Nikolay A. Zabotin, and Ludmilla Zabotina (CIRES, Univ. of Colorado and NOAA Earth System Res. Lab., Physical Sci. Div., Mail Code R/PSD99, 325 Broadway, Boulder, CO)

A field experiment designed to test the feasibility of noise interferometry was conducted in the Florida Straits in December 2012 in water of approximately 100 m depth. Ambient noise was recorded concurrently on three moored near-bottom instruments with horizontal separations of approximately 5 km, 10 km, and 15 km. Consistent with theoretical predictions, coherent sums (stacks) of many realizations of ambient noise at two measurement locations are shown to yield approximations to deterministic Green's functions that describe propagation between the two locations. Band-pass filtering of measured stacked cross-correlations reveal modal dispersion characteristics of the estimated Green's functions that are compared to mode-based simulations. [Work supported by NSF and ONR.]

10:30

1aUW3. Acoustic noise interferometry in the Straits of Florida at 600 m depth: Preliminary results. Michael G. Brown, Neil J. Williams, Xiaoqin Zang (RSMAS, Univ. of Miami, 4600 Rickenbacker Cswy, Miami, FL 33149, mbrown@rsmas.miami.edu), Oleg A. Godin, Nikolay A. Zabotin, and Ludmilla Zabotina (CIRES, Univ. of Colorado and NOAA Earth System Res. Lab., Physical Sci. Div., Mail Code R/PSD99, 325 Broadway, Boulder, CO)

A field experiment designed to test the feasibility of noise interferometry was conducted in the Florida Straits in September/October 2013 in water of approximately 600 m depth. Ambient noise was recorded concurrently on three moored near-bottom instruments with horizontal separations of approximately 5 km, 10 km, and 15 km. Consistent with theoretical predictions, coherent sums (stacks) of many realizations of ambient noise at two measurement locations are shown to yield approximations to deterministic Green's functions that describe propagation between the two locations. Ray- and mode-based simulations of approximations to Green's functions are compared to measured stacked cross-correlations. [Work supported by NSF and ONR.]

10:45

1aUW4. Shear-wave velocity estimation from different type of sediments. Hefeng Dong (Dept. of Electronics and Telecommunications, Norwegian Univ. of Sci. and Technol., NTNU, Trondheim NO-7491, Norway, hefeng@iet.ntnu.no)

This paper presents estimates of shear-wave velocity profiles for different types of sediments by inverting the dispersion curves of interface waves. Data used in this study were collected at different locations with different recording devices in different experiments. Four data sets were analyzed. Data from one set were recorded by hydrophone array, and data from other three sets were collected by multi-component ocean bottom seismic cables. The configuration of the experimental setup and the environmental conditions were different for the different experiments. Water depth varied from 2 m to 365 m. The frequency range of the interface-wave data was from 1.5 Hz to around 20 Hz. The phase-velocity dispersion curves of the interface waves were extracted using different time-frequency analysis methods. The maximum penetration depth of the interface waves to the sediments varied from 15m to 180 m. A Bayesian nonlinear inversion approach was used for estimating shear-wave velocity profiles as a function of depth in the sediments and the uncertainties. The estimated shear-wave velocity profiles from different experiment and different sediment type were discussed and compared.

11:00

1aUW5. Wavenumber analysis of interface wave characteristics using elastic parabolic equation solutions. Scott D. Frank (Marist College Mathematics, Marist College, 3399 North Ave., Poughkeepsie, NY 12601, scott.frank@marist.edu), Jon M. Collis (Appl. Mathematics and Statistics, Colorado School of Mines, Golden, CO), and Robert I. Odom (Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

Interface waves that travel along the ocean bottom are known as Scholte waves and can contribute to the deep ocean acoustic field, even at depths below the ray-theoretical turning point. Scholte waves spread by the inverse of the square root of range and propagate greater distances along the ocean floor than ocean acoustic energy. Elastic parabolic equation solutions are effective for analysis of Scholte wave behavior with respect to environmental parameters since these waves represent interactions between dilatational and rotational elastic waves resulting from elastic boundary conditions. Generation of interface waves by water column and buried seismic sources will be demonstrated. Hankel transforms of calculated acoustic pressure will be used to evaluate impact of elastic parameters on interface wave amplitudes and ducting effects in elastic sediment layers. The effect of large-scale bathymetry on interface wave propagation will also be investigated. [Work supported by ONR.]

11:15

1aUW6. Implementing physical constraints for noise-only mode shape estimation on real data. Ian M. Rooney, John R. Buck (Elec. and Comput. Eng., Univ. of Massachusetts at Dartmouth, 285 Old Westport Rd., North Dartmouth, MA 02747, irooney@umassd.edu), and Kathleen E. Wage (Elec. and Comput. Eng., George Mason Univ., Fairfax, VA)

Many underwater acoustic tasks sample a narrowband pressure field with a vertical line array. In the absence of strong local sources, the noise sampled by an array includes both a spatially correlated and a spatially uncorrelated component. The acoustic waveguide's modes form a basis for the spatially correlated noise component generated by distant sources. This basis can be estimated from the eigenvectors of the noise sample covariance matrix. Propagation physics constrain the mode shapes to be real, but the eigenvectors are generally complex. These complex vectors require a phase rotation for each eigenvector prior to taking the real part. Previous work found that the eigenvector mode estimates' phases include spatially correlated noise. This noise correlation degrades the performance of spatial averaging and minimum variance estimates of the unknown phase rotation. The best estimate is the phase of the array element with the maximum magnitude for each mode, which is robust to the noise's spatial correlation. This research evaluates the previously developed phase rotation techniques using a real data set. The data was collected off the north coast of Elba Island by the SACLANT Centre in 1993. [This work was supported by ONR.]

11:30

1aUW7. Remote nautical bottom estimation for the safety of navigation offshore Amazon River mouth. Qunyan Ren and Jean-Pierre Hermand (Acoust. & Environ. HydroAcoust. Lab., Labs. of Image, Signal Processing and Acoust.-Ecole polytechnique de Bruxelles / Faculté des Sci., U.L.B., ave. F.-D. Roosevelt 50, CP 165/57, Brussels 1050, Belgium, qunyren@ulb.ac.be)

In muddy areas, the fluid sediment layer can reach a level of "nautical bottom" that can contact with a ships keel causing either damage or unacceptable effects on controllability and maneuverability. Consequently, the minimum depth and allowed draught need to be determined for the safety of navigation. An acoustic remote sensing technique is proposed to facilitate navigation safety applications through the determination of sediment layer properties. It uses range and frequency-dependent features of the vertical waveguide characteristic impedance, as defined by the ratio of pressure and vertical particle velocity (or pressure gradient) at a given frequency. Such ratio can circumvent inversion uncertainty due to insufficient knowledge of complex time-varying ship noise spectrum because of its source spectral level independent. Real ship noise data recorded on a compact array offshore at the mouth of the Amazon River in Brazil, 2012, are processed by a global optimization based inversion scheme. The inverted results demonstrate that technique can estimate the effective water depth through resolving sediment layers properties, especially the density. The promising results demonstrate the feasibility of this technique to facilitate safety navigation applications at port areas.

11:45

1aUW8. Sound speed and attenuation measurements of sandy sediments in reverberation chamber. Qi Li, Qi S. Yu, and Wang Y. Huang (Acoust. Sci. and Technol. Lab., Harbin EngineeringUniv., No.45 Nantong St., Nangang District, Harbin, Heilongjiang, Harbin, Heilongjiang, China, liqi@hrbeu.edu.cn)

Sound speed and attenuation in sandy sediments are important acoustic parameters. But the uncertainties of current *in-situ* measurements at low-frequency are very large and the data are not sufficient to be used to test theory predictions. As an alternative, measurements in a water filled isolated reverberation chamber were attempted in laboratory, which preserves lower frequency limitation and smaller scale requirement of water tank. In order to know the feasibility of the method, a type of fine sand sediments was measured. Spatial averaged reflection and transmission coefficients of the sand layer and attenuation were calculated over the frequency range of 90–170 kHz. And sound speed was inverted with these measured parameters. After then measurements at low-frequency till to 2 kHz were tried in the same tank.

Session 1pAA

Architectural Acoustics: Room Acoustics Prediction and Measurement

Timothy Foulkes, Chair
Cavanaugh Tocci Assoc., 327 Boston Post Rd., Sudbury, MA 01776

Chair's Introduction—1:00

Contributed Papers

1:05

1pAA1. Audio/visual concepts for human/robot communication in immersive virtual environments. Jonas Braasch, Richard J. Radke (Ctr. for Cognition, Commun., and Culture, Rensselaer Polytechnic Inst., 110 8th, Troy, NY 12180, braasj@rpi.edu), John Wen (Ctr. for Automation Technologies and Systems, Rensselaer Polytechnic Inst., Troy, NY), Mei Si, Andrew Cunningham (Ctr. for Cognition, Commun., and Culture, Rensselaer Polytechnic Inst., Troy, NY), William Keddy-Hector, and Utkarsh Sinha (Ctr. for Automation Technologies and Systems, Rensselaer Polytechnic Inst., Troy, NY)

Communication between robots and humans is a challenge in complex built environments. In this project, we are exploring how previous achievements in this area apply to human/robot communication within immersive virtual displays. In our concrete example, we host human-scale robots that interact with humans in our Collaborative-Research Augmented Immersive Virtual Environment Laboratory (CRAIVE-Lab). The CRAIVE-Lab provides a physical/digital environment for collaborative tasks using seamless video from multiple projectors and a 128-channel wave-field system. The system is designed to monitor the whole floor-space area (10 m $\times$ 12 m) with a camera network mounted to the ceiling grid and a microphone array combining shotgun, spherical and possibly people-worn microphones. Based on the sensor data, a visual analysis and an auditory scene analysis are performed. The latter includes sound localization, speech and musical-feature recognition. The robots, a Rethink Baxter robot mounted on an electric wheelchair and a Robokind Zeno, are used to perform assistive technology and social communication tasks. Both robots have direct access to these data using a digital feed over a wireless network to augment their own sensor systems of built-in cameras and microphones. [Work supported by NSF grant #1229391.]

1:20

1pAA2. Comparison of computational predictions of broadband interior sound fields with experimental measurements. Krista A. Michalis (Structural Acoust. and Target Strength, NSWC Carderock, 9500 MacArthur Blvd., Bldg. 3 Rm. 341B, West Bethesda, MD 20817, krista.michalis@navy.mil) and Donald B. Bliss (Mech. Eng. and Mater. Sci., Duke Univ., Durham, NC)

Predictions of broadband interior noise using an energy-intensity boundary element method (EIBEM) are compared to experimental measurements. Steady-state sound fields with reflection boundaries are modeled with the EIBEM method. The specular reflection field is represented using uncorrelated broadband directional sources, expressed as spherical harmonics. For each boundary element, the amplitudes of the harmonics are determined from the incident field from all other elements and sources, and are subject to energy conservation using a Lagrange multiplier integral constraint. The computational solution utilizes a relaxation method starting with 3-D diffuse reflection. Earlier EIBEM results were compared to exact analytical solutions obtained from modal analysis, and to a broadband image method. For the experimental study, 1/3-octave band measurements were made within a full-scale steel-walled enclosure with different absorption levels, distributions, and source locations. Measurements were made along three 10-microphone linear arrays spanning the enclosure, and by microphones at other selected locations. Source power was measured independently, and random incidence absorption was inferred from both reverberation times and from steady state processing using multiple microphone locations. Predicted and measured spatial variation of the steady-state fields were in good agreement. The sensitivities of the predictions to various assumptions, geometric, anomalies, and experimental uncertainties are discussed.

1:35

1pAA3. Modeling the effects of air currents on acoustic measurements in large spaces. David H. Griesinger (Res., David Griesinger Acoust., 221 Mt Auburn St. #504, Cambridge, MA 02138, dgriesinger@verizon.net)

Historically reverberation times were accurately measured with an explosive or an interrupted tone or noise. But in theory using convolution methods for acoustic measurement requires that an impulse response be precisely time stationary. Air in large spaces is never stationary, and the further a sound wave travels—as it must in a reverberant tail—the more it will be affected by moving air. Long stimuli and synchronous averaging increase the time over which the air must be still. We propose that air currents can be modeled through the fluctuations they create in the time base of the received stimulus. We find that noise-based convolution methods such as MLS are particularly sensitive to this problem, as they must use long stimuli to achieve an adequate signal-to-noise ratio (S/N). For these methods, when there is no net flow, the principle degradation is a rapid reduction of S/N at high frequencies. If there is a net flow, the high frequency reverberation time can also be significantly shortened. Logarithmic sine stimuli are shorter than MLS for an equivalent S/N, and are less sensitive to air currents. The relevance of these effects in actual spaces is under investigation.

1:50

1pAA4. The position of the sound source in churches. Umberto Berardi (Civil and Environ. Eng. Dept., Worcester Polytechnic Inst., via Orabona 4, Bari 70125, Italy, u.berardi@poliba.it) and Francesco Martellotta (DICAR, Politecnico di Bari, Bari, Italy)

The position of the sound source is an aspect rarely considered in room acoustics because the shape of the space and its internal organization generally make clear where the sound source should be. However, this may not be the case in some buildings, such as churches. In these buildings, the source is not located in one position, but it moves from choir to the altar or the pulpit. This makes problematic the analysis of church acoustics. In particular, if the reverberation time is almost constant whenever the source may be, other acoustic parameters strongly depend on the position of the sound source. This paper focuses on the effect of the position of the sound source over the early lateral fraction and the center time. First, the results of computer simulations allow discussing the influence of some locations of the organ: above the entrance, in the middle of the room, between the altar and the worshipers, and behind the altar. Then, the values of the acoustic parameters for different sound sources are evaluated. In particular, the effect of speaking from the pulpit or from the altar are compared. Finally,

suggestions to evaluate several positions of different sound sources in churches are reported.

2:05

1pAA5. Research on the parametric design of concert hall and its acoustics. Lu W. Shuai and Xiang Yan (Acoust. Lab., School of Architecture, Tsinghua Univ., Rm. 104, Main Bldg., Beijing 100084, China, 359901298@qq.com)

Concert hall is a complex system containing different building elements and technical issues such as acoustics, sight, circulation, ventilation, etc. It takes designers a lot of efforts to modify the design or test its technical performance when using traditional design method. We explored the possibilities of applying parametric design method to concert hall design, developing an interactive parametric tool which could modify all the building elements according to designers' desires and functional needs, reducing designers' work burden; meanwhile, it could simultaneously show the acoustic attributes of the design, aiding to acquire a satisfactory technical performance. This paper introduces the basic framework of the parametric tool and also shows an example to demonstrate its validity.

2:20–2:35 Break

2:35

1pAA6. Measuring sonic presence and the ability to sharply localize sound using existing lateral fraction impulse response data. David H. Griesinger (Res., David Griesinger Acoust., 221 Mt Auburn St. #504, Cambridge, MA 02138, dgriesinger@verizon.net)

The ability to sharply localize sound sources creates a perception of presence that plays a large role in grabbing and holding attention, which is of major importance in learning and performance venues. The ability to localize accurately depends critically on how the audibility of the direct sound is reduced by reflected energy arriving in the first ~80 ms. Data analysis methods that quantify the ability to localize sound at a given seat from impulse response data are being developed—but binaural impulse response data is not common in acoustic data bases. Human hearing is not omnidirectional. It uses both ILD and ITD differences between the two ears to localize frontal sound sources with an acuity of about two degrees—far more accurately than current first-order microphones. Head shadowing also significantly lowers the strength of lateral reflections that arrive at each ear. This paper presents a method whereby existing data from omnidirectional and figure-of-eight microphones can be manipulated to emulate the ILD, ITD, and head shadowing of a binaural microphone. The method uses spherical harmonics and HRTF data to restore head shadowing and estimate the ITD and ILD of the direct sound and the first few reflections.

2:50

1pAA7. Research on the measurement and application of surface diffusivity—Takes Concert Hall at Gulangyu Music School in Xiamen as an example. Lu W. Shuai, Xiaoyan Xue, Peng Wang, and Xiang Yan (Acoust. Lab, School of Architecture, Tsinghua Univ., Rm. 104, Main Bldg., Beijing 100084, China, 359901298@qq.com)

The surface diffusivity is an important acoustical attribute of material, which is highly correlated with the sound quality of a concert hall. This paper introduces the theory and practical method to measure the surface diffusivity in laboratory, which is more precise and objective than traditional Surface Diffusivity Index (SDI), and then takes the design of the concert hall at Gulangyu Music School in Xiamen as an example, shows the diffusivity features of its six-sided-cylinder shaped GRG ceiling material, demonstrates the design method and significance of using diffusive material in room acoustic design.

3:05

1pAA8. The use of sport centers for musical shows. Amelia Trematerra (Dept. of Architecture and Industrial Design, Second Univ. of Naples, borgo san lorenzo, Aversa 83016, Italy, amelia.trematerra@unina2.it) and Gino Iannace (Dept. of Architecture and Industrial Design, Second Univ. of Naples, Aversa, Caserta, Italy)

The sport centers are often used for musical shows, these buildings are built for sporting events, and they have a large volume with surfaces acoustically reflective (concrete steps, plaster and concrete ceilings) and have high values of reverberation time. In this work, a study of sporting center "Pala Jacazzi" in Aversa (Italy) for musical shows is reported. This building was designed for sport such as basketball and volley, has a volume of 28.000 cubic meters, length is 60 m, width is 60 m, and the seating capacity is 2.000. The acoustic measurements were carried out in accordance with ISO 3382, with an omnidirectional sound source, feed with a MLS signal, put in the game rectangle, and the measurements microphone put in different point on the steps. The average value, without audience, of T30 at 1.0 kHz is over 5.0 s; this is an high value if the sporting center must be used for musical shows. The software for the architectural acoustic "Odeon" is used to choose the type of absorbent material to insert and to evaluate the area of the walls and ceiling to cover for reduce T30 values, without audience, to 3.0 s.

3:20

1pAA9. Mixing console design for telematic applications in live performance and remote recording. David J. Samson and Jonas Braasch (Rensselaer Polytechnic Inst., 1521 6th Ave., Apt. 303, Troy, NY 12180, samsod2@rpi.edu)

The purpose of this presentation is to introduce a hardware-based mixing console architecture for telematic applications that integrates key features germane to distributed performance and remote recording. The current practice in state of the art telematic performance uses simple software-based interconnection with complex routing schemes and offers minimal flexibility or control over key parameters necessary to achieve a professional workflow. In addition to all customary features, the console will have surround panning capability for both the motorized binaural manikin as well as all sources within the auralization module. Key features such as self-labeling channel strips, onboard latency monitoring, synchronized remote audio recording and monitoring, and a highly flexible routing architecture will be integrated into the console design. This console design will provide a platform for the audio engineer to realize the full potential of telematics for networked performance and remote recording.

Session 1pABa

Animal Bioacoustics and Psychological and Physiological Acoustics: Comparative Perspectives on the Cocktail Party Problem II

Mark Bee, Cochair
Dept. of Ecology and Evolutionary Biology, Univ. of Minnesota, 100 Ecology, 1987 Upper Buford Circle, St. Paul, MN 55108

Micheal L. Dent, Cochair
Psychology, Univ. at Buffalo, SUNY, B76 Park Hall, Buffalo, NY 14260

Invited Papers

1:00

1pABa1. Individual differences revealed by the challenges of listening in a complex, crowded scene. Hari M. Bharadwaj (Dept. of Biomedical Eng., Boston Univ., 19 Euston St., 1B, Brookline, MA 02446, hari@nmr.mgh.harvard.edu) and Barbara G. Shinn-Cunningham (Ctr. of Computational Neurosci. and Neural Technol., Boston Univ., Boston, MA)

To extract content and meaning from a single source of sound in a quiet background, the auditory system can use a small subset of a very redundant range of spectral and temporal features. In stark contrast, communication in a complex, crowded scene places enormous demands on the auditory system. Spectrotemporal overlap between sounds reduces modulations in the signals at the ears and causes masking, with problems exacerbated by reverberation. Often, the more sensitive neurons in the early auditory pathway are driven to saturation, yet precise sensory representation is crucial for the ability to extract spatial, pitch, and cues that support source segregation and selective attention. Consistent with this idea, many patients seeking audiological treatment seek help precisely because they notice difficulties in environments requiring auditory selective attention. Consistent with this, in the laboratory, even listeners with normal hearing thresholds exhibit vast differences in the ability to selectively attend to a target. Here, we highlight the issues faced by the auditory system in a complex scene and describe recent behavioral and electrophysiological findings that hint at the mechanisms underlying individual differences in the ability to communicate in such adverse situations.

1:20

1pABa2. Change detection in complex acoustic scenes. Maria Chait (Ear Inst., Univ. College London (UCL), 332 Gray, London WC1X 8EE, United Kingdom, m.chait@ucl.ac.uk)

The notion that sensitivity to temporal regularity (TR) plays a pivotal role in auditory scene analysis (ASA) has recently garnered considerable attention. Nevertheless, evidence supporting a primary role for TR is based on experiments employing simple stimuli consisting of one, or two, concurrent sound sequences. Whether the role of TR in mediating ASA is robust to more complex listening environments is unknown. The present study investigates sensitivity to TR in the context of a change detection task, employing complex acoustic scenes comprised of up to 14 concurrent auditory objects. Sequences of sounds produced by each object were either temporally regular (REG) or irregular (RAND). Listeners had to detect occasional changes (appearances or disappearances of an object) within these “soundscapes.” Listeners’ performance depended on the TR of both the changing object and the scene context (TR of other objects in the scene) such that RAND contexts were associated with slower response times and substantially reduced detection performance. Therefore, even in complex scenes, sensitivity to TR is critical to our ability to analyze and detect changes in a dynamic soundscape. Importantly, the data reveal that listeners are able to acquire the temporal patterning associated with at least 14 concurrently presented objects.

1:40

1pABa3. Behavioral and neuronal sensitivity concerning objective measures of auditory stream segregation. Georg Klump (Cluster of Excellence Hearing4all, School of Medicine & Health Sci., Oldenburg Univ.,Oldenburg 26111, Germany, georg.klump@uni-oldenburg.de), Lena-Vanessa Dollezal, and Naoya Itatani (Animal Physiol. & Behaviour Group, Dept. for Neurosci., Oldenburg Univ., Oldenburg, Germany)

To evaluate possible mechanisms of auditory stream segregation, it is desirable to directly compare perceptual stream segregation and its neurophysiological correlate. Objective measures of stream segregation, i.e., measures of perceptual sensitivity that differ between conditions in which one integrated stream or two segregated streams are perceived, lend themselves to the study of such phenomena. They can be applied both in studies involving human subjects and in animal studies on auditory streaming. On the one hand, we present results from a study on informational masking in Mongolian gerbils, in which better performance in an intensity discrimination task is observed if streams of target signals and distractors are processed in separate streams. On the other hand, we present results from a temporal pattern discrimination experiment in European starlings, in which better performance is achieved if the target signals and signals useful for temporal reference are processed within one stream rather than in separate streams. In both experiments we apply signal-detection theory both to the analysis of perception evaluated in behavioral experiments and to the analysis of neuronal response patterns on the midbrain and cortical level. The neuronal population response was observed to be well correlated with the behavioral sensitivity, which can shed light on the mechanisms underlying auditory stream segregation.

2:00

1pABa4. Performance-based and subjective measures of perceptual organization in humans. Andrew J. Oxenham (Psych., Univ. of Minnesota, 75 E River Parkway, Minneapolis, MN 55455, oxenham@umn.edu)

Measures of auditory perceptual organization in humans have traditionally relied on subjective reports from subjects. For instance, in alternating tone sequences, subjects might be asked to report whether they hear one or two streams. In order to investigate perceptual organization in non-human species, it can be useful to devise tests that are more objective, in that they have a right and a wrong answer and do not rely on introspection. Recent studies from our laboratory have employed various objective and subjective tasks to provide converging evidence in the search for the underlying principles of auditory perceptual organization. It is suggested that different tasks can bias potentially multi-stable percepts in one way or another, which in turn may be useful in uncovering neural correlates of perceptual organization that can vary even when the acoustic stimuli remain the same. [Work supported by NIH grant R01DC007657.]

2:20–2:40 Break

2:40

1pABa5. Cortical processes for navigating complex acoustic environments. Shihab Shamma (Univ. of Maryland, AV WIlliams Bldg., College Park, MD 20742, sas@umd.edu)

Humans and other animals can attend to one of multiple sounds and follow it selectively over time. The neural underpinnings of this perceptual feat are the object of extensive investigations. In this talk, it is argued that source segregation depends primarily on temporal coherence between responses that encode various features of a sound source. An algorithm for implementing this process will be discussed, especially the components that are inspired by adaptive auditory cortical mechanisms. The postulated necessary role of attention in this process will be addressed, and in particular how it leads to binding of all temporally coherent features of a source (e.g., timbre and location) to segregate them from the incoherent features of other sources.

Contributed Paper

3:00

1pABa6. Manatee hearing and sound localization can help navigate noisy shallow waters and cocktail events, no Lombard's needed. Edmund R. Gerstein and Laura Gerstein (Psych., Florida Atlantic Univ., 777 Glades Rd., Boca Raton, FL 33486, gerstein2@aol.com)

Simultaneous masking procedures were used to measure the hearing and underwater sound localization abilities of West Indian manatees. Auditory detection thresholds of pulsed, no-pulsed pure tones, and complex sounds were measured against white noise using forced-choice paradigms. Auditory thresholds as a function of intensity, center frequency, bandwidth, pulse rate, and spectral characteristics were measured. Resulting critical ratios for pure tone measurements demonstrated manatees have acute frequency filtering abilities compared with humans and other marine mammals. Signal repetition rate along with amplitude and frequency modulation providing temporal contrasts against aperiodic background noise and lowered detection thresholds. FM signal detection thresholds were measured at or below ambient background levels. Results with species specific calls and boat noise suggested loudness summation across distant critical bands, as well as FM and amplitude modulation reduced the masking effects observed with pure tones. Manatees are well adapted for hearing and locating high frequency sounds in noisy shallow water habitats where physical boundary and near surface phenomena impede the propagation of low frequencies. Narrow critical bands and selective perception of pulsed signals may be adaptations for detecting and locating species-specific vocalizations. High frequency harmonics in manatee calls provide essential directional cues between mothers and calves.

Session 1pABb

Animal Bioacoustics and Signal Processing in Acoustics: Dynamics of Biosonar: Time-Dependent Adaptations in Transmission, Reception, and Representation in Echolocating Animals I

Laura Kloepper, Chair
Brown Univ., 185 Meeting St., Box GL-N, Providence, RI 02906

Invited Papers

3:15

1pABb1. Hearing sensation change with loud sound warning in the false killer whale. Paul E. Nachtigall (Hawaii Inst. of Marine Biology, Univ. of Hawaii, P.O. Box 1106, Kailua, HI 96734, nachtiga@hawaii.edu) and Alexander Y. Supin (Severtsov Inst., Russian Acad. of Sci., Moscow, Russian Federation)

Work on hearing during echolocation has demonstrated that a whale was capable of changing its hearing sensitivity while it echolocated, perhaps to protect its hearing from its own intense emitted pulses. Would a whale similarly change hearing sensitivity when warned prior to receiving a loud sound? Hearing was measured using auditory evoked potentials (AEP). The whale had been trained to station within a hoop while wearing surface electrodes. Baseline AEP dependence on test-sound level and an auditory threshold were first established for a 20 kHz tone. In a second phase, the test sound was followed by a sudden increase in amplitude up to 170 dB re 1 μPa. Thus, the faint test sounds took on the role of a warning signal for the ensuing loud (unconditioned) sound. After a few trials, the test stimuli revealed a substantial reduction of hearing sensitivity before the loud sound. When the delay between the warning tone onset and loud tone was short (varied randomly from 1 to 9 s), the whale increased its hearing thresholds (reduced sensitivity) by around 13 dB. The data indicate that: (1) the whale learned to change hearing sensitivity when warned that the loud sound was about to arrive, and (2) the learning acted only when warnings were immediate.

3:35

1pABb2. Investigating the temporal dynamics of dolphin biosonar using phantom echoes and auditory evoked potentials. James J. Finneran (US Navy Marine Mammal Program, 53560 Hull St., San Diego, CA 92152, james.finneran@navy.mil), Jason Mulsow, and Dorian S. Houser (National Marine Mammal Foundation, San Diego, CA)

Phantom echo generation replaces physical targets with electronic signals that can be delayed in time, scaled in amplitude, and broadcast to an echolocating animal to simulate echoes from distant objects. Compared to physical targets, phantom echoes have the advantages of allowing for instantaneous changes in target characteristics, independent manipulation of echo delay and echo amplitude, and easy randomization of target range. At the Navy Marine Mammal Program in San Diego, California, phantom echo generation is combined with measurements of auditory evoked potentials to investigate the temporal dynamics of biosonar signal emission and reception in bottlenose dolphins. The studies are primarily focused on examining automatic gain control mechanisms by measuring changes in hearing sensitivity — assessed via the auditory steady-state response (ASSR) to an amplitude modulated tone — over time courses corresponding to single biosonar click-echo pairs. Results show the ASSR amplitude initially decreases at the time of click emission and then recovers following click emission, with the time course of recovery related to target range, click amplitude, and tone frequency. Additional studies are focused on dynamic changes in click emissions that occur with changes in target range. [Work funded by SSC Pacific Naval Innovative Science and Engineering (NISE) program.]

3:55

1pABb3. Mechanisms of distance-invariant recognition of target strength in the biosonar of odontocetes. Alexander Supin (Inst. of Ecology and Evolution, Russian Acad. of Sci., 33 Leninsky Prospect, Moscow 119071, Russian Federation, alex_supin@mail.ru) and Paul Nachtigall (Marine Mammal Res. Program, Hawaii Inst. of Marine Biology, Kailua, HI)

Invariant target recognition by sonar requires assessment of the target strength invariably of distance despite wide variation of the echo level. Information on the echo delay and the level of the emitted pulse allows computing of the target strength. The question is: which particular mechanisms perform this computation in the biosonar of odontocetes? Investigations of the auditory evoked potentials (AEPs) during echolocation in several odontocetes have shown that comparison of the emitted pulse level, echo delay, and echo level is based on the forward masking of the echo-response by the preceding self-heard emitted click. Prolongation of the echo delay results in releasing of the echo-related AEP from masking. This release from masking compensated for the echo attenuation with distance. As a result, the echo-related AEP amplitude depended on the target strength and little depended on the target distance. Moreover, the forward-masking duration depended on the level of the emitted pulse. As a result, the echo-related AEP featured amplitude little depended on the level of the emitted pulse. The constancy of the echo-related AEP amplitude indicates the dependence of sensation level of the echo only on the target strength and the independence of both emitted click level and target distance.

4:15

1pABb4. Biosonar auditory model for target shape perception and clutter rejection. James A. Simmons (Neurosci., Brown Univ., 185 Meeting St., Box GL-N, Providence, RI 02912, james_simmons@brown.edu), Michaela Warnecke (Psychol. and Brain Sci., Johns-Hopkins Univ., Baltimore, MD), and Jason E. Gaudette (NUWCDIVNPT, NAVSEA, Newport, RI)

Big brown bats (*Eptesicus fuscus*) emit widely beamed biosonar sounds that contain two prominent harmonics (FM1, FM2). They exploit the ripple spectrum of multiple-glint echoes to perceive target shape and the relative weakening of higher harmonic FM2 frequencies in lowpass echoes from the surrounding scene to suppress clutter. Delay discrimination experiments using electronically generated echoes show that (1) flat-spectrum or ripple-spectrum echoes are perceived as having sharply focused delay images, (2) lowpass filtering causes defocussing of perceived delay, and (3) masking release occurs between defocused images of lowpass echoes and focused images of full-band echoes. A time-frequency auditory model for the decomposition of FM echoes into temporal and spectral dimensions, followed by reconstitution of object images in terms of delay alone accounts for both shape perception and the release from clutter masking. The model's structure resembles a neuronal spectral pattern-recognizing network grafted onto a neuronal delay-line/coincidence-detection delay-determining network. The categorical segregation of focused target shape from defocused clutter emerges from a novel anticorrelation process that depends on the auditory system's restricted range of lateral interactions across adjacent frequencies. [Work supported by ONR, NSF, BIBS, and JSPS.]

4:35

1pABb5. Biosonar dynamics in horseshoe and old-world leaf-nosed bats. Rolf Mueller, Anupam K. Gupta (Mech. Eng., Virginia Tech, 1075 Life Sci. Cir, Blacksburg, VA 24061, rolf.mueller@vt.edu), Weikai He (Shandong Univ.-Virginia Tech Int.. Lab., Shandong Univ., Jinan, Shandong, China), and Mittu Pannala (Mech. Eng., Virginia Tech, Blacksburg, VA)

Horseshoe bats (family Rhinolophidae) have evolved a capable biosonar system to allow the pursuit of prey amid dense vegetation that produces large amounts of clutter echoes. Horseshoe bats have long been known to employ dynamic effects such as Doppler effect compensation and large-scale rotations of their pinnae to realize these capabilities. Recent research has produced evidence of even more pervasive dynamical biosonar properties in horseshoe bat biosonar as well as in the related Old World leaf-nosed bats (family Hipposideridae). On the emission side, these animals employ elaborate baffle shapes that surround the exit points of their ultrasonic pulses (nostrils). During echolocation, multiple parts of these noseleaves such as the anterior leaf and the lancet in horseshoe bats are set in motion, typically in synchrony with pulse emission, hence creating a time-variant channel for the exiting wave packets. Similarly, the pinnae which diffract the incoming echoes on the reception side are frequently in motion while echoes impinge on them. These motions include non-rigid changes in shape. All these effects add a dynamic dimension to the interface of the bats' biosonar with the external world, which could allow the animals to enhance the quantity and quality of the sensory information they receive.

4:55

1pABb6. Encoding phase information is critical for high resolution spatial imaging in biosonar. Jason E. Gaudette (Code 8511, Naval Undersea Warfare Ctr., 1176 Howell St., B1371/3, Newport, RI, jason.e.gaudette@navy.mil) and James A. Simmons (Dept. of Neurosci., Brown Univ., Providence, RI)

The auditory system is responsible for translating acoustic information into a robust neural representation. In echolocating mammals, precise timing of neural onset-responses is critical to reconstruct complex acoustic scenes. Phase is often ignored in transmit and receive beam patterns, but it holds significance when considering broadband signals. A beam pattern's phase alternates outside of the main lobe, which leads to a frequency-dependent structure that is useful for spatial localization. For imaging in azimuth, binaural spectral patterns and time delay between the ears encode angular position. Imaging in elevation relies principally on specific spectral patterns encoded by each ear. The additional phase information decorrelates broadband echoes arriving from off-axis. This decorrelation only occurs on the order of a single wave period; however, the pattern of dispersion across frequency is sufficient to defocus echoes arriving from off-axis in the peripheral region, while accepting echoes arriving from the focal area of attention. We propose that acoustic spectral pattern matching by echolocating animals includes both magnitude and phase components of beams in the form of timing the onset response. Computational modeling results are presented showing how encoding phase information leads to high-resolution images despite dynamic environments and variability in the target strength of objects.

5:15

1pABb7. Reconstructing the acoustic scenes encountered by free-flying, foraging bats. Wu-Jung Lee (Inst. for Systems Res., Univ. of Maryland, 1147 Biology/Psych. Bldg., College Park, MD 20742, wjlee@umd.edu), Sonja Sändig, Annette Denzinger, Hans-Ulrich Schnitzler (Animal Physiol., Inst. of Neurobiology, Univ. of Tbingen, Tbingen, Germany), Timothy K. Horiuchi, and Cynthia F. Moss (Inst. for Systems Res., Univ. of Maryland, College Park, MD)

Aerial insectivorous bats face the challenge of efficient echo scene analysis for localizing obstacles and capturing prey in flight. Data collected with a telemetry microphone mounted on the foraging bat's head provide a valuable opportunity to reconstruct acoustic scenes comprised of echoes returning to the bat's ears. This study explores the information embedded in echoes from a tethered insect and background clutter recorded by the telemetry microphone in laboratory experiments. Using images from high-speed video cameras and recordings from a far-field microphone array, angular information about different objects in the echoes are restored by assimilating the reconstructed bat's position and echolocation beam aim with respect to the objects along its flight trajectory toward prey capture. This procedure is further augmented by theoretical simulations using acoustic scattering principles to circumvent the limitation imposed by the sensitivity and signal-to-noise ratio of the telemetry microphone. The reconstructed acoustic scenes offer an avenue for detailed analysis of important cues for figure-ground separation in a cluttered environment, and serve as a basis for subsequent neurocomputational modeling of auditory scene analysis performed by the bat's sonar receiver.

Session 1pAO

Acoustical Oceanography and Signal Processing in Acoustics: Using Acoustics to Study Fish Distribution and Behavior II

Kelly J. Benoit-Bird, Cochair
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Timothy K. Stanton, Cochair
Woods Hole Oceanogr. Inst., MS #11, Woods Hole, MA 02543-1053

Contributed Papers

1:00

1pAO1. Concurrent inversion of bio and geoacoustic parameters from broadband transmission loss measurements in the Santa Barbara Channel. Orest Diachok and Glenn Wadsworth (Johns Hopkins Univ. APL, 11100 Johns Hopkins Rd., Laurel, MD 20723, orestdia@aol.com)

This paper describes result of an interdisciplinary experiment, BAS II, which included coincident measurements of broadband (0.3–5 kHz) transmission loss (TL), depths and length distributions of fish, geoacoustic parameters, and continuous temperature profiles. The objective: demonstrate the accuracy of bioacoustic parameters of fish inverted from TL data. TL measurements were conducted between a fixed source and a fixed 16 element, receiving array that spanned most of the water column. Trawls revealed that the dominant species were sardines and anchovies. The TL data at night exhibited absorption lines at resonance frequencies associated with 15 and 8 cm long sardines, and 11, 9.5, and 5.5 cm long anchovies at 12 m, in good agreement with coincident trawl and echo sounder measurements. TL data during the day exhibited an absorption line associated with sardine schools. Concurrent inversion of bio and geoacoustic parameters of nighttime data was based on the Genetic Algorithm. The layer of fish was characterized by selectable depth, thickness and attenuation coefficient. Inverted values of biological parameters at 1.1 kHz, the resonance frequency of 11 cm sardines, were in accord with echo sounder and trawl data.

1:15

1pAO2. Spatiotemporal variability of clutter due to fish aggregations in the vicinity of Heceta Bank. Roger C. Gauss, Joseph M. Fialkowski (Acoust. Div., Naval Res. Lab., Code 7164, 4555 Overlook Ave., S.W., Washington, DC 20375-5350, roger.gauss@nrl.navy.mil), and Richard H. Love (BayouAcoust., Abita Springs, LA)

Acoustic interactions with fish can be a significant source of clutter to mid-frequency (MF; 1–10 kHz) active sonars. To develop robust schemes for reducing sonar false alarm rates, it is thus important to accurately characterize the spatiotemporal nature of MF echoes from fish. In the summer of 2012, long-range MF measurements of coherent backscattering from fish aggregations were made at several sites in the vicinity of Heceta Bank off the coast of Oregon. The dominant fish species included Pacific sardines that typically school near the surface and Pacific hake that typically are above the bottom in loose aggregations or layers, and so represent spatially different echo classes. Measured spatiotemporal fish echo statistics are discussed in terms of the observed in-situ distribution and behavior of these two species. NRL's moment-based Poisson-Rayleigh clutter model is used to physically relate the measured amplitude distributions to scatterer attributes (spatial density and dispersion). The temporal persistence of the clutter echoes is quantified via georeferenced echoes over multiple broadband transmissions. The results suggest that in this area, while fish may present a significant source of false alarms, their impact can be reduced by exploiting their short time-scale behavior. [Work supported by the Office of Naval Research.]

1:30

1pAO3. Accounting for the non-Rayleigh echo statistics of individual elongated scatterers in an aggregation. Wu-Jung Lee (Inst. for Systems Res., Univ. of Maryland, 1147 Biology/Psych. Bldg., College Park, MD 20742, wjlee@umd.edu) and Timothy K. Stanton (Dept. of Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., Woods Hole, MA)

The statistics of echoes from active sonar systems, such as the shape of the probability density function (pdf) of echo magnitude, can be used to estimate the numerical density of scatterers in aggregations. This study investigates the importance of non-Rayleigh echo statistics of individual scatterers on the shape of the echo pdf of aggregations in which the echoes from individuals may overlap. The signals are broadband, and the geometry involves direct paths between the sonar and the scatterers without interference from the boundaries. Echo pdf models are generated by varying the number of scatterers randomly distributed in a half-space shell while accounting for the frequency-dependent system response and beampattern effects. Individual scatterers are modeled using elongated spheroidal shapes with varying distributions of lengths and angles of orientation. The non-Rayleigh echo statistics of individual scatterers were found to contribute significantly to the non-Rayleigh characteristics of the echo pdf of aggregations of those individuals. This model is applied to estimate the numerical density of fish in aggregations observed using broadband signals (30–70 kHz) in the ocean. The results show the importance of incorporating realistic parameters for modeling individual scatterers in echo statistical analyses.

1:45

1pAO4. Acoustic scattering characteristics of pelagic and coastal nekton. Joseph Warren, Kaylyn Becker (Stony Brook Univ., 239 Montauk Hwy, Southampton, NY 11968, joe.warren@stonybrook.edu), Dezhang Chu (NWFSC, NOAA, Seattle, WA), and Kelly Benoit-Bird (COAS, Oregon State Univ., Corvalis, OR)

Measurements of several acoustic scattering characteristics were made for a variety of different nekton. At-sea broadband high-frequency (100s to 1000s kHz) backscatter measurements were made on several species of myctophids and other types of pelagic nekton (e.g. fish, shrimp). Animals were caught in mid-water trawls off the coast of Oregon during the summer of 2012, and measurements were made on fresh specimens from multiple species. Both broadside (dorsal) and end-on (head/tail) measurements were recorded. There was strong variability among and within species as well as with animal orientation. We also report measurements of swim-bladder size, shape, and fullness for the pelagic myctophids. Additionally, high-resolution computerized tomography (CT) scans were made for several coastal nekton (including squid, silverside, and sea bass) species from New York. These scans provide information on the density contrast of the various organs and other structures in the animal. These data provide useful information for acoustic scattering models of these and other similar animals.

2:00

1pAO5. Modeling the acoustic color of large aggregations of fish. David Burnett (Naval Surface Warfare Ctr., Code CD10, 110 Vernon Ave., Panama City, FL 32407, david.s.burnett@navy.mil)

The Naval Surface Warfare Center Panama City Division has developed a 3-D finite-element computer simulation system, PC-ACOLOR, for modeling the "acoustic color" (target strength as a function of frequency and aspect angle) of realistic elastic objects, either singly or in aggregations, that are near the bottom of the ocean or in deep water. It employs 3-D continuum mechanics throughout the entire computational domain. All objects and fluids are treated collectively as a single heterogeneous continuum; no engineering approximations are used anywhere. PC-ACOLOR was developed originally for modeling manmade structures, but it is intrinsic to finite-element modeling that the same code can be applied, unaltered, to any elastic objects, whether they be manmade structures or biological organisms. The talk will give an overview of PC-ACOLOR: structural acoustic concepts; modeling techniques; verification and validation; and examples of different types of objects that have been modeled, including aggregations of many fish.

2:15

1pAO6. Material properties of Pacific hake, Humboldt squid, and two species of myctophids in the California Current. Kaylyn Becker and Joseph D. Warren (School of Marine and Atmospheric Sci., Stony Brook Univ., 239 Montauk Hwy, Southampton, NY 11968, kaylyn.becker@gmail.com)

We measured the material properties of Pacific hake (*Merluccius productus*), Humboldt squid (*Dosidicus gigas*), and two species of myctophids (*Symbolophorus californiensis* and *Diaphus theta*) collected from the California Current. Density contrast (g) was measured for pieces of hake and myctophid flesh, and the following Humboldt squid body parts: mantle, arms, tentacle, braincase, eyes, pen, and beak. Density contrast varied with fish species, as well as among squid body parts. Effects of animal length and environmental conditions on nekton density contrast were investigated. Sound speed contrast (h) was measured for hake and myctophid flesh, Humboldt squid mantle and braincase, and varied within and between nekton taxa. These material property measurements can be used to more accurately parameterize target strength models and increase the accuracy of nekton biomass from acoustic surveys.

2:30

1pAO7. Target phase information for acoustic target identification: Method and preliminary results. Alan Islas-Cital (R&D, Navico, 4500 S. 129th East Ave. Ste. 200, Tulsa, OK 74134-5885, alan.islas@navico.com), Rubén Picó (Departamento de Física Aplicada, Universitat Politécnica de Valéncia, Gandia, Spain), and Phil Atkins (School of Electron., Elec. and Comput. Eng., Univ. of Birmingham, Birmingham, United Kingdom)

Target phase information in acoustic backscattering has been proposed as an additional target identification parameter. In general, incorporating phase into sonar signal processing for acoustical oceanography could aid in the assessment of fish populations and ecosystems. In this work, a broadband sonar system calibrated in amplitude and phase is used to measure the response of submerged targets in a laboratory water tank. Frequency domain data processing is applied, with target phase measured as a phase angle difference between two frequency components. This approach aims to eliminate range factors, leaving only target-induced phase features. The method is developed and validated by comparing experimental results to analytical and numerical methods, in the characterization of some targets with regular geometries such as spheres, shells, and cylinders. A compensation algorithm is implemented to account for phase ambiguities and arrive to a figure of merit for template classification. Simplified scenarios are studied in order to demonstrate the potential applicability of this method.

2:45–3:00 Break

3:00

1pAO8. Calibration of a broadband acoustic system in near-field. Grant Eastland (NW Fisheries Sci. Ctr., Frank Orth & Assoc. (NOAA Affiliate), 2725 Montlake Blvd. E, F/NWC4, Seattle, WA 98112-2097, grant.eastland@noaa.gov) and Dezhang Chu (NW Fisheries Sci. Ctr., NOAA Fisheries, Seattle, WA)

This paper investigates the applicability of calibrating a broadband acoustic system in Near-field. The calibration was performed on a single transducer with a monostatic or backscattering configuration using a standard target, a 25-mm tungsten carbide sphere, in the near-field of both the transducer and the sphere. Theoretical model to quantify the near-field effect was developed in the paper. Theoretical simulations revealed that although the shape of the frequency responses of the received echoes at different distances varied significantly, the null positions were essentially invariant, a unique characteristic that was used to determine the compressional and shear wave speeds in the calibration sphere. The calibration curves obtained at different distances in the near-field by taking into account the near-field effect were consistent with each other. Since the transducer was located in the near-field, the signal-to-noise ratio was high, resulting in a much wider usable bandwidth, between 300 and 800 kHz, than the nominal bandwidth. The methods reported here could potentially be applied to the calibration of multibeam echosounder and sonar systems.

3:15

1pAO9. Model-based and *in-situ* observations of high-frequency (10s–100s kHz) acoustic scattering from multiple targets. Samuel S. Urmy and Joseph D. Warren (Marine and Atmospheric Sci., Stony Brook Univ., 239 Montauk Hwy., Southampton, NY 11968, samuel.urmy@stonybrook.edu)

The biomass of many fish and plankton stocks is estimated using active acoustics and the echo-integration method. This method relies on the assumption that the acoustic energy backscattered by schools or aggregations varies linearly with the number of scattering targets. While accurate under most circumstances, theory predicts this assumption will break down at high scatterer densities, a prediction that has been confirmed experimentally in previous studies. The number of studies exploring these effects, however, remains small. Departures from linearity may be caused by multiple scattering, shadowing effects, and/or resonant spacing of targets at the ensonifying frequency. We explored these effects on different configurations of scatterers using a combination of theory, computer modeling, and *in-situ* measurements on standard targets. The computer model in particular facilitated controlled testing on scattering configurations that would be difficult to achieve *in-situ*. We found that the linearity assumption is supported at most densities commonly encountered in fisheries surveys, but that there are some situations where nonlinear effects become important and should be considered. These results will be useful in the interpretation of echo-integration data from a variety of species and ecosystems.

3:30

1pAO10. Comparison of near-field acoustic coherent backscattering simulations with optics theory and experiments. Adaleena Mookerjee and David R. Dowling (Mech. Eng., Univ. of Michigan, 2010 W.E. Lay Automotive Lab., 1231 Beal Ave., Ann Arbor, MI 48109, adaleena@umich.edu)

Remote discrimination of fish schools from other scatterers in the water column is important for environmental awareness and monitoring, and for a variety of sonar applications. Depending on the school's geometry and number of fish, and the fish's scattering characteristics, there may be preferential backscatter of sound, a phenomenon known as coherent backscatter

enhancement (CBE). Thus, the presence and characteristics of CBE could help in discriminating fish schools from other objects. For CBE, the addition of in-phase scattered waves from the propagation path pairs yields a scattered intensity enhancement of a factor of two in the direction opposite to that of the incident wave. However, prior simulations based on the Foldy (1945) equations have suggested the enhancement may be greater than a factor of two for relatively large closely spaced scatterers. This presentation re-examines this topic and provides new CBE simulation results for near field scattering from finite-sized aggregations of point scatterers. Comparisons are made with equivalent results from the optics experiments of Wolf and Maret (1985) and the theory of Akkermans *et al*. (1986). Extension of this effort to comparisons of CBE from single frequency and broadband pulse illumination is anticipated. [Sponsored by the Office of Naval Research.]

3:45

1pAO11. Time domain investigations of acoustical scattering from schools of swim bladder fish. Maria P. Raveau (Departamento de Ingenieria Hidraulica y Ambiental, Pontificia Universidad Catolica de Chile, Vicuña Mackenna 4860, Macul, Santiago 7820436, Chile, mpraveau@uc.cl) and Christopher Feuillade (Departamento de Fisica, Pontificia Universidad Catolica de Chile, Santiago, Chile)

Recent studies of time independent scattering from schools of swim bladder fish reveal important differences between the predictions of a mathematical model, which fully incorporates multiple scattering processes between the fish, and a second approach which treats the school as an effective medium with complex sound speed determined by the swim bladder resonance function. In back scattering, both modeling and data comparisons show that the effective medium approach underestimates the scattering amplitude when the fish separation is greater than about a quarter of the incident wavelength. In contrast, comparisons in the forward scattering direction show good agreement. These results are critically significant for time domain investigations of fish school scattering, aimed at using spatial and temporal variations in the acoustic field to study the stochastic behavior of the distribution and motion of the fish ensembles. A simple approach, using the inverse FFT of the school model and effective medium harmonic solutions, again reveals the limitations of the effective medium approach in back scattering. However, to obtain high resolution time sampling, more sophisticated time domain solution techniques based upon numerical integration and perturbation theory approaches are necessary. Both computational studies and data comparisons will be presented. [Research supported by ONR.]

4:00

1pAO12. Extracting effective medium properties for fish schools from resonator and free-field measurements. Craig N. Dolder, Gregory R. Enenstein (Mech. Eng. Dept. and Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, dolder@utexas.edu), Kevin M. Lee (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), and Preston S. Wilson (Mech. Eng. Dept. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

Resonator and free-field measurements were performed with both real and model fish schools in order to determine the relationship between fish school density and sound speed and absorption of acoustic waves. The results are compared to effective medium models for sound propagation through fish and encapsulated bubbles. The current species under study is *Danio rerio* (zebrafish); however, this technique can be extended to other fish species. [Work supported by ONR.]

4:15

1pAO13. Exploration of a Bloch wave expansion technique for analyzing backscattering from large fish schools. Jason A. Kulpe, Michael J. Leamy, and Karim G. Sabra (Georgia Inst. of Technol., 771 Ferst Dr., Rm. 136, Atlanta, GA 30332, jkulpe@gatech.edu)

Scattering from large fish schools at typical SONAR frequencies (1–10 kHz) can significantly contribute to volume reverberation. An efficient modeling technique, which accounts for fish' spatial configuration and inter-fish multiple scattering effects, is sought to quantify the acoustic scattering from a large fish school. Recent work has exploited the near-periodic nature of the schooling fish to represent the school as an equivalent phononic crystal (PC) composed of air-filled swim bladders periodically arrayed in a water matrix. Using the Bloch theorem applied to infinite media discretized by finite elements, the approach is capable of quickly and accurately generating the band structure and reflection/transmission coefficients for an incident plane wave. In this work, we extend the analysis approach to finite-sized fish schools to quantify the backscattering as a function of incident wave frequency, school geometry, and weak internal disorder. Scattered field predictions are compared against a self-consistent scattering method as well as full-field finite element simulations. For the presented approach, we note fast and scalable computation (with respect to frequency and school size) with very good agreement in predicted scattered pressure fields and the frequencies corresponding to peak target strengths. This work shows promise for predictive SONAR modeling.

4:30–4:55 Panel Discussion

Session 1pBA

Biomedical Acoustics: Breast Ultrasound II

Koen W. A. van Dongen, Cochair
Lab. of Acoust. Wavefield Imaging, Faculty of Appl. Sci., Delft Univ. of Technol., P.O. Box 5046, Delft 2600 GA, Netherlands

Timothy E. Doyle, Cochair
Physics, Utah Valley Univ., MS 179, 800 W. University Parkway, Orem, UT 84058-5999

Invited Papers

1:00

1pBA1. Optimization of the aperture and transducers of a three-dimensional ultrasound computer tomography system. Hartmut Gemmeke, Michael Zapf, Torsten Hopp, and Nicole V. Ruiter (Inst. for Data Processing and Electronics, Karlsruhe Inst. of Technol., Postfach 3640, Karlsruhe 76021, Germany, hartmut.gemmeke@kit.edu)

In previous work we optimized the aperture of our 3D Ultrasound Computer Tomography (USCT) system with emphasis on reflection tomography. Based on the promising clinical results with speed of sound and attenuation images, the next generation aperture will be upgraded to contain also optimization for transmission tomography. The main changes to be implemented in aperture, transducers, and transducer arrays are: (1) Overall 3D aperture: due to the shape of the buoyant breast a simpler hemispherical aperture can be applied. (2) The diameter of the aperture will be increased and the diameter of the transducers will be decreased for further homogenization of the illumination. (3) The disjunctive sampling of the transducers will be increased and the transducers will be distributed randomly to enhance the uniformity of transmission tomography. (4) Transducers will be connected both as emitters and receivers to decrease the need for mechanical movement of the aperture.

1:20

1pBA2. A practical, robust approach to high resolution ultrasonic breast tomography. Peter Huthwaite (Mech. Eng., Imperial College London, City and Guilds Bldg., Exhibition Rd., London SW7 4JU, United Kingdom, p.huthwaite@imperial.ac.uk) and Francesco Simonetti (School of Aerosp. Systems, Univ. of Cincinnati, Cincinnati, OH)

Breast ultrasound tomography is considered a potentially safer, more reliable, and more sensitive alternative to the widely used mammography for breast cancer diagnosis and screening. Vital to achieving this potential is the development of imaging algorithms to unravel the complex anatomy of the breast. Bent Ray Tomography (BRT) is the most prominent algorithm, producing sound-speed maps, but the underlying approximation of ray theory means that the algorithm is unsuitable for structures where significant diffraction is present. Accordingly, the maximum resolution of the BRT algorithm is not sufficient to image the details of the breast on the scale of a few millimeters. Therefore, iterative full-wave inversion techniques are often applied to improve the resolution of the BRT image, but they are typically slow or fail because of the uncertainties such as 3D effects, noise, or transducer characteristics. Presented here is a solution where the BRT algorithm is combined with diffraction tomography (DT), avoiding iterations yet producing a high resolution sound-speed image. It is demonstrated with both numerical and experimental data how this can successfully—and robustly—deal with a range of phenomena present in breast ultrasound experiments such as attenuation, density, 3D effects, and transducer directivity while maintaining a high resolution.

1:40

1pBA3. Using higher-order scattering in seismic imaging. Dirk J. Verschuur (Faculty of Appl. Sci., Delft Univ. of Technol., Lorentzweg 1, Delft 2628 CJ, Netherlands, d.j.verschuur@tudelft.nl)

For seismic exploration, acoustic sources and receivers are positioned at the earth's surface in order to measure the reflection response from the subsurface inhomogeneities. However, in most current imaging algorithms, only the primary reflections are being taken into account and multiple reflections are being discarded as noise. In the recently proposed method of full wavefield imaging, the higher-order scattering is taken into account in the imaging process where they contribute to extend the illumination area and no longer produce spurious imaging artifacts. This method involves an inversion process, where a recursive modeling method is used to predict the measured reflection response—including all its higher-order scattering effects—based on estimated reflectivity values and background velocity model. Because the background velocity cannot be assumed to be homogeneous in the earth, background velocity model estimation is a crucial component of the imaging process. It appears that automatic estimation of velocity models can be accomplished within the concept of full wavefield migration, in which higher-order scattering events can also be fully accommodated. Finally, the results of full wavefield imaging can be combined with localized, target-oriented full waveform inversion in order to get the final details, being the elastic properties, at the target zone.

2:00

1pBA4. On the separate recovery of spatial fluctuations in compressibility and mass density in pulse-echo ultrasound imaging using linear inverse scattering. Martin F. Schiffner and Georg Schmitz (Medical Eng., Ruhr-Univ. Bochum, Universitätsstr. 150, Bochum 44801, Germany, martin.schiffner@rub.de)

In pulse-echo ultrasound imaging (PEUI) of soft tissues, the scattered sound field is governed by spatial fluctuations of the two mechanical parameters compressibility and mass density. Spatial fluctuations in compressibility act as isotropic monopole radiators while spatial fluctuations in mass density act as anisotropic dipole radiators. Conventional strategies for linear image reconstruction in PEUI, e.g., delay-and-sum, minimum variance, and synthetic aperture focusing, exclusively account for monopole scattering. This neglect of the inhomogeneous mass density might be accompanied by a loss of diagnostically relevant information, e.g., the detection of tissue abnormalities. In this study, we formulate a linear inverse scattering problem to recover separate, space-resolved maps of the spatial fluctuations in both mechanical parameters from measurements of the scattered acoustic pressure. The physical model accounts for frequency-dependent absorption and dispersion in accordance with the time causal model. The computational costs are effectively reduced by the usage of the fast multipole algorithm. The concept is evaluated using simulated and experimentally obtained radio frequency data.

2:20

1pBA5. A closer look at contrast source inversion for breast cancer detection. Koen W. A. van Dongen and Neslihan Ozmen-Eryilmaz (Lab. of Acoust. Wavefield Imaging, Faculty of Appl. Sci., Delft Univ. of Technol., P.O. Box 5046, Delft 2600 GA, Netherlands, k.w.a.vandongen@tudelft.nl)

Ultrasound is an emerging technology for breast cancer detection. It is an efficient and harmless method that can detect tumors in dense breasts, which may be missed using mammography. Currently, several fully automated ultrasound screening modalities are being developed. For some of those systems, accurate knowledge about the transducer location is available, making the measured data suitable for imaging using non-linear inversion methods. A promising, but costly, inversion method is contrast source inversion, which has been tested successfully on synthetic measurement data. To reduce the computational costs (computing time and memory load) various setups were tested. Results obtained with synthetic data show that inversion using a single frequency component only, still yielded excellent imaging results, while significantly reducing memory requirements. In addition, our data indicate that the number of source positions is less important than the number of receiver positions. Thus, while keeping the total number of A-scans identical, the inversion improved when the number of source positions was reduced and the number of receiver positions was increased. This approach efficiently reduced the computational costs associated with the inversion.

2:40–3:00 Break

Contributed Papers

3:00

1pBA6. Reproducibility of high-frequency ultrasonic signals in breast cancer detection. A. Mackay Breivik, Andrew J. Marshall (Biology, Utah Valley Univ., 800 W. University Parkway, Orem, UT 84058-5999, mackay.breivik@gmail.com), and Timothy E. Doyle (Phys., Utah Valley Univ., Orem, UT)

The central question of this project was to determine the reproducibility of high-frequency (HF) ultrasonic signals in breast cancer detection. Previous studies on surgical specimens of breast tissue have shown that HF ultrasound (20–80 MHz) appears sensitive to a range of breast pathologies. A measurement in the ultrasonic signal called the peak density appears most sensitive to the pathology of the breast. The reproducibility of this parameter has not been quantitatively measured in a comprehensive manner. In parallel to a clinical study being conducted at the Huntsman Cancer Institute, the reproducibility of peak density measurements was studied using chicken and bovine tissue. Tissue was cut into $4 \times 3 \times 0.5$ cm and $4 \times 3 \times 1.5$ cm cubes and tested at 23.4°C. Waveforms were obtained for two types of measurements: (1) where the transducer stayed in contact with the tissue, and (2) where the transducer was lifted from the tissue between measurements. Spectral peak densities were obtained from 640 measurements. Type 1 measurements showed high reproducibility. Type 2 measurements displayed greater variability but were consistent with previous measurements on lumpectomy tissue specimens. The variability of the type 2 measurements is believed to be due to transducer-induced pressure differences between each measurement and is currently being studied.

3:15

1pBA7. Utah Valley University/Huntsman Cancer Institute Collaborative Breast Cancer Study: High-frequency ultrasound for margin assessments. J. Andrew Chappell, Janeese E. Stiles (Biology, Utah Valley Univ., Orem, UT), Leigh A. Neumayer (Surgery, Univ. of Utah, Salt Lake City, UT), Rachel E. Factor (Pathol., Univ. of Utah, Salt Lake City, UT), and Timothy E. Doyle (Phys., Utah Valley Univ., MS 179, 800 W. University Parkway, Orem, UT 84058-5999, Timothy.Doyle@uvu.edu)

In a joint effort between Utah Valley University and the Huntsman Cancer Institute, high-frequency (HF) ultrasound (20–80 MHz) is being studied to determine the pathology of surgical margins from breast conservation surgery. Results from a 2010 NIH R21 study indicated that multiple parameters in the HF ultrasonic spectrum correlate to a range of breast tissue pathologies. This technology promises to provide rapid, intraoperative evaluation of surgical margins, thereby decreasing the number of additional surgeries for patients. A blind study is currently being conducted with conventional pathology as the gold standard for assessing the accuracy of the method. Specimens are delivered by the surgeon's team immediately following resection and ultrasonically tested outside the surgical suite. The margins are approximately $3 \times 20 \times 20$ mm and are oriented using a small staple inserted by the surgeon in one corner and a stitch on one side. The margin is tested at 2–5 locations and then sent to pathology for analysis. Pathology and HF ultrasound results will be compared for correlation at the end of the study, which is expected to last one year. The study will include approximately 80 patients, 360 tissue samples, and 1400 tested locations. If successful, the method will move into clinical trial.

3:30

1pBA8. High-frequency ultrasound study of tissue margins from breast conservation surgery: Preliminary results. Teresa L. Wilson, Amy A. Fairbrother (Biology, Utah Valley Univ., 800 W. University Parkway, Orem, UT 84058-5999, tlwilson59@gmail.com), Monica Cervantes, and Timothy E. Doyle (Phys., Utah Valley Univ., Orem, UT)

A critical issue in breast conservation surgery (lumpectomy) for breast cancer treatment is ensuring the tissue surrounding the excised tumor, the margins, are cancer-free. In collaboration with the Huntsman Cancer Institute at the University of Utah, researchers from Utah Valley University are using high-frequency (HF) ultrasound to test the pathology of lumpectomy surgical margins. This preclinical study is a blind study which will involve 80 patients, approximately 320 specimens, and use traditional pathology as the "gold standard" for measuring the accuracy of the HF ultrasound method. Ultrasonic waveforms of margins were acquired at the Huntsman Cancer Hospital in pitch-catch and pulse-echo modes using 50-MHz transducers with 6.35 mm-diameter active elements. The data were analyzed to obtain ultrasonic parameters such as ultrasonic wavespeed, attenuation, and spectral peak density (the number of peaks and valleys in a HF ultrasonic spectral band). Preliminary results indicate variations in peak density between margin specimens and individual locations on specimens that are indicative of malignant and atypical breast pathologies. The objective of this paper is to search for trends in the data acquired to date to provide an assessment of reliability, stability, and robustness of the study.

3:45

1pBA9. Ultrasonic phenotyping of breast cancer cells for molecular subtyping. Laurel A. Thompson (Chemistry, Utah Valley Univ., 800 W. University Parkway, Orem, UT 84058-5999, laurelathompson@gmail.com), Janeese E. Stiles, J. Andrew Chappell, Ashley N. Calder, Caitlin Carter, Janice E. Sugiyama (Biology, Utah Valley Univ., Orem, UT), and Timothy E. Doyle (Phys., Utah Valley Univ., Orem, UT)

Breast cancer can be divided into molecular subtypes which are defined by their genetic and protein expression profiles. Current methods aimed at testing these biochemical signatures are effective classifiers but are not easily transferable to real-time clinical applications. The rapid, cost-effective determination of molecular subtype by other means would be a significant advancement in cancer detection and treatment. Our studies suggest that high-frequency ultrasound (10–100 MHz) may be sensitive to variations among breast cancer subtypes through their cytoskeletal structures, which have distinct biomechanical signatures. To further test this hypothesis, four breast cancer cell lines of different subtypes were cultured and ultrasonically tested. Direct pulse-echo measurements were collected from the cell layers using a 50-MHz transducer immersed in the growth media of the culture plates. Cell reflections in the waveforms were isolated and spectrally analyzed using computationally modeled spectra and principal component analysis (PCA). Cell phenotypes were profiled by using heat maps to display the relative distances between the PCA scores of the experimental and model spectra. The results indicate the phenotype and thus molecular subtype of cancer cells could potentially be determined by comparing their measured spectra to model spectra using a feature classification program such as PCA.

4:00

1pBA10. High-frequency ultrasonic measurement of vascularization in phantoms and Avastin-treated mice with breast tumors. Andrea N. Quiroz, Michaelle A. Cadet (Biology, Utah Valley Univ., 800 W. University Parkway, Orem, UT 84058-5999, aquiroz1912@gmail.com), and Timothy E. Doyle (Phys., Utah Valley Univ., Orem, UT)

Tissue vascularization is an important aspect of tissue engineering and oncology. The objective of this study was to determine if direct ultrasonic measurements in the 10–100 MHz range could be used as an *in vivo* vascularization assay. To simulate vascularization of tissue, phantoms were fabricated from agarose gel inclusions embedded in a gelatin-soluble fiber mixture. Ultrasonic tests were performed using two broadband ultrasonic transducers centered at 50 MHz. Results showed the samples with multiple agarose vasculature structures decreased the ultrasound wavespeed. As the level of vasculature decreased, the wavespeed of the ultrasound increased. Further investigation of vascularization included the *in vivo* evaluation of grafted breast cancer tumors in mice. The experimental group was composed of mice treated with Avastin, an angiogenesis inhibitor. The heterogeneity of the vasculature in the control tissue resulted in the scattering of the ultrasound, decreasing the wavespeed. Because the treated group contained less vascularized, more homogeneous tissue, the wavespeed was significantly higher. Results from both the phantom and mouse tumor studies revealed that the ultrasound wavespeed was inversely proportional to the level of vasculature. The results indicate that direct ultrasound wavespeed measurements in the 10–100 MHz range can be used to identify vascularization in tissue.

Session 1pID

Interdisciplinary Student Council: Introduction to Technical Committee Research and Activities: Especially for Students and First-Time Meeting Attendees

Whitney L. Coyle, Cochair
The Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802

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

Chair's Introduction—2:00

Invited Papers

2:05

1pID1. An Introduction to the Acoustical Oceanography Technical Committee. Andone C. Lavery (Appl. Ocean Phys. and Eng., Woods Hole Oceanogr. Inst., 98 Water St., MS 11, Bigelow 211, Woods Hole, MA 02536, alavery@whoi.edu)

The Acoustical Oceanography (AO) technical committee focuses on the development and use of acoustical techniques to understand the physical, biological, geological, and chemical parameters and processes that occur in the ocean interior and its boundaries. Acoustical techniques are uniquely suited for investigating underwater environments due to the low attenuation of sound relative to other commonly used remote sensing techniques based on electromagnetic radiation. The research encompassed by the AO technical committee is highly inter- and multi-disciplinary, with strong overlap with research addressed in the Animal Bioacoustics, Signal Processing, and Underwater Acoustics technical committees. In this presentation, an overview of recent "hot topics" in Acoustical Oceanography will be given, in addition to embarking on a discussion of the role of the AO technical committee in fostering education and leading the way in determining new and innovative directions in the specialization area.

2:15

1pID2. Highlights of the Underwater Acoustics Technical Committee at the 167th Meeting of the Acoustical Society of America. Marcia Isakson (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78713, misakson@arlut.utexas.edu)

The underwater acoustics (UW) technical committee (TC) is a dynamic group of investigators researching such varied fields as acoustic tomography, underwater acoustic communications, and shallow water waveguide propagation. In this overview of the UW TC, a brief introduction of the goals and interests of the community will be followed with a series of highlights of what one can expect at the UW sessions.

2:25

1pID3. An introduction to the Physical Acoustics Technical Committee activities. Steven L. Garrett (Grad. Prog. in Acoust., Penn State, Appl. Res. Lab., P. O. Box 30, State College, PA 16804-0030, sxg185@psu.edu)

The primary activity of any ASA Technical Committee is to use collective wisdom of the Committee's membership to determine which research topics within its specialization area are most active and interesting. Based on that assessment, the Committee organizes special sessions at future meetings that will bring together experts from those areas, not necessarily limited to the Society members, who can share interesting results and provide guidance regarding the directions that will lead to further understanding. In Physical Acoustics, that is a particularly daunting challenge given the scope of topics that fall within its purview: use of sound to probe material properties, sound propagation and attenuation mechanisms on this planet and in other parts of the universe, and physical effects of sound and its interaction with other forms of radiation, all of which could also go well beyond the limitations of a linear acoustical theory. Needless to say, involvement in debates about "what's hot" is both interesting and educational. Other activities include proposals for Technical Initiatives that allocate ASA resources. Recently, PATC received funding to co-sponsor the Physical Acoustics Summer School.

2:35

1pID4. Introduction to the Structural Acoustics and Vibration Technical Committee. James E. Phillips (Wilson, Ihrig & Assoc., Inc., 6001 Shellmound St., Ste. 400, Emeryville, CA 94608, jphillips@wiai.com)

The Structural Acoustics & Vibration Technical Committee (SAVTC) includes the study of motions and interactions of mechanical systems with their environments and the methods of their measurement, analysis, and control. This talk will provide a broad overview of the many research areas of interest to SAVTC. A few topics will be explored in more depth to provide background on some of the more common analysis methods used by members of the technical committee.

2:45

1pID5. Noise and its impact on our world. Erica Ryherd (Woodruff School of Mech. Eng., Georgia Inst. of Technol., Atlanta, GA 30332-0405, erica.ryherd@me.gatech.edu)

Noise invades all aspects of our lives. The word noise is actually derived from the Latin word "nausea," with one possible connection being that unpleasant sounds were made by seasick passengers or sailors in ancient times. In modern times, the demand for noise research and consulting has intensified in concert with rising population densities, growing industrialized societies, escalating demands from consumers, and increasingly common standards and legislation related to noise. The Acoustical Society of America Technical Committee on Noise (TC Noise) is concerned with all aspects of noise, ranging from noise generation and propagation, to active and passive methods of controlling noise, to the effects of noise on humans and animals. This talk will explore the broad topic of noise and its impact on our world.

2:55

1pID6. Architectural Acoustics-Space for sound, and you. Alexander U. Case (Sound Recording Technol., Univ. of Massachusetts Lowell, 35 Wilder St., Ste. 3, Lowell, MA 01854, alex@fermata.biz)

The discipline of Architectural Acoustics consistently produces more than 100 papers across six or more special sessions, at each meeting of the ASA. Student paper awards, student design competitions, and Knudsen lectures augment these activities. Joint sessions, particularly with Noise, Musical Acoustics, Psychological and Physiological Acoustics, and Signal Processing in Acoustics, add more still to the architectural acoustics goings-on at every ASA conference. The sphere of influence is not limited to ASA alone, as TCAA members participate in the Green Construction Code of the International Code Council, Society of Motion Picture and Television Engineers Study Group: Movie Theater Sound System Measurement and Adjustment Techniques, Classroom Acoustics Standards, the American Institute of Architects Continuing Education System, and more. This busy committee also produces a steady stream of publications documenting recent work and deciphering standards for key stakeholders. Anyone with an interest in the field will find many opportunities to advance their own expertise, build a network of colleagues, friends, and mentors, and contribute to the essential activities of the Technical Committee on Architectural Acoustics.

3:05–3:20 Break

3:20

1pID7. Overview of Signal Processing in Acoustics. Richard L. Culver (Appl. Res. Lab., Penn State Univ., Po Box 30, 16804, State College, PA 16801, r.lee.culver@gmail.com)

The Signal Processing Technical Committee (SPTC) of the ASA provides a forum for discussion of signal processing techniques that transcend one acoustic application. Signal processing research typically presented at ASA meetings includes techniques that show promise in one application—say underwater acoustics—but may also have application to other areas, for example, speech processing or room acoustics. There are several good reasons to get involved in the SP TC. First, since signal processing is an important aspect of many acoustic research areas, you will have the opportunity to better understand new and potentially useful tools. Second, Signal Processing is a small technical committee and you can make an immediate contribution. This talk provides an overview of some of the current topics in Signal Processing.

3:30

1pID8. The Engineering Acoustics Technical Committee—Where practical applications begin. Stephen C. Thompson (Graduate Program in Acoust., The Penn State Univ., N-249 Millennium Sci. Complex, University Park, PA 16803, sct12@psu.edu)

Engineering Acoustics encompasses the theory and practice of creating tools for investigating acoustical phenomena and applying knowledge of acoustics to practical utility. This includes the design and modeling of acoustical and vibrational transducers, arrays, and transduction systems in all media and frequency ranges; instrumentation, metrology, and calibration; measurement and computational techniques as they relate to acoustical phenomena and their utility; and the engineering of materials and devices.

3:40

1pID9. An Introduction to the Musical Acoustics Technical Committee. Paul A. Wheeler (Elec. and Comput. Eng., Utah State Univ., 1595 N 1600 E, Logan, UT 84341, paul.wheeler@usu.edu) and Andrew C. Morrison (Joliet Junior College, Joliet, IL)

The technical committee on musical acoustics (TCMU) is focused on the discovery of novel advancements in the science and technology of music and musical instruments. Many of our members pursue topics related to the physics of musical sound production, music perception and cognition, and the analysis and synthesis of musical sounds and composition. The TCMU draws from many fields represented in the society. Our sessions have presentations made by scientists, engineers, musicians, instrument builders, psychologists, architects, and others interested in the study of the science of music. An overview of selected research topics and activities of the TCMU will be presented.

3:50

1pID10. Overview of Speech Communication research. Rajka Smiljanic (Linguist, Univ. of Texas at Austin, Calhoun Hall 407, 1 University Station B5100, Austin, TX 78712-0198, rajka@mail.utexas.edu)

Even though speech communication is a fundamental part of our daily behavior, its mechanisms are not yet well understood. Speech communication research examines how spoken language is produced, transmitted, and perceived. Speech communication research involves a number of different disciplines, from linguistics and experimental psychology to speech and hearing sciences, and electrical engineering. The field covers a wide range of physiological, psychological, acoustic, and linguistic phenomena. In this talk, I will focus on research examining variation in speech intelligibility, the degree to which spoken language can be comprehended. Even in ideal communicative settings, in quiet environments between normal-hearing, native speakers of a language, speech intelligibility is variable. The variation increases in adverse communication situations that can arise from degradation related to talker (e.g., when second language learners produce non-canonical signal), signal (e.g., when target speech is masked by competing speech), or listener (e.g., when listeners use cochlear implants) characteristics (Mattys *et al*., 2012). Examining speech intelligibility variation from all these perspectives provides insights into the perceptual, physiological, linguistic, cognitive, and neurophysiological mechanisms underlying speech processing. This line of inquiry also has implications for improving speech processing when communicative conditions are compromised.

4:00

1pID11. Psychological and Physiological Acoustics: Investigating the auditory system and its responses to sound. Frederick J. Gallun (National Ctr. for Rehabilitative Auditory Res., Portland VA Medical Ctr., 3710 SW US Veterans Hospital Rd., Portland, OR 97239, Frederick.Gallun@va.gov)

Psychological and physiological acoustics concerns the investigation of the auditory system and its responses to sound in humans and other species. This encompasses perception and perceptual organization of simple and complex sounds, including speech; anatomy and function of the auditory pathways, including all physical and biological responses to auditory stimulation; hearing disorders, hearing loss, and auditory prostheses; vibrotactile and vestibular sensation, and the interaction of hearing with other sensory modalities; developmental, aging, learning, and plasticity effects in auditory function; and theories and models of auditory processes. This talk describes several current areas of research, including the benefits of having two ears, how intense noise can damage the auditory nerve, and how computational models of the auditory system can complement behavioral and physiological experiments to broaden our understanding of how the auditory system responds to sound.

4:10

1pID12. Biomedical Acoustics: Making a quantum leap in medicine. Tyrone M. Porter (Mech. Eng., Boston Univ., 110 Cummington St., Boston, MA 02215, tmp@bu.edu)

At the dawn of a new era in medicine, the future of biomedical acoustics is bright. In the last century, we witnessed the introduction of ultrasound contrast agents, lithotripsy, and the visualization of ultrasound images in three dimensions. Currently, scientists are developing acoustic-based techniques for opening the blood–brain barrier transiently in order to treat brain tumors and neurological diseases. Additionally, researchers are developing echogenic liposomes and microbubbles for targeted ultrasound image enhancement and drug and gene delivery. Further excitement has been generated by the advances made with focused ultrasound to ablate or mechanically erode solid tumors in a noninvasive and site-specific manner. As technology and protocols continue to evolve, biomedical acoustics will have a dramatic impact on the diagnosis and treatment of debilitating diseases, thus improving patient care and quality of life.

4:20

1pID13. Animal Bioacoustics: Sounds and soundscapes. Andrea Simmons (Cognit., Linguistic & Psychol. Sci., Brown Univ., Box 1821, Providence, RI 02912, Andrea_Simmons@brown.edu)

There is considerable biological diversity in the mechanisms and strategies animals use to produce and to perceive acoustic cues and to communicate and navigate within complex soundscapes. Researchers in animal bioacoustics use a variety of experimental techniques, from passive tracking to active training, to understand and model this diversity. I will highlight recent experimental work from a few model species to show how knowledge of environmental acoustics enhances our appreciation of animal evolution and cognition.

Session 1pMU

Musical Acoustics: Topics in Musical Acoustics

James P. Cottingham, Chair
Physics, Coe College, 1220 First Ave., Cedar Rapids, IA 52402

Contributed Papers

1:30

1pMU1. Turning music into sound: Vincenzo Galilei's contributions to the history of acoustics. Marina Baldissera Pacchetti (History and Philosophy of Sci., Univ. of Pittsburgh, 333 s Pacific Ave., Apt. 3, Pittsburgh, PA 15224, mab360@pitt.edu)

I investigate the contributions of Vincenzo Galilei (1520—1591), father of Galileo Galilei (1564—1642) to the development of acoustic science, with an emphasis on the role of phenomenology of sound and mathematical explanation of consonances. Sixteenth century music theory mainly aimed at recovering standards of ancient Greek music theory, transmitted by the works of Boethius, and later by Claudius Ptolemy's Harmonics (200 AD). Gioseffo Zarlino (1517—1590), a major exponent of Renaissance music theory, relied on a priori mathematical quantification of sound, which was based on a particular class of so-called Pythagorean ratios. Mathematical properties of these ratios were used to justify *a priori* the consonance of contemporary music. This clashed with aesthetic perception of sound. Vincenzo argues in favor of the validity of sense perception: perception and aesthetic judgment are explanatory prior to mathematics, and ratios quantify an (undefined) element of sense perception. To prove his point, Vincenzo presented different experiments, demonstrating that the Pythagorean conception of consonance, according to which the octave was embodied by the ratio 1:2, cannot hold for all sound producing physical systems. The study of Vincenzo's experiments in their historical context provide a novel perspective on the relation between the development of scientific inquiry and perception in the history of acoustics.

1:45

1pMU2. Investigation of acoustical parameters in the South Indian musical intervals. Gopu R. Potty (Dept. of Ocean Eng., Univ. of Rhode Island, 115 Middleton Bldg., Narragansett, RI 02882, potty@egr.uri.edu) and Venkitanarayanan Parameswaran (Dept. of Mech. Eng., Indian Inst. of Technol., Kanpur, India)

The South Indian musical style is one of the two prominent styles of classical music styles in India whose roots can be traced back to hundreds of years in the past. The South Indian musical scale is said to have evolved from a set of seven primary notes on the basis of 22 intervals. Present day practice is to use 12 intervals in an octave. A scale is divided in to 12 intervals with fixed values that constitute the basis of musical notes. South Indian classical musical intervals are melodic and are therefore flexible. Compared to the twenty two interval scheme, the 12 interval scheme allows the freedom to perturb the fixed frequency values around their defined positions. In this study, pitch analysis of audio samples is carried out to investigate the deviations from fixed frequency values of musical intervals. Departures from theoretically calculated acoustical values will be demonstrated and discussed. The intervals vary dynamically depending on the artistic individuality and the musical context.

2:00

1pMU3. Clarinet playing frequency predictions: Comparison between analytic and numerical simulations. Whitney L. Coyle (The Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, wlc5061@psu.edu), Jean Kergomard, and Philippe Guillemain (LMA-CNRS, Marseille, France)

The input impedance measurement can provide the resonance frequencies of an instrument and is a standard method used by wind instrument makers in designing modifications. For a complete design, it is necessary to know the playing frequencies themselves, which depend on several control parameters, such as the blowing pressure and reed opening and the input impedance. Using the values of these parameters, we can determine the playing frequencies. This research will analytically deduce these frequencies from the different control parameters and from the input impedance curve. Four effects are known to influence the playing frequency and are examined separately: the flow rate due to the reed motion, the reed dynamics, the inharmonicity of the resonator, and the temperature gradient in the clarinet. The results for a particular clarinet are given and compared to numerically simulated playing frequencies. Experimental methods are also presented and discussed.

2:15

1pMU4. A cross correlation of stringed instruments. Katarzyna Pomian (Phys., Loyola Univ. Chicago, 1536 Courtland Dr., Arlington Height, IL 60004, kpomian@luc.edu)

Stringed instruments can be characterized by their individual body shapes, sizes, types of strings, and sounds. Studying the string resonance, body properties, and high speed photography of the strings, it is possible to describe the individual instruments in their specific functions. We gathered data for 13 different instruments and revealed the variances in their sound, based upon the unique characteristics of each instrument. Our focus was on the comprehensive analysis of the each instrument when correlating all the components of the data. We then aligned all the instruments together and investigated how the varying body shapes influence the sound that is produced by the specific instruments. Our analyses showed how these components work together to create the individually characterized sounds of the different instruments. [This paper is complementary to that of Gordon P. Ramsey.]

2:30

1pMU5. A comparative study of stringed instruments. Gordon Ramsey (Phys., Loyola Univ. Chicago, 6460 N Kenmore, Chicago, IL 60626, gprspinphys@yahoo.com)

Many studies for stringed instruments exist. Most analyses have been made on violins, guitars, and pianos, but comparative studies on these and less popular instruments have not been done. Our research consists of an in

depth analysis of a variety of stringed instruments. The experiments included spectrum analysis, body resonances using Chladni patterns, and high-speed videos to visually observe the string oscillation modes. A combination of these methods was performed on 13 different stringed instruments. The spectral analysis was done on all instruments with the strings plucked or picked at different locations. Corresponding high-speed video was taken on many to observe how the waves propagate along the string. The violin and viola were recorded when bowed to compare the images of the stings when plucked. String resonances were compared to the body resonances to see the synthesis between the two. Comparisons of the spectrum, body resonances, and string oscillations have been made between these instruments to gain a better understanding of how they operate and why each emits the unique sounds that it does. [This paper is complementary to the poster of Katarzyna Pomian at this conference.]

2:45–3:00 Break

3:00

1pMU6. Sound power levels of the Caxirola and different types of caxixis. Talita Pozzer and Stephan Paul (DECC-CT-UFSM, UFSM, Undergraduate Program in Acoust. Eng., Tuiuti, 925. Apto 21, Santa Maria, RS 97015661, Brazil, talita.pozzer@eac.ufsm.br)

In 2014, Brazil will host the FIFA World Cup and Brazilian Musician Carlinhos Brown created the caxirola as the official music instrument, adapting an old African instrument—the caxixi. In both instruments, the sound is generated by hard particles impacting on the walls of a closed basket. While the caxirola is made of environmental-friendly polymer, the caxixi is handcrafted of natural components. At a previous ASA meeting [Pozzer and Paul, J. Acoust. Soc. Am. 134(5), 4187 (2013)], we presented both instruments and sound pressure levels measured at users ears. Being handmade makes the caxixi to be highly variable in size and proportions, in contrast to the caxirola which is an industry product. We now present sound power level (SWL) measurements made by the hemi-anechoic room method and the reverberation chamber method. The SWL measured for the caxirola was 79 to 86 dB (80 to 85 dBA) for transversal and longitudinal use, respectively. The SWL of four different caixixis measured ranged between 69 and 80 dB (70 to 79 dBA). Octave band SWLs were found to be higher for the caxirola compared to all four caxixi in the 500 Hz to 4 kHz bands.

3:15

1pMU7. Detection of musical notes and chords using a holistic approach. Arturo Camacho (Comput. Sci. and Informatics, Univ. of Costa Rica, Escuela de Ciencias de la Computación e Informótica, Universidad de Costa Rica, San José, San José 2060, Costa Rica, arturo.camacho@ecci.ucr.ac.cr)

There are currently two main approaches for the automatic recognition of chords: (1) detecting chords from pitch class profiles, which gives information about chroma, but not height, and (2) detecting individual notes, either by peak picking from a score function or by detecting and then canceling individual notes. We propose a new method that combines both approaches: it detects chords in a holistic way, but at the same time, it gives information about the chroma and height of individual notes. The approach consists in computing scores for individual notes and chords, especially those in closed form (i.e., with small intervals between notes), and then picking the candidates with maximum score, either notes or chords. This approach, inspired in the way musicians perceive chords: as a whole and not as individual notes, avoids the iterative approach of detecting and canceling notes. The method works particularly well for notes within small intervals, which tend to be hard to detect in other approaches.

3:30

1pMU8. Perception of combination tones correlated to cochlear activity. Victoria Suha (Elec. and Comput. Eng., Northeastern Univ., #3430 337 Huntington Ave., Boston, MA 02115, victoria.suha@hotmail.com) and Michael Epstein (Speech-Lang. Pathol. and Audiol., Elec. and Comput. Eng., BioEng., Northeastern Univ., Boston, MA)

String players producing double stops at certain intervals perceive a third note that is not physically present (combination tone). This experiment was designed to determine whether or not this combination tone is a result of measurable cochlear activity. Young, normal-hearing musicians matched the pitch and rated the loudness of combination tones perceived in the presence of pairs of pure tones. These matches and ratings were compared with distortion-product otoacoustic emissions (DPOAEs) generated by the same tone pairs. While there was some variability in listener performance, with some listeners making inconsistent pitch matches across trials, listeners who gave consistent responses pitch matched the combination tone to a frequency close to the strongest DPOAE. This suggests that the perception of combination tones is associated with physical activity within the cochlea.

3:45

1pMU9. Investigating acoustic and electroglottograph features to characterize Passaggio in female singers. Shonda Bernadin (Elec. and Comput. Eng., Florida A&M University-Florida State Univ. College of Eng., 2525 Pottsdamer St., Tallahassee, FL 32310, bernadin@eng.fsu.edu), Richard Morris (Commun. Disord., Florida State Univ., Tallahassee, FL), Lance Ellerbe, and Demissew Kessela (Elec. and Comput. Eng., Florida A&M University-Florida State Univ. College of Eng., Tallahassee, FL)

The purpose of this study was to examine the acoustic and electroglottographic features in the characterization of passaggio in female singers. Three groups of female singers were instructed to sing the notes of the scale for one octave using an “ah” vowel. When singing this octave they sang through a register shift, which is called “passaggio.” Their singing voices were recorded in a two-channel dataset. The first channel captured the acoustic signal using a microphone, and the second channel captured the electroglottographic (EGG) signal using an EGG instrument. This study used VoiceSauce (Shue, Keating, and Vicenik, 2009) analysis software to analyze the features of the two-channel dataset that contribute to the characterization of the passaggio in female singers. Glottal measurements can give more robust information on precise glottal opening and closing moments using the derivative of the EGG signal (DEGG). The results of this investigation also provide an analytical framework for calculating the DEGG in female singers.

4:00

1pMU10. A study of the type and characteristics of relaxing music for college students. Wei-Chun Wang (National Taiwan Univ. of Sci. and Technol., No. 43, Sec. 4, Keelung Rd., Taipei 106, Taiwan, vgnwang@hotmail.com)

It is believed that music has the power to soften emotions and alleviate pains. The essence of the power has being encoded by researchers. This study was aimed to explore the effects of music preference and stress-associated responses of college students when they listen to music. The objectives of this study were (1) to survey the music types of relaxing music for college students, and the difference in gender and study majors; (2) to investigate the effects of musical preference, music expertise, and awareness of musical content on their perceptivity of relaxation; and (3) to analyze the relativities of musical emotions with musical characteristics, such as tempo, mode, and dynamic range. Participants were asked to listen to selected music pieces, and to rate their three-dimensional emotional responses, pleasant-unpleasant, calm-arousal, and relaxing-stress, on five-point Likert scales. Data collection of music compositions and personal music taste was acquired using surveys. The findings are expected to (1) understand college students’ listening habit, (2) collect repertoire suitable for university students in terms of stress releasing, and (3) offer advices for music appreciation teaching, psychological consultation personnel, and clinical therapist.

Session 1pNS

Noise, Architectural Acoustics, and ASA Committee on Standards: Soundscapes: Decisions on Measurement Procedures

Brigitte Schulte-Fortkamp, Cochair
TU Berlin, Einsteinufer 25 TA 7, Berlin 10587, Germany

Klaus Genuit, Cochair
HEAD Acoustics GmbH, Ebertstr. 30a, Herzogenrath 52134, Germany

Chair's Introduction—1:00

Invited Papers

1:05

1pNS1. Sounding Brighton: Update on soundscape planning with a user centric approach. Lisa Lavia (Noise Abatement Society, 8 Nizells Ave., Hove BN3 1PL, United Kingdom, lisa.lavia@noise-abatement.org)

The potential of soundscape planning has been widely illustrated in recent years. Sounding Brighton is a collaborative initiative pioneered by the Noise Abatement Society and Brighton and Hove City Council in 2010 with the support of the former COST Action TD0804. The project continues and is exploring the positive effects soundscapes can have on health, wellbeing, and quality of life. It recently undertook a city-wide soundscape survey and interviews leading to "West Street Story," a night-noise intervention pilot, to gauge whether ambient soundscapes might act as an antidote to the Saturday night drinking culture seen on the city's most dangerous street, and the follow on project: "West Street Tunnel" investigating the same approach in a disused pedestrian subway. The work has also gained inclusion in the Masterplan for the redevelopment of the city center, leading to its willingness to participate in and its acceptance into the European Union funded FP7 SONORUS project looking at holistic ways to include urban sound planning into city planning. This paper will provide an update of the project and its results so far.

1:25

1pNS2. Applicability of measurement procedures in soundscape context—Experiences and recommendations. Klaus Genuit and Fiebig André (HEAD Acoustics GmbH, Ebertstr. 30a, Herzogenrath 52134, Germany, klaus.genuit@head-acoustics.de)

In the context of the ISO/TC 43/SC 1/WG 54, different aspects of soundscape will be subject to standardization. Besides a standardized soundscape definition, understanding and terminology, minimum reporting requirements for measuring soundscapes including measurement uncertainties are currently discussed and will be subject to standardization later. All in all, a wide range of measurement procedures is applied for measuring, describing, documenting, and analyzing soundscapes. However, several aspects of and conditions for measurements are still unclear, which limits the comparability and compatibility of soundscape investigations. It is evident that a common basis of measurement procedures is needed to bring forward current standardization efforts. Consequently, it is very important to share experiences and knowledge about measurement procedures and their general applicability in soundscape context. A thorough discussion about the data quality achieved by certain measurement procedures is inevitable as well. Observations and experiences made in different soundscape studies regarding different measurement procedures will be presented and discussed with respect to their significance and applicability.

1:45

1pNS3. The need for a soundscape taxonomy. Brigitte Schulte-Fortkamp (Inst. of Fluid Mech. and Eng. Acoust., TU Berlin, Einsteinufer 25, Berlin 10587, Germany, b.schulte-fortkamp@tu-berlin.de)

It is for some time now that the standardization process in Soundscape in the ISO/TC 43/WG 54 12913 has started. Meanwhile, it has reached the level of decisions regarding evaluation approaches. Hence, as perception is the foreground of any assessment procedures here, on the one hand interviews do play a major role, but also procedures as sound walks. On the other hand, a time to evaluation related psychoacoustic measurement is needed to bring relevant datasets into triangulation. This paper will present the current state of the art in Soundscape and will discuss further input that is needed in this field.

2:05

1pNS4. Comparison of momentary and retrospective soundscape evaluations. Jochen Steffens (School of Information Studies & CIRMMT, McGill Univ., Josef-Gockeln-Strasse 9, Duesseldorf 40476, Germany, jochen.steffens@fh-duesseldorf.de), Johannes Petrenko (Inst. of Sound and Vib. Eng., Univ. of Appl. Sci., Duesseldorf, Germany), and Catherine Guastavino (School of Information Studies & CIRMMT, McGill Univ., Montreal, QC, Canada)

The Peak and End rule describes the effect that retrospective evaluations of temporal events significantly depend on the most extreme affect (peak) experienced during an episode and on the affect at the ending. Other features, like the duration of an event, seem to be widely negligible. We are testing this hypothesis in the context of soundscape evaluation in a series of listening tests conducted in MontrÕal, Canada, and Duesseldorf, Germany. The soundscapes consisted of recordings of different locations (e.g., railway station, park) and were edited so that there was one presumed emotional "peak moment." The task of the test group was to indicate momentary judgments by continuously adjusting a slider on a computer interface over the course of the stimulus presentation. Additionally, the participants had to make an overall retrospective rating of the soundscapes after listening to them. To investigate attention effects in the course of the test task a second group was asked to only judge the sounds retrospectively. Preliminary results indicate that the Peak and End rule in combination with the averaged momentary evaluations well predict the retrospective judgments. Within this contribution the results of the experiment will be presented and implications for soundscape design will be discussed.

2:25

1pNS5. Case studies of historic soundscapes with cultural importance. David Lubman (dlAcoustics, 14301 Middletown Ln., Westminster, CA 92683, dlubman@dlacoustics.com)

Case studies discussed include Bow Bells, in which the sounds of ringing bells of the East London church of St. Mary Le Bow became a soundmark that defined the "Cockney" ethnicity. (A "Cockney" is a person born within the sound of Bow Bells.) With London's growth, more churches were built. The combined sounds of their bells became the community soundscape for East London. It inspired beloved children's poetry (Oranges and Lemons ? Says the great Bell of Bow). That children's poem was learned by generations of English children, and the poem became the sound symbol for all of England. This became very important in WWII when Nazi arial bombing destroyed Bow Bells creating a serious morale problem for England. To reassure their countrymen that the "home fires" were still burning, BBC used a 1926 sound recording of Bow Bells as an interval signal in their worldwide radio broadcasts. On a smaller scale, accidental whispering arches can give buildings "a sense of place." A 6th C monastery at Clonmacnoise, Ireland, has a large and elaborately carved door installed that became a storied whispering archway. The storied accidental whispering archways at Grand Central Terminal and Pennsylvania's West Chester University are also described.

Contributed Papers

2:45

1pNS6. Cross-modal soundscape mapping: Integrating ambisonic field recordings with high dynamic range spherical panoramic photography to produce interactive maps. J. Parkman Carter (Architectural Acoust., Rensselaer Polytechnic Inst., 32204 Waters View Circle, Cohoes, NY 12047, cartej8@rpi.edu) and Jonas Braasch (Architectural Acoust., Rensselaer Polytechnic Inst., Troy, NY)

We cannot "measure" the soundscape any more than we can "measure" the ocean, the city, or the wilderness. Being comprised of myriad complex elements, conditions, and relationships between sound sources and sound perceivers, the soundscape—and any sufficient description of it—must account for several different, but significantly interrelated, dimensions: physical, spatial, temporal, perceptual, cultural, and historical. How, then, are we to meaningfully document the soundscape? If we are to begin to understand the soundscape's impact on us—and our impact upon it—we need new methods to capture and represent the multisensory extents of a soundscape without reverting to one-dimensional quantitative abstractions. This project proposes an interdisciplinary method to record a soundscape's multisensory attributes by combining aural and visual information in a structured way which links the directionality of view and sound arrival. This method integrates multi-directional Ambisonic audio recordings with high dynamic range (HDR) spherical panoramic photography in the form of interactive maps and virtual tours. Case studies using the cross-modal soundscape mapping method will be presented.

3:00

1pNS7. The sound pressure level observing transponder: A satellite-enabled sensor package for near real time monitoring. Peter Marchetto, Christopher W. Clark (BioAcoust. Res. Program, Cornell Lab. of Ornithology, Cornell Univ., 206 Langmuir Lab., 95 Brown Rd., #1012, Ithaca, NY 14850, pmm223@cornell.edu), and Daniel Aneshansley (Biological and Environ. Eng., Cornell Univ., Ithaca, NY)

Many *in situ* sensing applications for bioacoustic ecology have suffered from a lack of means to communicate information in near real time. The monitoring of incident noise on an individual animal and its behavioral response to it were the focus of this project. The sensor platform described herein may be used to create acoustic field maps of a habitat in near-realtime.

3:15–3:45 Panel Discussion

Session 1pPP

Psychological and Physiological Acoustics: From Protection to Perception

Charlotte M. Reed, Chair
Massachusetts Inst. of Technol., 77 Massachusetts Ave., Rm. 36-751, Cambridge, MA 02139

Chair's Introduction—1:00

Contributed Papers

1:05

1pPP1. Measurement of hearing-protector attenuation using auditory steady state responses. Olivier Valentin (Dept. of Mech. Eng., ècole de technologie supérieure, 1100 Rue Notre-Dame Ouest, Montréal, QC H3C 1K3, Canada, m.olivier.valentin@gmail.com), Michael Sasha John (Rotman Res. Institute/Inst. of BioMater. and Biomedical Eng., Univ. of Toronto, Toronto, ON, Canada), and Frédéric Laville (Dept. of Mech. Eng., ècole de technologie supérieure, Montréal, QC, Canada)

There is a need to assess hearing protection device (HPD) attenuation to ensure that individuals receive adequate protection from noise. Present methods of attenuation measurement have limitations. Objective measurements such as field microphone in real ear (F-MIRE) do not assess bone conducted sound. Psychophysical measurements such as real ear attenuation at threshold (REAT) are biased due to the low frequency masking effects from test subjects' physiological noise, and the variability of measurements based on subjective response. We explored using auditory steady state responses (ASSR) as a technique that may overcome these limitations. ASSRs were recorded in ten normal hearing adults, using both "normal" and "occluded" conditions. Stimuli included both narrow band noises and pure tones (500 and 1000 Hz), amplitude modulated at 40 Hz. Stimuli were presented through either loudspeakers or headphones, at 45, 55, and 65 dB SPL. "Physiological attenuation" was calculated as the difference between ASSR values (both amplitude and phase) for normal and occluded conditions. Physiological attenuation estimates were compared to in-ear (objective) and psychophysical (subjective) measurements. Grand mean ASSR data complied well with in-ear and subjective measurements. Further work is needed to provide accurate assessment of ASSR-based physiological attenuation for individual subjects.

1:20

1pPP2. A public science experiment on the link (if any) between hearing loss and a lifetime of loud music exposure. Michael Akeroyd, William Whitmer (MRC/CSO Inst. of Hearing Res.-Scottish Section, New Lister Bldg., Glasgow Royal Infirmary, Glasgow, Strathclyde G31 2ER, United Kingdom, maa@ihr.gla.ac.uk), Robert Mackinnon, Heather Fortnum (NIHR Nottingham Hearing Biomedical Res. Unit, Univ. of Nottingham, Nottingham, United Kingdom), and David R. Moore (Commun. Sci. Res. Ctr., Cincinnati Childrens Hospital, Cincinnati, OH)

It is well known that a lifetime of exposure to loud industrial noise can lead to substantial amounts of hearing damage. The effect of a lifetime of loud music exposure is far less certain, however. To assess if there is such a link, we have launched a public science experiment that combines (a) a questionnaire on the amount of listening to loud music over one's life as well as standard questions on their hearing, and (b) a triple-digit speech-in-noise test specifically crafted for high-frequency hearing losses. The experiment has a dedicated website and takes advantage of the 2013 Centenary of the Medical Research Council to act as a "pull" for the public to participate. Over the 100 years, the MRC has been funding research, there has been revolution after revolution in the technology for music reproduction for the home or stage and in the volumes they can reach. The talk will describe the design of the experiment, the results so far, and our experience-based suggestions for running future web-based, public-science experiments. [Work supported by the Medical Research Foundation, the Medical Research Council, and the Chief Scientist Office, Scotland.]

1:35

1pPP3. A novel mode of off-frequency hearing as a result of defective outer hair cells hair bundles unveiled by Nherf1-/- mice. Aziz El-Amraoui, Kazusaku Kamiya, Vincent Michel (Neurosience, Genetic and Physiol. of Audition, Institut Pasteur, 25 rue du Dr Roux, Paris 75015, France, aziz.el-amraoui@pasteur.fr), Fabrice Giraudet (Faculté de Médecine, Université d Auvergne, Clermont-Ferrand, France), Maria-Magdalena Gerogescu (Neuro-Oncology, Texas M.D. Anderson Cancer Ctr., Houston, TX), Paul Avan (Faculté de Médecine, Université d Auvergne, Clermont-Ferrand, France), and Christine Petit (Neurosience, Genetic and Physiol. of Audition, Institut Pasteur, Paris, France)

Nherf1, a PDZ-domain-containing protein, was identified in the hair bundle in differentiating outer hair cells (OHCs). Nherf1-/- mice showed apparently mild hearing-threshold elevations at mid/high sound frequencies, associated to OHC hair-bundle shape anomalies, prominent in the basal cochlea. This mild impact on hearing sensitivity was discordant with the finding of almost non-responding OHCs in the basal cochlea as assessed by distortion-product otoacoustic emissions and cochlear microphonic potentials. Unlike normal mice, responses of Nherf1-/- mice to high-frequency test tones were not masked by tones of neighboring frequencies. Efficient masker tones displayed unusual characteristics: maximal efficiency at lower frequencies (up to two octaves lower than the test tone), and at low levels (up to 25 dB below test-tone level). This, and the relative growth of the masker and test tones, suggests that mid-high frequency tones of moderate intensity are detected off-frequency, in the functionally unaffected apical cochlear region. Our results establish that Nherf1 is critical for hair bundle morphogenesis and reveal a novel mode of off-frequency detection, probably involving the persistent contact between OHCs and the tectorial membrane. These findings suggest how to circumvent major pitfalls in hearing assessment of some patients, by avoiding misleading interpretations of hearing thresholds.

1:50

1pPP4. Evidence against power amplification in the cochlea. Marcel van der Heijden and Corstiaen Versteegh (Neurosci., Erasmus MC, P.O.Box 2040, Rotterdam 3000 CA, Netherlands, m.vanderheyden@erasmusmc.nl)

Sound-induced traveling waves in the mammalian inner ear peak at a frequency-dependent location. Some form of motility is widely believed to boost this peaking by injecting extra power into the wave. We determined the power carried by the wave from two-point recordings of basilar membrane motion in sensitive cochleae. Up to moderate intensities, the peak wave power was slightly less than the acoustic power entering the middle ear. At higher intensities, an increasingly smaller fraction of the acoustic

power reached the peak region. Thus, cochlear dynamic compression stems from variable dissipation rather than saturating amplification. Additional measurements revealed that the peaking of the wave envelope is realized by focusing the acoustic power rather than amplifying it.

2:05

1pPP5. Effects of self-generated noise on estimates of detection threshold in quiet in school-age children and adults. Emily Buss, Heather L. Porter (Otolaryngol.—Head and Neck Surgery, UNC Chapel Hill, 170 Mannng Dr., G190 Physicians Office Bldg., CB# 7070, Chapel Hill, NC 27599, ebuss@med.unc.edu), Lori J. Leibold (Allied Health Sci., UNC Chapel Hill, Chapel Hill, NC), John H. Grose, and Joseph W. Hall (Otolaryngol.—Head and Neck Surgery, UNC Chapel Hill, Chapel Hill, NC)

Detection in quiet develops earlier in childhood for high than low frequencies. The present study tested the hypothesis that self-generated noise could play a role in this finding. When adults listen for sounds near threshold, they tend to engage in behaviors that reduce physiologic noise (e.g., quiet breathing), which is predominantly low frequency. Children may not suppress self-generated noise to the same extent as adults. This possibility was evaluated by measuring sound levels in the ear canal simultaneous with adaptive threshold estimation for 250-, 1000-, and 4000-Hz pure tones. Stimuli were delivered and recordings were made using a single foam insert. Listeners were children (4.3–16.0 yr) or adults. Consistent with previous data, the effect of child age was robust at 250 Hz, whereas thresholds of even the youngest listeners were nearly adult-like at 4000 Hz. The spectral shape of self-generated noise was generally similar across listener age groups, although the magnitude was higher in younger listeners. Trial-by-trial data were evaluated to assess the relationship between noise and the accuracy of listener responses: there was an association for younger listeners. These results provide preliminary evidence that self-generated noise may play a role in the prolonged development of low-frequency detection in quiet.

2:20

1pPP6. Adaptation reveals automatic pitch-shift detectors. Samuel R. Mathias, Barbara Shinn-Cunningham (Ctr. for Computational Neurosci. and Neural Technol., Boston Univ., 677 Beacon St., Boston, MA 02215, smathias@bu.edu), Christophe Micheyl (Starkey Hearing Res. Ctr., Berkeley, CA), and Andrew J. Oxenham (Dept. of Psych., Univ. of Minnesota, Minneapolis, MN)

Previous work suggests that the auditory system contains automatic frequency- or pitch-shift detectors (PSDs). These hypothetical units may serve to bind successive sounds together, and appear to be tuned to optimally detect frequency shifts of about 1 semitone. The present work describes several experiments that provide more evidence for PSDs. In all experiments, listeners judged the direction (up or down) of small relevant frequency shifts whilst ignoring prior irrelevant shifts that were usually much greater in magnitude. In almost all conditions and experiments, there was a nonmonotonic relationship between sensitivity to the relevant shift (d') and the magnitude of the irrelevant shift (Δ). d' declined as a function of increasing Δ up to about 1 semitone, and further increases in Δ increased d'. The "dip" in d' could not be explained by selective attention, frequency-specific adaptation, or energetic masking. PSDs provide a convenient explanation for these results if one assumes that (a) listeners discriminate small frequency shifts using PSDs, (b) PSDs are automatically activated by irrelevant frequency shifts, causing them to adapt, and (c) maximal adaptation of PSDs occurs at around 1 semitone.

2:35

1pPP7. Effects of presentation method and duration on alarm detection threshold in the presence of loud pink noise. Buddhika Karunarathne, Richard So (Industrial Eng. & Logistics Management, Hong Kong Univ. of Sci. & Technol., Clear Water Bay, Kowloon, Hong Kong, Kowloon 00000, Hong Kong, jbpdk@ust.hk), and Anna Kam (Dept. of Otorhinolaryngology, Head & Neck Surgery, Chinese Univ. of Hong Kong, Hong Kong, Hong Kong)

Detection of pure tone signals in the presence of noise has been thoroughly studied. Most of these studies have used monaural presentation of audio stimuli. Also, studies testing alarm detection in the presence of noise are limited. In 2013, Karunarathne *et al.*, conducted a study and found out that human listeners were able to detect an alarm in negative signal-to-noise ratios (SNRs), as low as −-24 dB. This study aims to investigate the effects of presentation method and duration of the alarms on detection threshold. Eight conditions varied by presentation method (monaural vs spatial) and alarm duration were tested. Sixteen human subjects with normal hearing were given the task of identifying which one of two sound intervals contained an alarm along with 80dBA pink noise. Thresholds were estimated as the 79.4% points on the psychometric functions, using adaptive 2-Interval Forced Choice (2IFC) procedure with a 3-down 1-up rule. Results indicated that detection thresholds were statistically significantly lower in spatial condition compared to monaural. The effect of alarm duration was not significant in both spatial and monaural conditions. Thresholds lower than −-30 dB SNR were observed in the spatial condition, which agreed with the findings of Karunarathne *et al*. and further extended threshold boundaries.

2:50

1pPP8. Predictions of the magnitude of tonal content on the basis of partial loudness. Jesko L. Verhey, Jan Hots (Dept. of Experimental Audiol., Otto von Guericke Univ., Leipziger Str. 44, Magdeburg 39120, Germany, jesko.verhey@med.ovgu.de), and Matthias Vormann (Hörzentrum Oldenburg GmbH, Oldenburg, Germany)

Environmental sounds containing clearly audible tonal components are considered to be more annoying than sounds without these components. Several standards include sections dedicated to the assessment of tonal components in sound. These standards have in common that they estimate the magnitude of the tonal components (in the following referred to as tonalness) as the level above the noise background. Recent studies indicate that partial loudness of the tonal component determines the tonalness. The present study tests this hypothesis by comparing experimental data on tonalness of sounds with multiple tonal components with predictions of a loudness model. It is shown that partial loudness is a better predictor of the perception of tonal portions of a sound than the intensity. This approach may be useful for future standardization of the perception of clearly audible tonal components in noise.

3:05–3:20 Break

3:20

1pPP9. The unknown effects of amplitude envelope: A survey of *Hearing Research*. Jessica Gillard (Psych., Neurosci. & Behaviour, McMaster Inst. for Music and the Mind, 1280 Main St. West, Hamilton, ON L8S 4M2, Canada, gillarj@mcmaster.ca) and Michael Schutz (School of the Arts, McMaster Inst. for Music and the Mind, Hamilton, ON, Canada)

In auditory research, the use of amplitude-steady tones with abrupt onsets and offsets is quite common. While these types of "flat" tones offer a great deal of control, they are not representative of the types of sounds we hear outside the laboratory. In everyday listening we are much more likely to encounter "percussive" (i.e., exponentially decaying) sounds, with offsets conveying detailed information such as the materials and force used to produce the sounds—information that is absent in abruptly ending flat tones. Given that differences in perception have been reported when using different amplitude envelopes (Grassi and Pavan, 2012; Neuhoff, 1998; Schutz, 2009), we became interested in determining the prevalence of flat and percussive tones in auditory research publications. Here, we surveyed the journal Hearing Research and classified the temporal structure of sounds used into five categories: flat, percussive, click train, other, and undefined. We found 42.5% of sounds were flat (approximately 13% were click trains, 4% other, and 40% undefined). This finding is consistent with our previous surveys of *Music Perception* and *Attention, Perception, and Psychophysics*, suggesting that flat tones dominate auditory research, and the perceptual effects of more naturalistic sounds are relatively unknown and ripe for future exploration.

3:35

1pPP10. Could binaural listening help segregate competing sound sources? Marion David (Laboratoire Génie Civil et Bãtiment, ENTPE, rue Maurice Audin, Vaulx-en-Velin 69518, France, marion.david@entpe.fr), Nicolas Grimault (Ctr. de Recherche en NeuroSci. de Lyon, Université de Lyon, Lyon, France), and Mathieu Lavandier (Laboratoire Génie Civil et Bãtiment, ENTPE, Vaulx-en-Velin, France)

When a sound is alternatively played from two positions, the resulting sequence at one ear is an alternate of two spectrally different sounds, due to head coloration. A previous study showed that these monaural spectral differences can induce segregation. Here, binaural cues are introduced to investigate whether they could strengthen segregation. A rhythmic discrimination task evaluated obligatory streaming with speech-shaped noises. In a first experiment, head-related transfer-functions were modified to introduce independently the interaural time and level differences (ITD and ILD). The results suggested that both ITD and ILD favored segregation. Moreover, the perceptive organization was rather based on the monaural variations in spectrum and intensity at one ear rather than on the spectral interaural differences. Since the binaural cues allow the auditory system to lateralize sounds, a second experiment was intended to determine to which extent the influence of ITD was due to the interaural difference and/or to the resulting associated perceived position. Temporal delays were introduced to simulate different ITDs. The perceived position was modified by manipulating these ITDs independently across frequency. The results, combined with a subjective test of lateralization, showed that both ITD and perceived position influence stream segregation.

3:50

1pPP11. Feedback loops in engineering models of binaural listening. Jens P. Blauert, Dorothea Kolossa (Inst. of Commun. Acoust., Ruhr-Universitaet Bochum, Bochum 44780, Germany, jens.blauert@rub.de), and Patrick Danés (LAAS-CNRS, Univ. Toulouse III Paul Sabatier, Toulouse, France)

Hearing models for tasks like auditory scene analysis or sound-quality judgments can run into severe problems when acting in a purely bottom-up, that is, signal driven manner, as they may have to follow up on all possible output options until a final decision has been taken. This may lead to a combinatorial explosion. A way out is the inclusion of top-down, that is, hypothesis-driven processes. In top-down processing, the number of states to be evaluated can be reduced substantially, when the system knows what to look for and thereby focuses attention on states which make sense in a given specific situation. To implement adequate top-down processes, various feedback loops will be included in our models, some more specific, others more general. The general ones originate from the concept that the listener model ("artificial listener") actively explores acoustic scenes and thereby develops its aural world in an autonomous way. Following this notion, it is attempted to model listeners according to the autonomous-agents paradigm, where agents actively learn and listen. [Work performed in the context of the EU project TWO!EARS,<www.twoears.eu>.]

4:05

1pPP12. Quantifying the better ear advantage in the presence of interfering speech. Esther Schoenmaker and Steven van de Par (Acoust. Group, Univ. of Oldenburg, Carl von Ossietzkystrasse 9-11, Oldenburg D-26129, Germany, esther.schoenmaker@uni-oldenburg.de)

In cocktail party listening with spatially separated speech sources, better ear listening is known to make a major contribution to speech intelligibility. The better ear is generally defined as the ear that receives the highest signal-to-noise ratio (SNR). Usually, this SNR is calculated based on the total length of the signal. However, this seems inappropriate when speech signals are involved since these are highly modulated both in the time and frequency domain. On a perceptual level, modulated maskers give rise to a higher target speech intelligibility than their unmodulated counterparts through the presence of glimpses. A simple measure to quantify the better ear advantage while taking these spectrotemporal fluctuations into account is introduced. In a headphone experiment, three simultaneous sequences of vowel-consonant-vowel utterances were presented at a fixed target-to-masker ratio. The stimuli were rendered with head-related transfer functions and contrasted against stimuli that did not contain any interaural level differences (ILDs) and, as a consequence, allowed no better ear listening. Using the proposed metric, we are able to explain differences in intelligibility for these speech-in-speech mixtures that would remain unexplained by the conventional SNR both for stimuli with and without ILDs.

4:20

1pPP13. Relative importance of individual spectral features for intraconic localization. Griffin D. Romigh (711th Human Performance Wing, Air Force Res. Labs, 4064 chalfonte, beavercreek, OH 45440, griffin.romigh@wpafb.af.mil), Brian D. Simpson (711th Human Performance Wing, Air Force Res. Labs, Dayton, OH), Eric R. Thompson (Ball Aerosp. & Technologies Corp., Dayton, OH), and Nandini Iyer (711th Human Performance Wing, Air Force Res. Labs, Dayton, OH)

Most researchers agree that physical features present in the monaural spectra act as the primary set of cues for sound source localization within a cone-of-confusion. Less consensus has been reached as to whether these localization judgments are based on the presence of simple spectral features, or whether a more broad spectrum pattern matching occurs in which information across many bands are utilized. The present work first describes a head-related transfer function decomposition technique by which spectral cues are separated into components utilized for lateral localization judgments (when combined binaurally) and those used for localization judgments within a cone-of-confusion. This decomposition allows us to modify individual spectral features of an arbitrary virtual stimulus while maintaining its naturalness and perceived lateral location, criteria that were not adhered to in previous studies. Using this technique, a set of virtual localization studies was conducted in which individual spectral features were removed to examine their relative importance to intraconic localization judgments. Results indicate that while both spectral peaks and notches contribute to localization judgments, spectral information appears to be integrated across multiple frequency bands.

4:35

1pPP14. The role of modulation processing in binaural masking-patterns. Bjoern Luebken (Dept. of Experimental Audiol., Otto von Guericke Univ., Leipziger Str. 44, Magdeburg 39120, Germany, bjoern.luebken@med.ovgu.de), Steven van de Par (Acoust. Group, Carl von Ossietzky Univ., Oldenburg, Germany), and Jesko L. Verhey (Dept. of Experimental Audiol., Otto von Guericke Univ., Magdeburg, Germany)

Binaural masking pattern experiments indicate a continuous decrease in the binaural masking-level difference (BMLD) with increasing spectral distance of a tonal signal to a narrowband noise masker. Previous studies suggested that this decrease in BMLD is due to additional modulation cues in monaural off-frequency masking conditions. An own masking pattern experiment with an additional interferer to mask modulation cues supported this hypothesis. The interferer is positioned spectrally below/above the masker for the signal above/below the masker with a spectral distance equal to the distance between masker and signal. This interference tone has a large impact on the thresholds without a binaural cue, as expected. Such an increase in the diotic thresholds is predicted on the basis of a modulation-filterbank model, but only if an across-channel modulation processing is assumed. The interferer also increases the dichotic thresholds, indicating an influence of modulations processing also in conditions where binaural cues are present. Assuming that modulation cues are masked by the interference tone and thus the detection is based on energy cues, the binaural data indicate effectively wider binaural filter, as previously suggested on the basis of notched-noise experiments.

4:50

1pPP15. Hearing better with interaural time differences and bilateral cochlear implants. Zachary M. Smith (Res. & Technol. Labs, Cochlear Ltd., 13059 E Peakview Ave., Centennial, CO 80111, zsmith@cochlear.com), Alan Kan, Heath G. Jones, Melanie Buhr-Lawler, Shelly P. Godar, and Ruth Y. Litovsky (Univ. of Wisconsin Waisman Ctr., Madison, WI)

While bilateral cochlear implant (BiCI) recipients generally receive significant benefits from the addition of a second ear, evidence suggests that much of the benefit is attributed to monaural effects or to the availability of interaural level differences. The benefits are less than those in normal-hearing listeners, however, in part because the sound localization and speech unmasking that are measured may show greater benefits if interaural time difference (ITD) cues were also available. To date, there is a paucity of evidence that ITDs can be captured and saliently delivered by cochlear implant processors. In this study, we used a research processing strategy that explicitly codes ITD cues. We measured BiCI listeners' ITD sensitivity to broadband speech material and ITD-based unmasking in a multi-talker listening scenario by directly presenting sounds through the accessory inputs of their sound processors. Performance was compared to that with the commercial ACE processing strategy. Initial results show that some subjects with good ITD sensitivity can also take advantage of ITD to better understand a target talker in the presence of a masking talker at low target-to-masker ratios. This suggests that improving ITD perception in BiCIs may lead to better hearing outcomes in real-world listening situations.

5:05

1pPP16. Resolution and integration within auditory temporal windows. Xiangbin Teng, Xing Tian, and David Poeppel (Psych., NYU, 6 Washington Pl., New York, NY 10003, david.poeppel@nyu.edu)

Temporal integration in auditory and speech perception is investigated in a variety of experimental contexts, using different types of signals. How the auditory system manages the tension between resolution, on the one hand, and integration, on the other—and in particular how the system integrates acoustic information over time to form a unitary percept—remains unclear. Using non-speech signals with temporal structure at different scales (30 ms, 200+ ms), we tested in four psychophysical experiments how "local" (shorter-scale) and "global" (longer-scale) auditory information is resolved and integrated. We provide evidence (supported by independent electrophysiological data) that the auditory system extracts temporally detailed acoustic information using small temporal windows, and then integrates that information over approximately 200 ms. Importantly, the fine-detailed information is not fully accessible as sound duration increases to over ~100 ms. Further, we show that representation of acoustic information requires ~150–200 ms, and that the representation of fine-detailed information can compromise temporal integration. The findings demonstrate the time scales over which the integration and resolution of auditory information cooperate and conflict, and thus shed light on the mechanisms of temporal integration.

MONDAY AFTERNOON, 5 MAY 2014 553 A/B, 1:25 P.M. TO 4:25 P.M.

Session 1pSA

Structural Acoustics and Vibration and Underwater Acoustics: Undersea Vehicle Noise

Robert M. Koch, Cochair
Chief Technol. Office, Naval Undersea Warfare Ctr., Code 1176 Howell St., Bldg. 1346/4, Code 01CTO, Newport, RI 02841-1708

Nickolas Vlahopoulos, Cochair
Univ. of Michigan, 2600 Draper Dr., Ann Arbor, MI 48109

Chair's Introduction—1:25

Invited Papers

1:30

1pSA1. Acoustic signature of underwater vehicles. Joe M. Cuschieri (Lockheed Martin MST, 100 East 17th St., Riviera Beach, FL 33404, joe@cuschieri.us)

Underwater vehicles are becoming more commonly used for a number of applications ranging from commercial to defense. While the acoustic signature of underwater vehicles is understood to be important, there are presently more pressing issues which are fundamental to the success of using underwater vehicles beyond the lab or controlled environment. Such issues incorporate power sources, navigation accuracy, reliable control, etc. However, while acoustic signature may not be in the forefront, need for low acoustic signature and low self-noise is important as the noise form the underwater vehicle can impact the operation of the acoustic sensors. In this paper, components that influence the radiated noise from underwater vehicles are identified and discussed, especially as these relate to their impact on the overall acoustic radiation. Approaches to estimate the underwater radiated noise based on other data when in water data is not available are discussed.

1:50

1pSA2. Decreasing the radiated acoustic and vibration noise of both prop-driven and buoyancy-driven autonomous underwater vehicles. Richard Zimmerman, Gerald L. D'Spain (Marine Physical Lab, Scripps Inst. of Oceanogr., 291 Rosecrans St., San Diego, CA 92106, gdspain@ucsd.edu), Peter Brodsky (Appl. Phys. Lab, Univ. of Washington, Seattle, WA), Mark Stevenson (SPAWAR SSC Pacific, San Diego, CA), Mark Zumberge, and John Orcutt (Marine Physical Lab, Scripps Inst. of Oceanogr., San Diego, CA)

Our previously published results from decreasing the radiated acoustic and vibration noise of a mid-size, prop-driven autonomous underwater vehicle (AUV) show self noise levels recorded at sea by an AUV-mounted hydrophone array that are at, or below, typical background ocean noise levels across the frequency band above 200 Hz. The remaining noise below 200 Hz is primarily vibration induced. The modifications required to achieve this 20–50 dB reduction in propulsion and steering system noise levels will be reviewed in this talk. In addition, at-sea measurements of the acoustic noise radiated by the large (30 L) buoyancy engine on a 20-ft wing span flying wing autonomous underwater glider are presented. Whereas a prop-driven system operates continuously, the buoyancy-driven propulsion system has a very low duty cycle of a few percent; it is on only for about 3 min during each dive cycle. Onboard self noise from the glider's internal fluid-based roll control system far exceeds that from an aileron-based system. However, the former system provides control authority at or near neutral buoyancy. [Work supported by the Office of Naval Research and BP.]

2:10

1pSA3. Aspect-dependent acoustic characterization of an underway autonomous underwater vehicle. John Gebbie, Martin Siderius (Northwest Electromagnetics and Acoust. Res. Lab., Portland State Univ., 1900 SW 4th Ave., Ste. 160, Portland, OR 97201, jgebbie@ece.pdx.edu), and John S. Allen (Dept. of Mech. Eng., Univ. of Hawai i-Manoa, Honolulu, HI)

Acoustic emissions emitted by an underway REMUS-100 autonomous underwater vehicle (AUVs) are analyzed with array beamforming. Characterizing emissions of underway AUVs is challenging due to a variety of factors such as low source levels of some vehicle types, continually varying propagation conditions, and inherent uncertainties in vehicle location. While aspect-dependent spectral and source level emissions are available for a wide range of surface craft types, few articles have analyzed these for underway AUVs. Array beamforming is a known method of increasing gain of a weak signal in the presence of interference and noise, and propagation modeling tools can provide estimates of transmission loss between a source and receiver. These techniques are used to measure the aspect-dependent source level and spectral content from the propulsion system of a REMUS-100 AUV deployed near a fixed array near Honolulu Harbor, Hawaii, by fusing measured acoustic data with multi-sensor navigational records recorded on the vehicle. As AUVs become more widely used, characterizing their acoustic radiation will help predict how these platforms interact with and impact the environments in which they are deployed.

2:30

1pSA4. Concurrent control and plant identification to support overall noise reduction of autonomous underwater vehicles. Jason D. Holmes, Alison B. Laferriere, and Christopher G. Park (Sensor Systems, Raytheon BBN Technologies, 10 Moulton St., Cambridge, MA 02375, jholmes@bbn.com)

Commercial, off-the-shelf (COTS) autonomous underwater vehicles (AUVs) are not usually designed to be low-noise in the 100 Hz–10 kHz band. Options to mitigate the noise include mechanical re-design, adaptive filtering of the noise on any acoustic sensor using the vehicle as a platform, or active noise control (ANC). The first of these options eliminates the benefits of using a COTS platform. The second option can be attractive but is most effective for propulsor noise (and not other sources like the depth sounder) and does not mitigate the issue if multiple vehicles are used in coordination. An on-board projector can enable ANC of the vehicle noise. For small, nearly acoustically compact vehicles, the directionality of propulsor noise is not very complex, suggesting global control using a limited set of the acoustic sensors as a control metric. This paper explores methods for performing concurrent control of narrow-band vehicle noise while performing the low-noise, broad-band plant identification necessary to support the control. In addition to the reductions in self noise (on a single vehicle and multiple vehicles), the ability to use the information in the plant estimate to perform "quiet" depth sounding is explored.

2:50

1pSA5. Modeling the acoustic color of undersea structures. David Burnett (Naval Surface Warfare Ctr., Code CD10, 110 Vernon Ave., Panama City, FL 32407, david.s.burnett@navy.mil)

The Naval Surface Warfare Center Panama City Division has developed a 3-D finite-element computer simulation system, PC-ACOLOR, for modeling the "acoustic color" (target strength as a function of frequency and aspect angle) of single and multiple realistic elastic structures that are near to or straddling the water/sediment interface at the bottom of the ocean. Target strength is a measure of the intensity of scattered fields, but radiated noise can also be modeled, yielding sound pressure level as a function of frequency and aspect angle. PC-ACOLOR employs 3-D continuum mechanics throughout the entire computational domain; no engineering approximations are used anywhere. The talk will give an overview of PC-ACOLOR: important structural acoustic concepts, e.g., the relationship between 3-D modeling and evanescent wave solutions to the Helmholtz pde; modeling techniques; verification and validation; and examples of different types of vehicles and other structures that have been modeled.

3:10–3:30 Break

3:30

1pSA6. A computational approach to flow noise. Donald Cox, Daniel Perez, and Andrew N. Guarendi (Naval Undersea Warfare Ctr., 1176 Howell St., Newport, RI 02841, donald.l.cox@navy.mil)

The ability to calculate the noise internal to a structure due to external flow is a necessity for the optimal design of a low noise structure. Whether the structure is an automobile, an airplane or an acoustical array, the goal is the same: to in some way minimize the acoustic pressure/particle velocity resulting from flow excitation at a design location. The excitation is usually that due to wall pressure fluctuations resulting from turbulent boundary layer (TBL). There is a long history of modeling plate excitation due to TBL loads. In most cases, the existing work makes use of statistical, empirically based models for the TBL excitation. This work focuses on combining the capabilities of computational fluid dynamics with computational structural acoustics to enable the calculation of flow noise primarily for undersea vehicles. The work is limited to the non-coupled problem, where the flow calculations are made over a non-deforming boundary with the goal of calculating wall pressure fluctuations and using them as loads on a finite element structural acoustics model. The ultimate goal of this work is to develop the capability to calculate flow noise for three dimensional undersea structures for which analytical approaches are not possible.

3:50

1pSA7. Energy finite element formulation for unbound acoustic domains. Nickolas Vlahopoulos (Univ. of Michigan, 2600 Draper Dr., Ann Arbor, MI 48109, nickvl@umich.edu) and Sergey Medyanik (Michigan Eng. Services, Ann Arbor, MI)

The Energy Finite Element Analysis (EFEA) method has been developed for conducting structural-acoustics simulations for complex vehicles at mid-to-high frequencies where conventional finite element methods are no longer computationally efficient. In applications where exterior heavy fluid loading effects must be included in the model, these effects are currently accounted in the EFEA formulation through an added mass and a radiation damping approach. This is suitable when there is an interest in computing the vibration of the structure and the total radiated power emitted in the exterior fluid. A new formulation for modeling the exterior fluid explicitly with energy finite elements is presented. There are three main components in the new development: deriving the differential equation; solving it numerically using a finite element approach; and introducing infinite finite elements in order to represent the non-reflective conditions at the outer domain of a finite element model. In this presentation, the technical aspects associated with the main elements of the new EFEA formulation for exterior non-reverberant acoustic domains will be discussed. Comparisons with analytical closed form solutions for validating the new developments and their numerical implementation will be presented.

Contributed Paper

4:10

1pSA8. Effectiveness of energy finite element analysis applied to submerged undersea vehicle noise prediction. Michael A. Jandron, Robert M. Koch (Naval Undersea Warfare Ctr., Code 8232, Bldg. 1302, Newport, RI 02841, michael.jandron@navy.mil), Allan F. Bower (School of Eng., Brown Univ., Providence, RI), and Nickolas Vlahopoulos (Dept. of Naval Architecture and Marine Eng., Univ. of Michigan, Ann Arbor, MI)

Traditionally, solving structural-acoustics problems has posed significant computational challenges at very high wavenumbers because of the mesh refinement required. As such, instead of resolving each wavelength, the fundamental goal in Energy Finite Element Analysis (EFEA) is to average the energy over many wavelengths. The envelope of average energy behaves as a slowly varying exponential, which is much easier to solve numerically. EFEA thus computes the average energy over a region which can be used to determine undersea vehicle self- and radiated noise in much the same way as conventional FEA modeling but the size of the mesh does not suffer the same restrictions. In this talk, self-noise model predictions for a representative undersea vehicle are discussed to demonstrate the effectiveness of EFEA. This model includes an explicitly modeled acoustic domain to allow radiated energy to propagate in the acoustic medium and possibly reenter the vehicle structure. Results are validated with dense FEA models and the post-processing algorithms used for this purpose are discussed.

Session 1pSC

Speech Communication: Methods and Models for Speech Communication (Poster Session)

Stephanie Del Tufo, Chair
Dept. of Psychol., Univ. of Connecticut, 406 Babbidge Rd., Storrs, CT 06269-1020

All posters will be on display from 1:00 p.m. to 5:00 p.m, To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 1:00 p.m. to 3.00 p.m. and contributors of even-numbered papers will be at their posters from 3:00 p.m. to 5:00 p.m.

Contributed Papers

1pSC1. Validation of relative fundamental frequency using an aerodynamic estimate of vocal effort. Yu-An S. Lien (Biomedical Eng., Boston Univ., 540 Memorial Dr., Apt. 810, Cambridge, MA 02139, slien@bu.edu), Carolyn M. Michener, and Cara E. Stepp (Speech, Lang., and Hearing Sci., Boston Univ., Boston, MA)

Clinical assessment of the voice disorder vocal hyperfunction currently relies on subjective interpretations of clinicians, which can be unreliable, necessitating the development of objective measures. One relatively established objective measure of vocal hyperfunction is the ratio of sound pressure level to subglottal pressure (dB SPL/cm H_2O). However, this measure provides unreliable results in individuals with vocal fold lesions, which often accompany vocal hyperfunction. A new acoustic measure, relative fundamental frequency (RFF), defined as the normalization of voiced fundamental frequencies surrounding a voiceless consonant, has also been developed for vocal hyperfunction assessment. Here, RFF was validated against dB SPL/cm H_2O in a cohort of healthy speakers simulating various degrees of vocal effort. Twelve healthy participants (M = 22 years; five males) produced stimuli at five levels of effort (ranging from relaxed to maximally strained). The relationship between dB SPL/cm H_2O and RFF was determined. RFF parameters were able to explain 37–47% (R^2) of the variance seen in the aerodynamic measures when averaged across speakers. For six of the speakers, RFF parameters explained 43–97% of the variance, suggesting that the relationship between RFF and dB SPL/cm H_2O may depend on the strategy for modulating vocal effort employed by speakers.

1pSC2. Independent and interacting effects of sentential context and phonological neighborhood structure in spoken word production. Neal P. Fox, Megan Reilly, and Sheila E. Blumstein (Cognit., Linguistic & Psychol. Sci., Brown Univ., Box 1821, Providence, RI 02912, neal_fox@brown.edu)

Models of speech production typically invoke at least two cognitive systems: a structurally static lexicon and a dynamic, compositional system. A speaker must have access to a network of lexical representations that is vested with certain intrinsically associated information (e.g., words' phonological structure), but typical speech involves sequential, online production of words to construct meaningful sentences. Thus, a complete model of speech communication must overlay the lexicon's static architecture with a dynamic, context-driven system. The present study examines effects of these two systems on spoken words' phonetic realizations. Participants produced monosyllabic words beginning with voiceless stop consonants (e.g., *coat*) that varied in their phonological neighborhood density. Targets were produced aloud after subjects silently read sentence contexts that were either highly predictive (e.g., *She sported her stylish new fur...*) or relatively neutral (e.g., *The artist had trouble drawing the...*). Results show that subjects hyperarticulate (i.e., produce with longer voice-onset times) initial stop consonants of words with denser neighborhoods and words that appear in less predictable contexts. Interestingly, the effect of a word's context on its phonetic realization only arises for words in sparse neighborhoods, supporting a model in which the lexicon's static architecture constrains the influence of dynamic processing during production.

1pSC3. Using criterion voice familiarity to augment the accuracy of speaker identification in voice lineups. Julien Plante-Hébert and Victor J. Boucher (Laboratoire de Sci. phonétiques, Université de Montréal, Montréal, QC H3C 3J7, Canada, julienph85@hotmail.com)

Voice familiarity is a principal factor underlying the apparent superiority of human-based vs machine-based speaker identification. Our study evaluates the effects of voice familiarity on speaker identification in voice lineups by using a Familiarity Index that considers (1) recency (the time of last spoken contact), (2) duration of spoken contact, and (3) frequency of spoken contact. Three separate voice-lineups were designed each containing ten male voices with one target voice that was more or less familiar to individual listeners (13 per lineup, n = 39 listeners in all). The stimuli consisted in several verbal expressions varying in length, all of which reflected a similar dialect and the voices presented a similar speaking fundamental frequency to within one semitone. The main results showed high rates of correct target voice identification across lineups (>99%) when listeners were presented with voices that were highly familiar in terms of all three indices of recency, duration of contact, and frequency of contact. Secondary results showed that the length of the verbal stimuli had little impact on identification rates beyond a four-syllable string.

1pSC4. Domain of final lowering in spontaneous Japanese. Kikuo Maekawa (Dept. Corpus Studies, Natl. Inst. for Japanese Lang. and Linguist., 10-2 Midori-cho, Tachikawa-shi, Tokyo 190-8561, Japan, kikuo@ninjal.ac.jp)

There are two opposing predictions about the domain of final lowering (FL) in Japanese intonation. One predicts that the domain is the last mora of utterance whereas the other predicts that the domain is much wider (with no clear specification of the domain). X-JToBI annotated part of the Corpus of Spontaneous Japanese (the CSJ-Core, 44 h speech spoken by 201 speakers) was analyzed to determine the domain of FL in Tokyo Japanese. Mean normalized F0 values of the four constituent tones of accented accentual phrases (AP) were compared across utterances differing both in the number of constituting APs (from 1 to 5) and the syntactic strength of sentence- or clause-boundaries (three levels). It turned out that all tones in the last AP of utterance were considerably lowered in all utterances regardless of utterance length. This suggests strongly that the domain of FL is the last AP in Japanese. It also turned out that FL occurred in much wider contexts than hitherto believed. FL was observed not only in typical sentence boundaries but also in various syntactic clauses, and there was correlation between the degree of FL and the strength of syntactic boundaries.

1pSC5. Within-session stability of acoustic features of conversational and clear speech. Sarah H. Ferguson, Shae D. Morgan, Lydia R. Rogers (Commun. Sci. and Disord., Univ. of Utah, 390 South 1530 East, Rm. 1201, Salt Lake City, UT 84112, sarah.ferguson@hsc.utah.edu), and Eric J. Hunter (Communicative Sci. and Disord., Michigan State Univ., East Lansing, Utah)

In several planned studies, talkers will perform a set of speech tasks several times in each recording session. For example, one study will have talkers perform a set of tasks four times: in quiet, in two levels of noise, and in reverberation. It is unknown, however, whether or how much the acoustic details of speech are affected by simple repetition of the speech task. Over the course of a recording session, talkers' speech might become less careful due to fatigue, boredom, familiarity with the speech materials, or some combination of these factors. Such repetition effects could offset the effects of speaking style instructions given at the beginning of a session. The present study assessed speech acoustic changes over four repetitions of a speech production task set performed under conversational or clear speech instructions. Sixteen talkers performed three speech production tasks (a passage, a list of sentences, and a picture description) four times in each condition. The two speaking styles were recorded in separate test sessions. Several acoustic features relevant to clear speech (e.g., vowel space and speaking rate) will be compared between the first and fourth repetition in each speaking style as well as between the styles.

1pSC6. A noise robust Arabic speech recognition system based on the echo state network. Abdulrahman Alalshekmubarak and Leslie S. Smith (Computing Sci. and Mathematics, Univ. of Stirling, Stirling, Scotland FK9 4LA, United Kingdom, l.s.smith@cs.stir.ac.uk)

A major challenge in the field of automated speech recognition (ASR) lies in designing noise-resilient systems. These systems are crucial for real-world applications where high levels of noise tend to be present. We introduce a noise robust system based on a recently developed approach to training a recurrent neural network (RNN), namely, the echo state network (ESN). To evaluate the performance of the proposed system, we used our recently released public Arabic dataset that contains a total of about 10 000 examples of 20 isolated words spoken by 50 speakers. Different feature extraction methods considered in this study include mel-frequency cepstral coefficients (MFCCs), perceptual linear prediction (PLP) and RASTA- perceptual linear prediction. These extracted features were fed to the ESN and the result was compared with a baseline hidden Markov model (HMM), so that six models were compared in total. These models were trained on clean data and then tested on unseen data with different levels and types of noise. ESN models outperformed HMM models under almost all the feature extraction methods, noise levels, and noise types. The best performance was obtained by the model that combined RASTA-PLP with ESN.

1pSC7. Sub-cultural acoustical perversions and deviations: Where sensory rituals inform transcendence. Judy E. Battaglia (Loyola Marymount Univ., One LMU Dr., Los Angeles, CA 91501, judy.battaglia@lmu.edu), Roxanne A. Banuelos (Commun. Studies, Westwood College, Los Angeles, CA), Ashley L. Cordes (Loyola Marymount Univ., Northridge, CA), and Amanda McRaven (Theatre Arts, California State Univ., Northridge, Northridge, Northridge, CA)

This study investigated the perversions (to the normative culture that the subculture comes in contact with) and perceptions of sound in various subcultures, specifically in the Barona Band of Mission Indians and tribal subsets of rave culture. Ethnographic approaches lead us to understand alteration of sounds as a means to create a space of transcendence and liminality. Specifically, we studied the Barona Band of Mission Indians and their use of silence and sound swallowing as acoustics. Similarly, at various rave cultures, acoustic overstimulation was perceived as a key ingredient to achieving similar forms of collective effervescence. This subculture engaged in discourse likening acoustics to a variety of creative metaphors. Both sound isolation and the unique sense-making processes and speech codes of the subcultures were key to the study. Above all, we investigated the alteration of acoustics within a socio-spiritual context. In an increasingly postmodern and globalized world of tenuous connections and synchronous communication we argued for a need to put a focus on community-based subcultures in order to understand our perceptions of sound in the contemporary cultural moment.

1pSC8. Individual differences in the perception of fundamental frequency scaling in American English speech. Nanette Veilleux (Simmons College, 300 The Fenway, Boston, MA 02139, veilleux@simmons.edu), Jon Barnes, Alejna Brugos (Boston Univ., Boston, MA), and Stefanie Shattuck-Hufnagel (Massachusetts Inst. of Technol., Cambridge, MA)

Although most participants ($N = 62$) in an F0 scaling experiment judged open syllables (day) as higher in pitch than closed syllable tokens (dane, dave) with the same F0 contour, a subset did not. Results indicate that, in general, listeners perceptually discount F0 over coda regions when judging overall F0 level, and the degree of discount is related to the (lack of) sonority in the coda: day tokens are judged significantly higher than dane tokens which are judged significantly higher than dave tokens with the same F0 contour (dane-dave $p < 0.001$, dane-day $p < 0.01$). However, individual differences are observed: ten listeners showed no significant differences in the perception of F0 levels between the three types of tokens. On the other hand, a contrasting subset of ten subjects demonstrated highly significant differences ($p < 0.001$). The remaining 42 subjects behaved similarly to the entire subject pool with only slightly less significant differences between dane and day F0 level judgments ($p < 0.05$). Therefore, for about 16% of subjects, the F0 over the coda is not discounted in judging F0 level. These individual responses in F0 scaling perception mirror differences found in the Frequency Following Response (e.g., [1]) and could indicate individual differences in F0 processing.

1pSC9. Characteristics of speech production in a post-glossectomy speaker with a free flap: A case study. Xinhui Zhou (Elec. Eng., Univ. of Maryland, College Park, 4325 Rowalt Dr., Apt. 101, College Park, MD 20740, zxinhui2001@gmail.com), Jonghye Woo, Maureen Stone (Dept. of Neural and Pain Sci. and Orthodontics, Univ. of Maryland Dental School, Baltimore, MD), and Carol Espy-Wilson (Elec. Eng., Univ. of Maryland, College Park, College Park, MD)

It is unclear in glossectomy whether a flap will improve or impair speech, at least in the moderate sized (T2) tongue tumors. To gain some insights into this question, we studied the speech production of a post-glossectomy speaker, who had a T2 tumor surgically removed from the left side of his tongue and, closed with a radial forearm free flap (RFFF). Our acoustic analysis showed that this speaker had a significantly smaller vowel space and a significantly higher center of gravity in "sh", but not in "s", compared with the averages of normal controls or post-glossectomy speakers with primary closures. Based on cine and tagged magnetic resonance (MR) images, we analyzed the vocal tract shapes of two vowels and two fricatives and studied the tongue motion in transition of phonemes on this speaker and two controls. We will compare the vocal tract models between the flap patient and the controls. [This study was supported by NIH R01CA133015.]

1pSC10. The scope of boundary lengthening as a function of lexical stress and pitch accent. Argyro Katsika (Haskins Labs., 300 George St., New Haven, CT 06511, katsika.argyro@gmail.com), Jelena Krivokapić (Dept. of Linguist, Univ. of Michigan, Ann Arbor, MI), Christine Mooshammer (Inst. of German Lang. and Linguist, Humboldt Univ. of Berlin, Berlin, Germany), Mark Tiede (Haskins Labs., New Haven, CT), and Louis Goldstein (Dept. of Linguist, Univ. of Southern California, Los Angeles, CA)

Although the phenomenon of boundary lengthening is well established, the scope of the effect and its interaction with prominence is not well understood. It is known that phrase-final prominence is a determining factor. However, it is unclear whether it is lexical stress or pitch accent that drives the effect, and whether the affected domain is continuous or discontinuous. An electromagnetic articulometer (EMA) study of five speakers of Greek was conducted to examine the effect of (1) boundary (word and IP), (2) stress (ultima, penult, or antepenult), and (3) prominence (accented and de-accented) on the duration of phrase-final word articulatory events. In both accented and de-accented conditions, lengthening affected events that immediately preceded the boundary in stress-final words, but was initiated earlier in words with non-final stress. The affected domain was continuous. The stress effect could also be observed in pausing behavior: pauses following phrase-final words were realized with specific vocal tract configurations, and the articulatory movements forming them were longer when stress was final than when non-final. Based on these results, a theoretical account of

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boundaries is proposed within the Articulatory Phonology framework, with implications for a cross-linguistic model of prosody. [Work supported by NIH.]

1pSC11. Secondary gestures for consonants: An electromagnetic articulometer study of troughs. Hosung Nam, Christine H. Shadle, Mark Tiede (Haskins Labs., 300 George St. Ste. 901, New Haven, CT 06511, nam@haskins.yale.edu), Elliot Saltzman (Physical Therapy and Athletic Training, Boston Univ., Boston, MA), and Douglas H. Whalen (Speech-Language-Hearing Sci., City Univ. of New York, New York, NY)

Troughs, defined as a discontinuity in anticipatory coarticulation (Perkell 1968), have been shown to occur in the tongue body (TB) in /ipi/ and lips in /usu/. In V_1CV_1 contexts articulators retract from their vowel target during C when not active during C. We investigate the trough phenomenon by testing three hypotheses: (1) high intraoral air pressure during the consonant pushes TB down; (2) absence of vowel articulator activation during C in VCV causes TB to move to the neutral position; (3) the trough is a gesture in itself. Electromagnetic articulometer (EMA) data of the tongue, lips, and jaw were tracked for a trained phonetician producing non-words and real words, varying vowel context and the place, manner, and consonant length. For H1, /iCi/ productions with C={/p,f,m/} and three C durations were studied. For H2, /h/ and /ʔ/ in /iCi/ were examined. Troughs were found in all the labials but not in /h, ʔ/, disproving hypothesis 2; trough magnitude decreased in the order: /f/ > /p/ > /m/, disproving hypothesis 1. Trough magnitude increased with C duration, consistent with H3. Results from asymmetric vowel contexts also support H3, and will be discussed within the context of Articulatory Phonology. [Work supported by NIH.]

1pSC12. Principal component analysis of formant transitions. Jamie Wennblom-Byrne, Peter J. Watson (Speech-Language-Hearing Sci., Univ. of Minnesota—Twin Cities, 164 Pillsbury Dr., Shevlin 115, Minneapolis, MN 55455, wennb012@umn.edu), He Huang, and Erwin Montgomery (Greenville Neuromodulation Ctr., Greenville, PA)

Formant transitions represent dynamic articulation processes and are useful to describe differences of articulation (e.g., normal and disordered speakers). Our method, considers the formant frequency change over time as a vector. We first reduce this vector to a single value which is plotted multi-dimensional space. Each dimension represents a data point in time and the position of the point of each dimension is the value of the formant frequency. Once the vector has been converted to a single point in the multi-dimensional space, principal component analysis is applied to reduce the dimensionality. These dimensions can then be applied to multi-variant statistical analysis. Additionally, the eigenvectors in each principal component can be used to plot the differences between sets of formant transitions through time. The eigenvectors are first weighted by calculating the Pythagorean distance in the multidimensional space of the first set of principle components and then each eigenvector is multiplied by the percentage of the variance explained by that principle component. Then, the eigenvectors are plotted and superimposed over the formant transitions. The change of height along the length of the curve represents areas along the transition of maximum and minimum differences between sets of transitions.

1pSC13. Phonetics exercises using the Alvin experiment-control software. James Hillenbrand (Western Michigan Univ., 1903 W Michigan Ave., Kalamazoo, MI 49008, james.hillenbrand@wmich.edu)

A collection of computer exercises was developed for use in teaching phonetic transcription to students taking introductory coursework in phonetics. The exercises were developed using Alvin, a software package for the design and online control of behavioral experiments [Hillenbrand and Gayvert, J. Speech, Lang., Hearing Res. **48**, 45–60 (2005)]. The main goal was to allow students to work independently on the routine drill that is needed to learn sound-symbol associations. For example, for a vowel transcription exercise, students hear naturally spoken /hVd/ syllables and are asked to choose among twelve buttons labeled with phonetic symbols. Feedback is provided on each trial, and students have the option of repeating any trials with incorrect responses. Also included are word/phrase transcription exercises in which students hear an utterance and are asked to provide a phonetic transcription. Correct transcriptions are provided following each trial, and a summary of the student's performance is displayed at the end of the exercise. Reverse transcription exercises are also included in which a phonetic transcription is displayed and the student's job is to enter the word or phrase in ordinary orthography.

1pSC14. Determining functional units of tongue motion from magnetic resonance imaging. Jonghye Woo (Dept. of Neural and Pain Sci., Univ. of Maryland, Baltimore, 650 W. Baltimore St., 8 South, Baltimore, MD 21201, jschant@gmail.com), Fangxu Xing (Dept. of Elec. and Comput. Eng., Johns Hopkins Univ., Baltimore, MD), Maureen Stone (Dept. of Neural and Pain Sci., Univ. of Maryland, Baltimore, Baltimore, MD), and Jerry L. Prince (Dept. of Elec. and Comput. Eng., Johns Hopkins Univ., Baltimore, MD)

The human tongue produces oromotor behaviors such as speaking, swallowing, and breathing, which are executed by deforming local functional units using complex muscular array. Therefore, identifying functional units and understanding the mechanisms of coupling among them in relation to the underlying anatomical structures can aid significantly in the understanding of normal motor control and the development of clinical diagnoses and surgical procedures. Magnetic resonance imaging (MRI) has been widely used to observe detailed structures in the vocal tract and to measure internal tissue motion of the tongue. This work aims at determining the functional units from tagged MRI and muscle maps are extracted from high-resolution MRI. A non-negative matrix factorization method with a sparsity constraint is utilized to extract an activation map for each tissue point using a set of motion quantities extracted from tagged MRI including information from point trajectories (i.e., displacement, angle, and curvature) and strain. The activation map is then used to determine the coherent region using spectral clustering, revealing functional units and their relations to the underlying muscles. We test our algorithm on simple protrusion and speech tasks, demonstrating that the proposed algorithm can determine the correlated patterns of the tissue point tracking trajectories and strain.

1pSC15. Towards the computation of a variability index in orofacial muscle activity patterns during speech production using performance verification studies. Shonda Bernadin (Elec. and Comput. Eng., Florida A&M Univ.-Florida State Univ. College of Eng., 2525 Pottsdamer St., Tallahassee, FL 32310, bernadin@eng.fsu.edu), Megan Macpherson (Commun. Disord., Florida State Univ., Tallahassee, FL), Itiel Agramonte, and Tejal Udhan (Elec. and Comput. Eng., Florida A&M Univ.-Florida State Univ. College of Eng., Tallahassee, FL)

In previous work (Wohlert and Smith, 2002), it was determined that variability in children's speech production is reflected in upper lip muscle activity using electromyographic (EMG) data across repetitions of a phrase. Later studies (MacPherson and Smith, 2013) showed that a lip aperture variability index, which represents the difference in upper lip displacement and lower lip displacement, can also be used as a reliable variability measure from EMG data to determine the effects of multiple repetitions of the same utterance on speech motor production. This study is an extension of previous work and examines the orofacial muscle activity patterns (i.e., EMG data) during speech production in efforts to quantify EMG variability. This information can yield significant insights into how well the speech motor system is functioning in different groups of speakers (e.g., healthy young adults vs. healthy older adults). Initially for this study, kinematic measures of articulatory variability in healthy young adults and healthy older adults were verified using previously developed customized, interactive MATLAB programming. The analysis provided the platform for studying the quantification of variability in EMG data. The results of these performance verification analyses are described in this paper.

1pSC16. Tracking four dimensional muscle mechanics from high-resolution and tagged magnetic resonance imaging. Fangxu Xing (Elec. and Comput. Eng., Johns Hopkins Univ., 3400 N Charles St., Baltimore, MD 21218, fxing1@jhu.edu), Jonghye Woo, Joseph K. Ziemba, Maureen Stone (Neural and Pain Sci., Univ. of Maryland, Baltimore, MD), and Jerry L. Prince (Elec. and Comput. Eng., Johns Hopkins Univ., Baltimore, MD)

Assessment of tongue muscle mechanics during speech helps interpret clinical observations and provides data that can predict optimal surgical outcomes. Magnetic resonance imaging (MRI) is a non-invasive method for imaging the tongue that provides information about anatomy and motion. In this work, we aim to develop a pipeline to track 4D (3D space with time) muscle mechanics in order to measure motion similarities and differences in normal and glossectomy speakers. The pipeline comprises of several modules including super-resolution volume reconstruction of high-resolution MRI (hMRI) and cine-MRI, deformable registration of hMRI with cine-MRI to establish muscle correspondences, tracking tissue points using incompressible motion estimation algorithm (IDEA) from tagged MRI, followed by calculation of muscle mechanics including displacement, rotation, elongation, etc. IDEA is the 3D motion estimated from harmonic phase (HARP) motion that is obtained from tagged MRI. The proposed pipeline was evaluated on five subjects including both normal and glossectomy speakers, yielding accurate tracking results as visually assessed. In addition, we were able to differentiate normal and abnormal muscle mechanics, potentially providing invaluable information for interpreting clinical observations and predicting surgical outcomes.

1pSC17. Accelerometric correlates of nasalized speech in children. Lenny A. Varghese (Ctr. for Computational Neurosci. and Neural Technol., Boston Univ., 677 Beacon St., Boston, MA 02215, lennyv@bu.edu), Joseph O. Mendoza (Dept. of Biomedical Eng., Boston Univ., Boston, MA), Maia N. Braden (Dept. of Surgery, American Family Children's Hospital, Madison, WI), and Cara E. Stepp (Dept. of Speech, Lang., and Hearing Sci., Boston Univ., Boston, MA)

Instrumentation-based scoring methods can be used to supplement auditory assessments, providing a more objective assessment of voice quality. One such metric that can be used to assess resonance disorders, the Horii Oral Nasal Coupling Index (HONC), has been shown to successfully separate nasal from nonnasal utterances, but has not been extensively studied in children. We have previously found that using low-pass filtered versions of the nasal accelerometer and microphone signal used in its computation can reduce variability due to vowel placement that is traditionally reported for these scores, and sought to determine whether this reduction in variability would extend to children's speech. We obtained nasal acceleration and speech signals from 26 children, aged 4–9, during the production of various consonant-vowel-consonant tokens and running speech with controlled vowel and consonant loading. HONC scores were compared using broadband and low-frequency portions of these signals. It was found that applying a low-frequency filter reduced the variability of the HONC scores due to vowel type relative to when broadband signals were used, and the scores could discriminate vowels produced in nasal and non-nasal contexts in children with high accuracy. These results demonstrate the potential of HONC scores as an aid in rehabilitating hypernasal speech.

1pSC18. Tense/lax discrimination in articulatory vowel space: Evidence from electromagnetic articulography. Sonya Mehta, Jun Wang, and William F. Katz (Commun. Sci. and Disord., Univ. of Texas at Dallas, 1966 Inwood Rd., Dallas, TX 75235, naya@utdallas.edu)

A kinematic experiment investigated tongue movement for vowels in American English with the aim of developing spatial targets for use by subjects in real-time interactive learning/remediation environments. A 3D EMA system (AG501, Carstens Medezinelektronic, GmbH) was used to track the position of the tongue (tongue tip, TT, and tongue back, TB) and lips (upper lip, UL, and lower lip, LL) of adult subjects producing the vowels of American English (12 monophthongs and 3 diphthongs) in the consonant environment /bVb/. Here, we asked which of the tongue sensors yields the best spatial separation of the corner vowels /i/, /u/, /æ/ and /ɑ/, and whether vowels that are acoustically close in formant space (e.g., the pairs /i/-/I/ and /e/-/ɛ/) are distinguishable by tongue sensor position. Data were recorded for ten talkers. Points taken at the vowel midpoint of each production were plotted to determine the spatial separation (Euclidean distance) between each vowel region (centroid). Preliminary results suggest TB sensors yield the most discriminable patterns for corner vowels. Tense/lax pairs show smaller differences, as predicted by acoustic theory. For most vowel contrasts examined, patterns of spatial separation appeared sufficient for real-time feedback applications.

1pSC19. Gradient prosodic boundary perception and recursion in syntactic disambiguation. Gregg A. Castellucci and Dolly Goldenberg (Linguist, Yale Univ., Dow Hall (370 Temple St.), Rm. 204, New Haven, CT, gregg.castellucci@yale.edu)

Most research examining prosodic structure has made the assumption that prosodic boundaries are categorical and discrete, and all instances of a specific category are linguistically equivalent (Snedeker and Casserly, 2010). The empirical evidence for this notion is the finding that certain within-category differences do not cause changes in attachment ambiguity (Carlson *et al.*, 2001, experiment 2). However, several studies have shown that prosodic boundaries which arguably instantiate the same category are not only produced differently (Ladd, 1988, 2008; Krivokapić and Byrd, 2012), but listeners are also sensitive to variations in production within categories (Wagner and Crivellaro, 2010; Krivokapić and Byrd, 2012). Such findings suggest that, contrary to traditional assumptions, a certain degree of recursion may exist in prosodic structure. Our study further examines this controversial concept, as we test whether listeners are able to differentiate various meanings of coordinative structures using gradient strengths of a single boundary type while holding all other variables constant. If prosodic structure is non-recursive and phonetic gradiency within a certain boundary category is not linguistically meaningful listeners should not be able to perform this task. However, listeners in this study were able to successfully differentiate coordinative structures based on within-category boundary strength alone.

1pSC20. The path from trilling to frication: Synthesizing Puerto Rican Spanish velar fricative realizations of apical trills. Mairym Llorens (Linguist, Univ. of Southern California, 2130 Park Grove Ave., Los Angeles, CA 90007, llorensm@usc.edu)

Puerto Rican Spanish (PRS) speakers variably produce velar fricative realizations of etymological apical trills. One way to understand this alternation is by attributing it to the two types of trill realization having different gestural goals in PRS. Alternatively, both types of realization may be the result of invariant gestural goals that give rise to distinct attractor states. This study tests whether invariant articulator tasks specified for PRS apical trills can result in successfully synthesized trill and fricative realizations. Comparative measures present in the acoustics of tokens of both types of realization as recorded by native PRS speakers serve as landmarks to determine the success of the articulator model used to produce the synthesized versions. Manipulation of tongue tip gestural stiffness on the one hand and constriction degree on the other provide a way to test which parameter values best capture a bi-goal articulator task. While these findings do not argue against the possibility that PRS velar fricative realizations arise from gestural goals that differ from their apical trill counterparts, they show that this possibility is not the only candidate account for the alternation exhibited in PRS.

1pSC21. Automatic analysis of emotional intonation in Brazilian Portuguese. Waldemar F. Netto, Daniel O. Peres, Marcus M. Martins (Letras Clóssicas e Vernóculas, Univ. of Sao Paulo, Rua Etiópia, 81, Sao Paulo, Sao Paulo 03122020, Brazil, danielperes@usp.br), and Maressa F. Vieira (Letras, Faculdade Sudoeste Paulista, Avaré, Brazil)

This study analyzes emotional speech (anger, sadness, and neutral) in Brazilian Portuguese by testing 13 acoustical parameters, which were automatically processed by the software ExProsodia. The data analyzed consisted of excerpts of spontaneous speech of male and female subjects which were collected on Internet. The software selected Intonation Units (IU) based on fundamental frequency (FO), duration, and intensity information. A pilot test was carried out and its result showed that five out of 13 parameters analyzed were statically significant: skewness of fundamental frequency—sF0; coefficient of variation of fundamental frequency—cvF0; difference between medium tone and IU value—dMTIU; standard deviation of intonation unit—sdIU; and coefficient of variation of positive focus/emphasis values (above MT)—cvposF/E. A two-way ANOVA test showed a significant result. A Tukey's HSD test pointed out to differences between anger and other emotions in both genres. A significant difference between genres was found only in sadness. A Cluster test was performed based on the acoustic parameters taken into consideration and showed that only anger (male and female) could be described based on these parameters. Regarding gender differences, only sadness and neutral speech could be clustered. The results indicate that the automatic analysis done by ExProsodia can be a reliable way to analyze emotional speech produced by male and female subjects.

1pSC22. Development of a parametric basis for vocal tract area function representation from a large speech production database. Adam Lammert and Shrikanth Narayanan (Signal Anal. & Interpretation Lab., Univ. of Southern California, 3740 McClintock Ave., Rm. 400, Los Angeles, CA 90089, lammert@usc.edu)

Parametric representations of the vocal tract area function have long been of interest in speech production research because of their close relationship with speech acoustics. Many representations are possible depending on which qualities are considered desirable. For instance, several representations have been developed which trade articulatory interpretability for low-dimensionality and other useful mathematical properties, such as orthogonality. It has not been well-established whether these abstract representations are related to inherent constraints on vocal tract deformation (e.g., articulatory subspaces), or whether these representations can achieve high accuracy with a small number of parameters when applied to area functions from large amounts of continuous speech. The present study approaches these issues by extracting a data-driven basis representation for vocal tract area functions, and subsequently demonstrating the relationship between that basis and previously proposed spatial Fourier bases. Analysis was performed on more than a quarter-million individual area functions extracted from the USC-TIMIT database using region-of-interest analysis. Results suggest that a spatial Fourier basis with only four harmonics provides over 90% accuracy, and that one additional basis function corresponding to labial aperture increases representation accuracy to near 100%.

1pSC23. Misparsing coarticulation. John Kingston (Linguist, Univ. of Massachusetts, 150 Hicks Way, 226 South College, Amherst, MA 01003-9274, jkingston@linguist.umass.edu), Alexandra Jesse (Psych., Univ. of Massachusetts, Amherst, MA), Amanda Rysling (Linguist, Univ. of Massachusetts, Amherst, MA), and Robert Moura (Psych., Univ. of Massachusetts, Amherst, MA)

Considerable evidence shows that listeners often successfully compensate for coarticulation, and parse the speech signal's acoustic properties into their articulatory sources. Our experiments show pervasive misparsing of the acoustic effects of anticipatory coarticulation. Listeners categorized more of a front-back vowel continuum as back before [p] than [t]. A following [p] causes F2 and F3 to fall at the end of preceding vowels, and these listeners treated these falls as information that the vowel was back because F2 and F3 are also lower in back vowels. In experiment 1, misparsing was more extensive in a lax front-back continuum [ɛ-ʌ] than in a tense one [e-o], but the lax vowels were also shorter. In experiment 2, the durations of lax and tense vowels were equalized, and the steady-state:transition ratios were varied from the naturally occurring 70:30 to 50:50 and 30:70. Listeners still misparsed consonantal formant transitions more for lax than tense vowels, likely because the acoustic differences between [ɛ] and [ʌ] are smaller than those between [e] and [o]. They also misparsed more as transitions lengthened relative to the steady-state, which suggests that they did not adjust for differences in speaking rate that would affect the steady-state:transition ratios.

1pSC24. Nonspeech-nonspeech auditory contrast. Tracy Krug and John Kingston (Linguist Dept., Univ. of Massachusetts, Amherst, MA 01003-9274, tkrug@umass.edu)

Many perceptual adjustments for a target speech sound's context have two explanations: listeners could compensate for coarticulation or perceive the target as contrasting auditorily with its context. Only auditory contrast can explain listeners' similar adjustments for non-speech contexts that acoustically resemble the original speech contexts. This experiment tests auditory contrast further by replacing the target sound with non-speech, too. Stimuli consisted of a sequence of two equal ERB fraction spaced tone complexes separated by a short gap. Either the same higher or lower 1 ERB-wide band of tones was amplified in both complexes, creating HH and LL stimuli, or a band 1 ERB higher or lower, creating LH and HL stimuli. Tones were amplified to create HL, LH, HH, and LL stimuli near lower, intermediate, and higher frequencies (1800, 2300, and 2800 Hz). If frequency change within a stimulus evokes auditory contrast, then HL-LH stimulus pairs should be more discriminable than HH-LL pairs, where frequency only changes between stimuli. Listeners discriminated HL-LH stimulus pairs better than HH-LL pairs in all three frequency ranges, as predicted if the complexes contrast within but not between stimuli. They also discriminated both kinds of pairs worse at higher than lower frequencies.

1pSC25. Machine classification versus human judgment of misarticulated pediatric speech. Madhavi Ratnagiri, Kyoko Nagao (Ctr. for Pediatric Auditory and Speech Sci., Alfred I. duPont Hospital for Children, Wilmington, DE), Sara Krieger (Dept. of Communicative Disord., West Chester Univ., West Chester, PA), Linda Vallino, and H. Timothy Bunnell (Ctr. for Pediatric Auditory and Speech Sci., Alfred I. duPont Hospital for Children, 1701 Rockland Rd, Wilmington, DE 19807, bunnell@asel.udel.edu)

In this study, we compared machine classification to expert human judgments of segments produced by children with speech sound disorders and typically developing children. Stimuli were presented for classification in a five-alternative forced-choice paradigm with the target segment presented either in isolation or in word context. The phonemes we were mainly interested in were /s/ and /ʃ/, but instances of /θ/, /t/ and /f/ were also included among the stimuli since they could be confused with some error productions. Experienced speech language pathologists identified the target phonemes in both context conditions and their judgments were compared to the results of an HMM classifier that was trained on the speech of typically developing children. For segments presented in context, percentage agreement between the machine classifier and the consensus human classification was 95.83%. For segments presented in isolation, agreement dropped to 87.29%. Thus, the machine classifier responded more like human listeners responding to segments in word context rather than in isolation. An analysis of the differences between specific stimuli responsible for the lower agreement with isolated segments will be presented.

1pSC26. Using lateral sensors in flesh-point tracking of /l/ and /ɹ/ in American English. William F. Katz, Jun Wang, and Sonya Mehta (Commun. Sci. and Disord., Univ. of Texas at Dallas, 1966 Inwood Rd., Dallas, TX 75235, wkatz@utdallas.edu)

Whereas electromagnetic articulography (EMA) studies commonly use a midsagittal sensor array to record articulatory patterns, higher spatial imaging data (e.g., MRI, ultrasound) suggest that some sounds, such as /ɹ/ and /l/, involve tongue concave/convex shape differences that are more effectively measured along a coronal axis. We therefore explored the use of a lateral sensor in the EMA measurement of liquid consonants. Ten adult subjects produced /l/ and /ɹ/ in /ɑCɑ/, /iCi/, and /uCu/ syllables while seated in a 3D electromagnetic articulography system (AG501, Carstens Medezinelektronic, GmbH). Speech movement was tracked for tongue sensors (tongue tip, TT, tongue lateral, TL, and tongue body, TB) and lips (upper lip, UL, and lower lip, LL). Preliminary results suggest that the TL sensor, taken together with TT and TB, provide an improved characterization of American English liquid consonants. Further results will be discussed in the context of developing methods to optimize real-time speech training and speech rehabilitation systems.

1pSC27. PRAATR: An architecture for controlling the phonetics software "PRAAT" with the R programming language. Aaron L. Albin (Dept. of Linguist, Indiana Univ., Memorial Hall 322, 1021 E 3rd St, Bloomington, IN 47405-7005, aaalbin@indiana.edu)

An increasing number of researchers are using the R programming language (http://www.r-project.org/) for the visualization and statistical modeling of acoustic-phonetic data. However, R's digital signal processing capabilities are still limited compared to free-standing phonetics software such as PRAAT(http://www.fon.hum.uva.nl/praat/). As such, it is typical to extract the acoustic measurements in PRAAT, export the data to a textfile, and then import this file into R for analysis. This process of manually shuttling data from one program to the other slows down and complicates the analysis workflow. The present work reports on a software architecture ("PRAATR")

designed to overcome this inefficiency. Each of its R functions sends a shell command to the operating system that invokes the command-line form of PRAAT with an associated PRAAT script that imports a file, applies a PRAAT-command to it, and then either brings the output directly into R or exports the output as a textfile. Since all arguments are passed from R to PRAAT, the full functionality of the original PRAAT command is available inside R, making it possible to conduct the entire analysis within a single environment. Moreover, with the combined power of these two programs, many new analyses become possible. Further information on PRAATR can be found at the author's website.

1pSC28. Phonetic realization of Japanese vowel devoicing. Kuniko Nielsen (Linguist, Oakland Univ., 320 O'Dowd Hall, Rochester, MI 48309-4401, nielsen@oakland.edu)

Despite the widely used term "vowel devoicing", previous studies have reported that the phonetic realization of Japanese devoicing ranges from devoiced vowel to complete vowel deletion, and is often described as deleted or reduced as opposed to "devoiced" (Beckman, 1982; Keating and Huffman, 1984; Tsuchida, 1997; Kondo, 1997; Maekawa and Kikuchi, 2005, Ogasawara and Warner, 2009). The current study presents an acoustic investigation of phonetic implementation of Japanese vowel devoicing to examine the relative likelihood of vowel devoicing, reduction, and deletion. Twenty-four native speakers of Tokyo Japanese produced 80 words containing high vowels in single devoicing environments. The data revealed that the majority of vowel devoicing tokens (>95%) were phonetically realized as vowel deletions, with no trace of devoiced vowels (= energy excited at frequencies of vowel formants with an aspiration source). Given this result, vowel "deletion" may better characterize the phonetic implementation of Japanese devoicing. Theoretical implications of this finding for Japanese phonological structure will be discussed, including an account that considers mora as the fundamental phonological unit at which devoicing takes place and subsequently views devoicing as a loss of mora sonority.

1pSC29. Gestural coordination of the velum in singing can be different from coordination in speech. Reed Blaylock, Adam Lammert, Louis Goldstein, and Shrikanth Narayanan (Univ. of Southern California, 1150 W 29th St. Apt. 4, Los Angeles, CA 90007, reed.blaylock@gmail.com)

Professional singers are trained to maximize vowel duration and minimize consonant interference, while still maintaining intelligibility. The mechanisms by which they do this, however, are unclear. A deeper understanding of the gestural mechanisms utilized during professional-quality singing could be useful to train singers more effectively. It has been well-established that nasal sounds in syllable coda position have longer and larger velum motion in formal speech as compared to casual speech. The hypothesis of this study was that coda velum gestures would be temporally extended proportional to the duration of the entire syllable. To test this, real-time magnetic resonance (rtMR) images of trained soprano singers were used to track the motion of the velum. The boundaries of the vocal tract were mapped to a semi-polar grid, and the velocity of the velum at every frame of its movement was calculated. An important result shows that operatic singing has minimal change in velum gesture duration compared to speech, while a musical theater style can have velum lowering for the duration of a word. Observed gestural kinematics and their implications for speech are discussed from within the framework and perspective of Articulatory Phonology.

1pSC30. A comparison study of emotional speech articulations using the principal component analysis method. Sungbok Lee, Jangwon Kim, and Shrikanth Narayanan (Elec. Eng., Univ. of Southern California, 3601 Watt Way, GFS-301, Los Angeles, CA 90089, sungbokl@usc.edu)

In this study, we will investigate differences in the tongue movements across emotions by utilizing the principal component analysis (PCA), which enables to detect major and minor variations in the entire tongue surface movements under different speaking conditions. For the purpose of the study, we analyze an acted emotional speech production corpus collected from one actor and two actresses using an electromagnetic articulography (EMA). Discrete emotion types considered in this study are anger, sadness, and happiness as well as neutral one as reference. Specifically, we will investigate the number of principal components that are needed to capture emotional variations, and the differences in tongue shaping across different emotion types. Outcome of the study would provide supplementary information to the previous PCA-based production studies which have mainly focused on normal, or neutral, speech articulation. Major principal components that are found in the study also can be utilized as a basis by which effective but compact articulatory correlates (i.e., component scores) can be derived for a further investigation of emotional speech production such as an interaction between articulatory kinematics and prosodic modulations of pitch and loudness patterns, which is another important motivation of the study.

1pSC31. The effect of contextual mismatches on lexical activation of phonetic variants. Kevin B. McGowan (Linguist, Stanford Univ., Margaret Jacks Hall, Bldg. 460, Stanford, Michigan 94305, kbmcgowan@stanford.edu) and Meghan Sumner (Linguist, Stanford Univ., Stanford, CA)

Listeners' everyday experience of speech is highly varied and much of this variation is richly informative. We routinely make accurate use of phonetic cues in fast speech, for example, that would be ambiguous if decontextualized. In an influential study, Andruski *et al.* (1994) showed that a word like "king" facilitates the recognition of a semantically-related target like "queen" better with a canonically long VOT on the initial [k] than when portions of this VOT are spliced out. These and similar findings are cited as evidence for idealized or canonical lexical representations–a surprising result given the rarity of these forms in listeners' experience. In a series of semantic priming experiments, we test the hypothesis that such findings are attributable not to a benefit for the canonical variant but to a mismatch between the manipulated variants and the contextual phonetic frame in which they are presented. The short VOT variants facilitate recognition equally robustly as their long VOT counterparts when each is presented in an appropriate phonetic frame. Our results make it difficult to argue for a canonical bias in perception but, instead, connect such findings to converging evidence from the coarticulation literature showing that coarticulation facilitates perception.

Session 1pSP

Signal Processing in Acoustics: Sonar and Underwater Acoustic Communications

Paul J. Gendron, Chair
ECE Dept., Univ. of Massachusetts Dartmouth, 285 Old Westport Rd., North Dartmouth, MA 02747

Contributed Papers

1:30

1pSP1. A derivation and discussion of a mutual information lower bound for dynamic acoustic response functions under a Gauss Markov law. Paul J. Gendron (ECE Dept., Univ. of Massachusetts Dartmouth, 285 Old Westport Rd., North Dartmouth, MA 02747, pgendron@umassd.edu)

A derivation and discussion of a mutual information lower bound for broadband acoustic channels under a Gauss-Markov assumption is provided. This lower bound is associated with dynamic broadband acoustic response functions that obey a Gauss Markov law. Considered here is the difference of the mutual information between source and receive signals given the acoustic response and the mutual information between acoustic response and receive signal given the source signal. The derivation illuminates the features of this mutual information and permits identifying two key information loss mechanisms. The first is an added self noise term proportional to signal power that is only associated with the innovation variance of the Gauss Markov response. The second term also increases with the innovation variance but is associated with the uncertainty in the state of the channel operator at the initial time of signaling. This later variance is identified as a forward-backward channel estimation error. Platform motion effects and their impact on these two loss mechanisms are discussed and the structure of an effective signal to noise ratio (eSNR) ceiling at high SNR are identified.

1:45

1pSP2. Blind deconvolution of extended duration underwater signals. Jane H. Kim and David R. Dowling (Mech. Eng., Univ. of Michigan, 2010 Lay Auto Lab, Ann Arbor, MI 48109, janehjki@umich.edu)

Synthetic time reversal (STR) is a passive blind deconvolution technique for estimating the original source signal and the source-to-array impulse responses in an unknown multipath sound channel. Previous investigations of STR involved relatively short duration chirp signals (50 ms) where the assumption of a static underwater sound channel was acceptable. However, the static-channel assumption is inadequate for longer duration underwater acoustic communication signals lasting one or more seconds. Here, wave-driven changes in the ocean's surface shape and water-column sound speed lead to significant temporal variations in the source-to-array impulse responses. This presentation describes an effort to accommodate such temporal variations by decomposing a long duration signal into smaller overlapping pieces, applying STR to each piece, and then stitching the resulting sequence of signal estimates together to blindly recover the original long-duration signal. Simulations of a one-second-duration synthetic signal propagating in a static underwater sound channel to a 16-element vertical array are used to determine how the technique's performance depends on the duration and overlap of the signal pieces in relation to the signal's bit rate and the sound channel's time-delay spread. Extension of this effort to recordings from dynamic sound channels is anticipated. [Sponsored by the Office of Naval Research.]

2:00

1pSP3. A simulation based study of the effective signal to noise ratio of broadband underwater acoustic communications between moving platforms. Shrey Joshi and Paul J. Gendron (ECE Dept., Univ. of Massachusetts Dartmouth, 285 Old Westport Rd., North Dartmouth, MA 02747, sjoshi@umassd.edu)

Characterization of the effective signal to noise ratio (eSNR) of broadband underwater acoustic environments as a function of platform motion provides a practical and useful predictive metric of the adversity of an acoustic communication channel. Presented here is an analysis of estimated and actual eSNR associated with diverse moving platform scenarios for the practical case of an acoustic response function modeled as a Gauss-Markov process. A ray model of the arrival structure of various refractive environments along with a set of canonical motion scenarios provides a space of time-varying channel operators that captures the bulk effects of platform motion. Considered here are various superpositions of elliptical and translational motion effects. It is demonstrated that the eSNR exhibits both a low SNR gap and a high SNR ceiling relative to the actual received SNR (rSNR). These results give insight into useful source power levels and provide a correspondence between coherence degradation and vehicle speed and acceleration rates. These model based results lend credence to previous observations of eSNR made from high frequency broadband underwater acoustic observations [2008 NRL Review, pp. 123–125], [J. Acoust. Soc. Am. **130**(4), (2011)].

2:15

1pSP4. Development of a high frequency underwater acoustic communication modem. Brady Paradis (Elec. Eng., UMass Dartmouth, 37 Horizon Dr., Tiverton, Rhode Island 02878, bparadis@umassd.edu), Corey Bachand (BTech Acoust. LLC, Fall River, MA), Paul Gendron, and David A. Brown (Elec. Eng., UMass Dartmouth, Dartmouth, MA)

The development of a high frequency (300 kHz) underwater acoustic communication modem is reported. The transmit and data acquisition components have been designed for QPSK waveforms, intended to be computationally simple, power efficient, and cost effective. Applicability to other types of waveforms will be discussed. The design incorporates a digital signal processor, an efficient power amplifier, an analog to digital converter, and broadband transducers. An electrical equivalent circuit of the transducer was constructed as an accurate test load for bench-top measurements. Modeled results are compared to in-water test tank measurements with broadband transducers. The system is intended to serve as a development and demonstration platform to test novel waveforms and serve as a communication system on unmanned underwater vehicles (UUVs).

2:30

1pSP5. The generalized sinusoidal frequency modulated active sonar waveform ambiguity function: Theoretical and experimental results. David A. Hague and John Buck (Elec. and Comput. Eng., Univ. of Massachusetts Dartmouth, 285 Old Westport Rd., North Dartmouth, MA 02747, david.a.hague@gmail.com)

The generalized sinusoidal FM (GSFM) waveform modifies the sinusoidal FM (SFM) waveform to use an instantaneous frequency (IF) function that resembles a FM chirp waveform and as a result of this the GSFM possesses a thumbtack ambiguity function (AF) [Hague, Buck, Asilomar (2012)]. This is a drastic improvement over the SFM, which suffers from poor range resolution as its AF contains many ambiguous peaks generated by the periodicity of the SFM's IF. Analyzing the Taylor Series expansion of the GSFM's AF near the origin proves that the GSFM achieves minimal range-Doppler coupling for single target measurements. This minimizes the error in jointly estimating target range and velocity. The GSFM's AF Peak Sidelobe Level (PSL) and Integrated Sidelobe Ratio (ISLR) are comparable to or better than that of other waveforms with a thumbtack AF. Consequently, the target masking effect experienced in multi-target environments is less severe for the GSFM than other thumbtack waveforms. Lastly, test tank experiments demonstrate that the GSFM's desirable AF properties are robust to the signal conditioning necessary for transmission on practical piezoelectric transducers. [Work supported by ONR and the SMART Scholarship Program.]

2:45

1pSP6. An effective Sine-Chirp signal for multi-parameter estimation of underwater acoustic channel. Guang Yang, Wei J. Yin (Sci. and Technol. on Underwater Acoust. Lab., Harbin Eng. Univ., Nantong St. 145, Harbin City 150001, China, edit231@163.com), Ming Li (College of Eng., Ocean Univ. of China, Harbin City, China), Rong Z. Pan, and Ling H. Zhou (Sci. and Technol. on Underwater Acoust. Lab., Harbin Eng. Univ., Harbin City, China)

An effective Sine-Chirp signal is proposed for the multi-parameter estimation of underwater acoustic channel. The Sine-Chirp signal is made up of a sine signal and a Chirp signal with a fixed time interval. The sine part is adopted to estimate the multi-path number and its corresponding Doppler shift, while the Chirp part is used to estimate the multi-path delay. Furthermore, the Chirp-based decomposition character of the fractional Fourier transform (FRFT) is applied to analyze the Chirp part of the Sine-Chirp signal for the delay estimation, which greatly improves the process efficiency of the Sine-Chirp signal. The simulation results verify the effectiveness of the Sine-Chirp signal.

3:00

1pSP7. Fuzzy statistical normalization constant false alarm rate detector for non-Rayleigh sonar data. Yanwei Xu, Chaohuan Hou, Shefeng Yan, and Jun Li (IOA CAS, Beisihuanxilu No. 21, Beijing 100191, China, xyw@mail.ioa.ac.cn)

A new CFAR detector for non-Rayleigh data based on fuzzy statistical normalization is proposed. The proposed detector carries out the detection with two stages. The first stage of the fuzzy statistical normalization CFAR Processor is background level estimation based on fuzzy statistical normalization. The second stage is signal detection based on the original data and the defuzzification normalized data. Performance comparisons are carried out to validate the superiority of the proposed CFAR detector.

MONDAY AFTERNOON, 5 MAY 2014 — PROVIDENCE 1, 5:15 P.M. TO 6:30 P.M.

Meeting of Accredited Standards Committee (ASC) S1 Acoustics

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

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

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

People interested in attending the meeting of the TAGs for ISO/TC 43 Acoustics, ISO/TC 43/SC 3, Underwater acoustics, and IEC/TC 29 Electroacoustics, take note - those meetings will be held in coniunction with the Standards Plenary meeting at 9:15 a_m. on Tuesday, 6 May 2014.

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

Note: Payment of separate fee required to attend

MONDAY AFTERNOON, 5 MAY 2014 BALLROOM D, 7:00 P.M. TO 9:00 P.M.

Session 1eID

Interdisciplinary: Tutorial Lecture on Sound Reproduction: Science in the Service of Art

Alexander U. Case, Chair
Sound Recording Technol., Univ. of Massachusetts Lowell, 35 Wilder St., Ste. 3, Lowell, MA 01854

Invited Paper

7:00

1eID1. Sound reproduction: Science in the service of art. Floyd E. Toole (Harman Int. Industries, Inc., 8500 Balboa Blvd., Northridge, CA 91329, soundnwine@sbcglobal.net)

The vast majority of music we enjoy is generated by loudspeakers of differing pedigree and propagated to our ears through spaces that can mostly be described as acoustically arbitrary. In spite of the obvious huge variations, humans have managed to not only enjoy reproduced music, but sometimes even to exhibit enthusiasm for it. Common acoustical measurements confirm the variations. Are they wrong? In double-blind subjective evaluations of loudspeakers in rooms, listeners exhibit strong and remarkably consistent opinions about the sound quality from loudspeakers. The challenge has been to identify those technical measurements that correlate with the subjective ratings. What is it that these listeners are responding to? A clue: it is not in the specification sheets of most loudspeakers, nor in a secret formula for room design. This tutorial will examine the acoustical properties of loudspeakers (the sound source), rooms (the acoustical conveyance), and listeners (the powerfully perceptive, and adaptable receptor). In some respects, our problems began when we started to make certain kinds of simplistic measurements. Two ears and a brain do not respond to complex sound fields the way an omnidirectional microphone and analyzer do. What our eyes see in these measurements is not always well-matched to what our ears hear.