ducts. The strength of ducting depends on the size of internal waves, the width of the gap between waves and the curvature of the wave front, and also on the acoustic frequency and the vertical mode number. It has been seen in numerical simulations and simplified ray theory that for a given internal wave structure and a given frequency, higher vertical modes are easier being trapped in a curved internal wave duct. Also, the number of the lowest mode trapped between curved waves increases as the frequency goes up. In this talk, a 3-D normal mode theory is employed to analyze these observed characteristics. The analysis is carried out in a cylindrical coordinates, and two types of horizontal modes are found: whispering-gallery modes and full bouncing modes. Both types of modes can be described by Bessel functions, and the asymptotic formulas can be used in some limiting cases. [Work supported by the ONR.]

11:30

4aUWc14. Detection performance modeling and measurements for convergence zone (CZ) propagation in deep water. Kevin D. Heaney, Richard L. Campbell, James J. Murray (Ocean Acoust. and Instrumentation Systems, Inc. 11006 Clara Barton Dr. Fairfax Station, VA 22039, oceansound04@yahoo.com), Gerald L. D'Spain (Scripps Inst. of Oceanogr., UCSD), and Arthur B. Baggeroer (Massacheussets Inst. of Technol., Cambridge MA)

A novel parabolic equation algorithm in the C programming language has been developed based upon the RAM model. This model (CRAM) permits modeling of the full-field sonar equation to estimate towed array performance of the detection of quiet targets in a dynamic environment with both environmental range dependence and source/receiver kinematics. During an experiment in the northern Philippine Sea in 2009, a ship towing Penn State's Five-Octave Research Array (FORA) was towed at various depths in a star pattern about the station-keeping source ship, thereby sampling the first CZ in range, depth, and azimuth. Measurements and modeling of the CZ arrivals will be compared. A simple detection processor is applied to the CZ receptions. Comparison of passive ASW performance modeling results with measurements will be made. One of the primary science issues in the statistics associated with probabilistic detection is the time between independent samples, or the sample-to-sample correlation. This will be evaluated from the data for a portion of the test where the receiver was towed in an arch-fixing the source–receiver range for several hours. [Work supported by ONR.]

11:45

4aUWc15. An investigation of the effects of rough seas and bubble injections on high frequency propagation using a parabolic equation method. Joseph M. Senne, Aijun Song (CEOE, Univ. of Delaware, Robinson Hall, Newark, DE 19716, sennejm@udel.edu), Kevin Smith (Naval Postgrad. School, Monterey, CA 93943), and Mohsen Badiey (Univ. of Delaware, Newark, DE 19716)

High frequency underwater acoustic transmissions (>10 kHz) are heavily influenced by scattering from both rough surfaces and bubbles. These interactions are recorded through the prevalence of micro multi-paths in observed data. To study these scattering effects, a rough-surface variant of the Monterey Miami parabolic equation model was combined with a hydrodynamic surface model that produces non-linear waves along with depth- and range-dependent bubble distributions. Parabolic equation setup parameters were taken from collected environmental data, while a wave-rider buoy was used for time-evolving sea surface generation. Bubble plume densities were calculated using surface white-cap distributions along with a bubble evolution scheme. Comparisons of the simulated results are made against collected acoustic data for calm and rough sea states [Work supported by ONR Code 322OA.]

THURSDAY AFTERNOON, 3 NOVEMBER 2011 — SUNSET, 1:00 TO 5:25 P.M.

Session 4pAAa

Architectural Acoustics, Noise, and Committee on Standards: Networking in Soundscapes—Establishing a Worldwide Collaboration II

Gary W. Siebein, Cochair
Dept. of Architecture, Univ. of Florida, 231 Arch, P.O. Box 115702, Gainesville, FL 32611

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

Chair's Introduction—1:00

Invited Papers

1:05

4pAAa1. Getting it together—Interdisciplinary sound environment research. Frans Mossberg (Dept. of Cultural Studies, Lund Univ., Biskopsgat 7, 22100 Lund, Sweden, fransmo@glocalnet.net)

The Sound Environment Center at Lund university is an interdisciplinary center created to coordinate research on sound and soundscape issues and is known to be the first of its kind worldwide. Ranging from acoustics to medicine, psychology, and cognitive sciences, as well as humanities like musicology and linguistics, soundscape research adresses many interdependent areas and touches upon health as well as philosophical, aesthetic, and technical issues. To get a holistic comprehension, these perspectives need to be synchronized. Therefore, the center has an interdisciplinary board and a mission to study sound environments from multidisciplinary perspectives. Focus lies on research and contact between researchers. The center has external funding for larger research collaborations on topics such as teachers voice strain and rooms acoustics, health effects of combined exposure to noise and airborne particles, cognition, and sound exposure. In addition to initiating research projects, the center arranges symposiums adressing topics such as Noise and health, Seductive Sounds, Operational Sounds, Dangerous Sounds' and Sound, Cognition and Learning. Further topics have been Sound Design, Sounds and Silence for Mental Recreation, Teachers Voice Comfort, and recently Wind Turbine Noise. The symposiums facilitates cross disciplinary contacts and discussions, many of them producing published papers and reports.

1:25

4pAAa2. Soundscape ecology: A worldwide network. Catherine Guastavino (McGill Univ., School of Information Studies and CIRMMT, 3661 Peel, H3X 1X1, Montreal, QC, Canada, catherine.guastavino@mcgill.ca) and Bryan C. Pijanowski (Purdue Univ., West Lafayette, IN 47906)

The overarching objective of our network is to bring together acousticians, cognitive psychologists, ecologists, and creative artists to integrate how they study and perceive soundscapes and use this knowledge to help shape a research agenda for the conservation of soundscapes. Many natural soundscapes are being threatened from various directions, e.g., habitat destruction, climate change, invasive species. This project aims at recording and documenting soundscapes in remote locations and identifying conceptualizations of these soundscapes across different cultures and disciplines. The network will help to (1) foster open communication between different disciplines and communities about soundscapes; (2) coordinate soundscape monitoring sites where acoustic data are being collected long-term; (3) develop a common vocabulary, long-term monitoring standards, and metadata standards for acoustic data for use by ecologists; (4) increase awareness of this new field among ecologists and social scientists; and (5) increase public awareness of the importance of their acoustic connection to nature. This project on the soundscapes of natural ecosystems is a logical complement to research underway at the European level on urban soundscapes (COST action TD0804). Together, this worldwide network of researchers will be uniquely situated to contribute decisively to a cross-cultural conservation framework for soundscapes.

1:45

4pAAa3. Mapping the landscape of soundscape research. Hill Kobayashi and Saoru Saito (World Forum for Acoust. Ecology (WFAE) P.O. Box 268, Fairfield Victoria, 3078 Australia)

In this paper, I give a personal view on what could be the landscape of soundscape research. I will describe the philosophical content of this discipline, the current state of aesthetic aspect, the challenges in technical issues and the future direction of Soundscape activities among researchers, practitioners, designer, and composers. Given that to map a landscape of this discipline with discussion is a very challenging task. It requires us to connect knowledge from different disciplines with perceptional sense. The paper aims to address recent collaboration opportunities for interdisciplinary engagement and key topics for debate.

2:05

4pAAa4. Sounds in cause: Soundscape and evolution. José Manuel Berenguer (Caos-Sonoscop. CCCB. Montalegre, 5. 08001. Barcelona. Spain)

Beyond the therm of landscape, that has often been defined as “a painting, drawing, or photograph depicting natural scenery,” the concept of soundscape should lack any aesthetic significance and be thought as a technical therm naming the whole sonic experience of animals having sense of hearing. In general, from a methodological point of view, approaches describing landscape as “a view of some natural place” does not seem to be useful, because artificial and natural are often indistinguishable. Soundscape does not need to be considered natural or artificial. For instance, it seems evident that in a city, most sources of sound objects in soundscapes are human activities; anyway, even in a city, sounds produced by non-human species can easily be found. Soundscapes are complex structures that can be considered in terms of evolution. They evolve sound sources adapt their sonic productions to sonic productions of other sound sources sharing the same environment. It happens in human and non-human soundscapes. This way of thinking underlies sounds in cause, a project that started building a database of soundscapes recorded following a strict methodology and now proposes an Internet2 Network of Laboratories and Stations for Permanent Listening to the Soundscape.

2:25

4pAAa5. On the soundscape of urban parks. J. Luis Bento Coelho and Mohammed Boubezari (CAPS, Instituto Superior Tecnico, TULisbon, Lisbon, Portugal)

Urban parks are parts of every city fabric and usually well appreciated by the citizens for providing restoration and some “quiet,” at least when compared to most other city areas. Research on the soundscape of parks in cities in Portugal and in Brazil have been conducted in order to assess what makes such areas sonically interesting. Work is being directed to the differentiation of the sound components of the overall sound environment in different types of parks and on techniques for mapping the perceived sound components. The work also aims at understanding effects of climate and culture on the perception of the soundscape. Results are presented and discussed.

2:45

4pAAa6. A livingscape approach to characterize urban historical places. F. La Malva, A. Astolfi, P. Bottalico, and V.R.M. Lo Verso (TEBE Res. Group, Polytechnic of Turin, Dept. of Energetics, Turin, Italy)

The quality of life concerned with open spaces has more and more become an essential part of urban culture. The evaluation of environmental effects as perceived by people is primarily a subjective issue, rather than being simply based on objective parameters. This paper presents an approach called livingscape, used to assess the quality of urban spaces. It consists of analyzing and correlating psychometric tools to measure the perception of environmental quality with different aspects related to the urban blight (both in architectural and environmental terms) and objective investigation of environmental quality through the measurement of acoustic, visual, thermal, and IAQ physical parameters. Livingscape data were collected in 13 key-spaces of St. Salvario, an historical district in Turin (Italy), during summer 2010 and winter 2011, selected based on an historical analysis to characterize the past and present district soundscape and subdivided in nodes, paths, and edges. The subjective environmental perceptions are delineated through the analysis of the questionnaires submitted to the users of the area, outdoor. Objective measures (acoustical, lighting, and thermal parameters) were combined to subjective responses, thus providing a more complete key-spaces characterization. The investigation aims to describe the changes in the key-spaces characterization from 19th century to nowadays.

3:05–3:15 Break

3:15

4pAAa7. Using a soundscape approach to develop an acoustic ecology plan for a city. Lisa R. Lavia (Noise Abatement Society, Ste. 2, 26 Brunswick Terrace, Brighton, England, BN3 1HJ, United Kingdom), Max Dixon (Independent Consultant, London, England, United Kingdom), Osten Axelsson (Decorum Commun.s, Passvagen 30, SE-147 53 Tumba, Sweden), and Harry Witchel (Brighton and Sussex Med. School, Falmer, Brighton, BN1 9PS, United Kingdom)

Sounding Brighton is a collaborative project exploring practical approaches toward better soundscapes focusing on soundscape issues related to health, quality of life, and restorative functions of the environment. The project provides the opportunity to raise awareness and promote communication on soundscapes among the general public, stakeholders and those involved in policy, including encouraging exploration of new ways of listening in local soundscapes, and new ways of tackling noise and improving local soundscape quality. The project is working to provide opportunities to discuss how soundscape concepts might, alongside tackling conventional noise problems, contribute to local planning and environmental improvement as part of a city wide engagement process in the city of Brighton and Hove in England in the United Kingdom. A range of environments, e.g., seafront, foreshore, historic terraces, squares, lanes, parks, and gardens, are being considered. A soundmap of the city is being developed utilizing the Swedish Soundscape-Quality Protocol (developed by Osten Axelsson, Mats E Nilsson and Birgitta Berglund); a public outreach exhibition is being developed; and a night noise intervention study is planned to explore the relationship between soundscapes and the brain, community well being, social cohesion, and the physical and mental health of individuals.

3:35

4pAAa8. The introduction of the concept of soundscape to urban design analysis. Maria Tomalova MA (46 Leinster Gardens, W2 3AT, London, the UK, m_tomalova@hotmail.com)

The objective of this paper was to provide pragmatic approach through the analysis of Covent Garden Piazza's soundscape with intention to show the way of its implementation into urban design analysis, whereas analysis of soundscape is missing part of urban design analysis. So, soundscape suppose to be a new tool for designer to create more pleasant sound ambient environments not only for users who are able to perceive urban space visually but also for those who can use another senses only like aural and tactile to identify relevant quality of urban places. In this research were employed methods of subjective evaluation and spatial analysis of soundscape (entirely qualitative research). The data were collected through exploration of the soundscape preference and level of awareness of ambient sound environment within explorative single case study and thorough observation of activities on the study site and investigation of urban form. Thus, the urban uses and activities of the location and its acoustic identification are closely linked. Recent urban analyses generally take only human and technical noises into consideration, excluding the rest of noises of sonic environment and way of their perception by human beings. Acoustic dimensions must be available for design.

3:55

4pAAa9. Urban sound design—Utopia or urgent need. Nina Hllgren (Dept. of Architecture, KTH Royal Inst. of Technol./Konstfack Univ. of Arts Crafts & Design, Box 3601, 12627 Stockholm, Sweden, nina.hallgren@konstfack.se)

Today we are facing the consequences of about 100 years of urbanization, confronting questions about quality of life in relation to efficiency and economical benefits. Hard facts are considered to be more reliable than values which are not so easily measurable. The quality of sound is one of them. Urban designers and architects are currently not fully aware of the interaction between the built outcomes of their work and the process of propagation and perception of sound. Designing houses, relations between houses, connections between places, whole neighborhoods, and cities is a serious task affecting many different aspects of life. But we can still note a recurring absence of knowledge regarding the complex relation between visual and sonic realities. This reality is in fact what surrounds the urban inhabitant for an entire lifetime. But what can or should the architects and planners do? As has been recently pointed out in a report on the subject, this professional group is lacking the tools, language, and guiding examples for being able to implement anything in reality. Implementing what? What can really be done or designed to improve the sonic environment and for what purpose?

4:15

4pAAa10. Piazza del Marchese Paolo: An architectural and soundscape design to redevelop an outdoor public space. Achille Sberna, Francesco Asdrubali (Dept. of Industrial Eng., Univ. of Perugia, Via G. Duranti 67, 06125 Perugia, Italy), and Brigitte Schulte-Fortkamp (Technische Universitaet Berlin, Germany)

This paper will report about a design procedure regarding the redevelopment of an open square located in the historical center of Città di Castello, Italy. The square has five entrances and it is surrounded by old buildings. The Public Library of the town is also located at this place. Currently, the square is used as a parking lot. The goal of the design is to redevelop this square matching the given context. For the design procedure, first, the visual and acoustical status of the place will be described. Second, binaural recordings will be carried out to measure the acoustical climate and third, soundwalks will be conducted to help to detect the soundmarks of the area. Moreover, the idea is to transform the space in a pedestrian area and to consider the square as an acoustical "outdoor floor" for the library. The design process will be focused on the preservation of the genuine Soundscape.

4:35

4pAAa11. How the soundscape approach enlightens the knowledge about the feeling of safety in urban spaces: A matter of networking and interdisciplinarity. Kay Sebastian Voigt and Brigitte Schulte-Fortkamp (Technische Universite, Inst. Fluid Mech. and Eng., Secr TA7, Einsteinufer 25, Berlin, Germany)

In a current project on dynamics of urban safety and its arrangements funded by the Federal Ministry of Education and Research, Germany http://www.dynass-projekt.de/projekt-dynass/ Dynamische Arrangements städtischer Sicherheitskultur/, the Soundscape approach is one of diverse approaches to investigate areas of different cities with respect to the perception and production of safety in such environments. As to the perception of safety, areas as well as its production special attention will be given to the reconstruction of

decisive factors. The current status of the international standardization process of the ISO/TC 43/SC 1/WG 54 and the contribution of the COST network are used here as the basis to improve procedures and measures in the applied Soundscape approach. Moreover, this paper will discuss the Soundscape research as an interdisciplinary approach and in an interdisciplinary approach with regard to investigations upon safety management in urban areas. First results give hints to the importance of the acoustic, visual, and social structures of an area in the face of the Soundscape procedure.

Contributed Papers

4:55

4pAAa12. Water features and acoustic diversity of urban parks. Osten Axelsson (Passvaegen 30, SE-147 53 Tumba, Sweden) and Mats E. Nilsson (Stockholm Univ., SE10691 Stockholm, Sweden)

Water features are well-acknowledged in architecture and urban planning for their visual characteristics. But, how do water features contribute to acoustic diversity and soundscape quality? Visitors in an urban park were recruited to complete a questionnaire on how they perceived the park including its soundscape. Meanwhile, the soundscape was manipulated by turning a fountain on or off at irregular hours. The fountain sounds had a positive effect on soundscape quality in an area close to the fountain, by masking background road-traffic noise. The fountain sound also masked other natural sounds, which may have a negative influence on acoustic diversity and soundscape quality. In addition, some participants may have mistaken the fountain sounds for distant road-traffic noise. Hence, when introducing a water feature in an urban park it is necessary to consider the acoustic characteristics of the water sounds, as well as the placement of the water feature.

5:10

4pAAa13. Researching sound in silence. Jurgen De Blonde (Aifoon vzw, Nieuwevaart 117a, B-9000 Gent, Belgium, jurgen@aifoon.org)

Aifoon is an educational arts organization that investigates sound in silence. We investigate poetic and communicative possibilities of everyday sounds. Our aim is to have people communicate about their immediate surroundings in a non-verbal way by using sounds taken from those surroundings. In our workshops, we teach people to record sounds, we teach microphone awareness, we open ears, draw sounds, and compose with these drawing, compose with sound, make a montage on a computer, and talk about the results. We avoid using music and words since these two elements are too coded. We take the results and the questions that rise from our work back to artists, researchers, and the public for further interpretation and investigation. We have taken our core philosophy as a basis for an exposition, a number of events, an ensemble, and a couple of installations in an attempt to take our educational role beyond the workshop session into the public space. This has proven successful and has often helped us to explain our ideas and outsiders to understand them.

THURSDAY AFTERNOON, 3 NOVEMBER 2011 SUNRISE, 1:25 TO 5:45 P.M.

Session 4pAAb

Architectural Acoustics: Variable Acoustics—Methods for Effective Collaboration

Roger W. Schwenke, Chair
Research and Development, Meyer Sound Laboratories, 2832 San Pablo Ave., Berkeley, CA 94702

Chair's Introduction—1:25

Invited Papers

1:30

4pAAb1. What you call it does matter: A vocabulary for active acoustics. Kurt M. Graffy (Arup Acoust., 560 Mission St., San Francisco, CA 94105, kurt.graffy@arup.com)

Active acoustics (Electronic Architecture) systems have reached a maturity in terms of their acoustic performance which makes them a viable option for multi-use venues as well as renovations of existing facilities. However, the awareness on the part of owners and facilities regarding the capabilities and potential acoustic benefits of such systems is still developing. More significantly, the corresponding level of awareness for designers and architects, which active acoustics can actually represent options for how architectural, structural, and building service designs may proceed, is even less developed. At this intersection of past techniques versus current abilities, we are in need of a common vocabulary for active acoustics; a common vocabulary to interact with the design team, a common vocabulary to assist venue operators and owners understand the benefits, and a common vocabulary to interact with the musicians and artists in tuning and configuration. The presentation references some recent projects utilizing active architecture primarily focused on the interaction with the design team and subsequently with the artists, and the development of a common language as the projects moved forward.

1:50

4pAAb2. 2006 International Music Festival. Edward Dugger III (4490 SW Long Bay Dr. Palm City, FL 34990, edward@edplusa.com)

Throughout the 3-week run of the 2006 International Music Festival held in Boca Raton, Florida, the 4000 seat Mizner Park Amphitheatre was fitted with an active acoustic system. We will present a case study that explores many of the elements of the project including the initial audio system selection, installation, rehearsals, and performances. Since this was an unusual project both in its exterior application and large scale, we will present the design features that had significant implications for the musicians, concert presenters, stage hands, and audience. At the request of the venue operators, a new design has been developed to install a permanent active acoustic system for the amphitheater. This design will also be presented.

2:10

4pAAb3. Successfully merging architectural and electronic acoustical treatments. Steve Barbar (30 Dunbarton Rd., Belmont, MA 02478, steve@lares-lexicon.com)

In enclosed volumes, the integration of electronic acoustical components with architectural surface treatments forms a hybrid system that produces the perceived acoustical conditions. Since the underlying operating principles for electro-acoustic enhancement systems differs considerably between manufacturers, the requirements for system infrastructure are not germane, nor is the optimum integration of architectural treatments. As a result, the nature of the work performed by the acoustical consultant changes to accommodate optimum performance of the specific “hybrid” system, which may also include other forms of variable treatments.

2:30

4pAAb4. Working with musicians to increase low-frequency performance of an active acoustic system in a music practice room. Ron Freiheit (Wenger Corp., 555 Park Dr. Owatonna, MN 55060, ron.freiheit@wengercorp.com)

To enhance the performance of an active acoustic system for music practice rooms, a new speaker was developed with extended low-frequency response. To better understand the performance desired from certain musicians, a small number of cellists provided observations and opinions about what environment was most pleasing to them, specifically related to the expanded low-frequency response of the active system. A system was developed that allowed the tuning of the low-frequency response to better ascertain at which point it was optimized and below or above which very little improvement was noted. Once these parameters were optimized, design criteria for the speakers were determined. Also discovered during this research were challenges musicians faced in discriminating between the direct sound and active sound field.

2:50

4pAAb5. The care and feeding of clients with variable room acoustics systems. Edward Logsdon (D. L. Adams Assoc., Inc., 1701 Boulder St., Denver, CO 80211, elogsdon@dlaa.com)

The care and feeding of clients with variable room acoustics systems. Now that the Variable Room Acoustics System (VRAS) is designed and installed in the room, what do you do? An Acoustical Consultant’s Perspective One of the primary educational goals for the Colorado College, Edith Kinney Gaylord Cornerstone Arts Center in Colorado Springs, CO, is to encourage collaboration between the music, drama, dance, film, and visual artists who perform and display their work in the facility. The building offers many opportunities for film/video and theater students, or music and sculptural artists to work together on multimedia presentations. We used “collaboration” to further encourage the student and faculty performers and artists to communicate their needs and priorities for use of the multipurpose main theatre early on in the design. Subjective opinions of how the room sounds, or how best to control the VRAS system, had to be carefully addressed and assessed to avoid confusion and to achieve unanimity. We will review our process of establishing an artistic vocabulary that allowed a jury of users to advise us on the types and number of room presets needed for the various uses of the theatre and the validation and approval of the system.

3:10–3:30 Break

3:30

4pAAb6. Various applications of active field control. Takayuki Watanabe and Masahiro Ikeda (Spacial Audio System Group, Yamaha Corp., 10-1 Nakazawa-cho, Nakaku, Hamamatsu, Japan, watanabe@beat.yamaha.co.jp)

Several types of various active field control (AFC) applications are discussed, while referring to representative projects for each application. (1) Realization of acoustics in a huge hall to classical music program, E.g., Tokyo International Forum: This venue is a multi-purpose hall with approximately 5000 seats. AFC achieves “loudness” and “reverberance” equivalent to those of a hall with 2500 seats or fewer. (2) Compensation of acoustics on stage without rigid shell using the electro-acoustic method. E.g., High school auditoriums: In these renovation projects, AFC achieves “acoustical support” for performer on stage and “uniformity” throughout the auditorium from the stage to the audience area, etc. (3) Improvement of the acoustics under the balcony in auditoria. E.g., Experiments on a full-scale model and the school auditorium. The system is a non-regenerative system, and the loudspeakers, located at positions corresponding to measurement points across the balcony, recreate the reflecting sound from above the balcony area, which otherwise fail to reach to the listeners under the balcony. The results of the experiment show that the system is significantly better for all tests to the use of no system and that the system is superior to a standard PA (delay system).

3:50

4pAAb7. Acoustic repair: Recent experience with the acoustic control system (ACS) for improving acoustic conditions in two existing venues. Timothy E. Gulsrud (Kirkegaard Assoc., 954 Pearl St., Boulder, CO, 80302 tgulsrud@kirkegaard.com) and Arthur van Maurik (Acoust. Control Systems BV, Speulderweg 31 3996 LA Garderen, The Netherlands)

Active acoustics systems are becoming more prevalent in architectural acoustics practice, particularly in the context of repairing or improving acoustics in existing venues. Governmental policies to reduce funds and subsidies put into new facilities for the performing arts are another reason for designers to consider the use of active acoustics. This paper highlights two recent examples of such installations of ACS systems, one at the Sydney Opera House Concert Hall, and the other at MBCCH, Winnepeg, Canada. Collaboration between the system designer, the musicians, and the acoustics consultant will be emphasized, along with techniques used to evaluate the systems’ performance in the halls.

4:10

4pAAb8. Indoor outdoor acoustics: Active acoustics at the New World Center, Miami Beach Soundscape. Frederick R. Vogler (Sonitus, fred@sonitusconsulting.com), John Pellowe (Meyer Sound Labs, Inc.), and Steve Ellison (Meyer Sound Labs, Inc.)

The Frank Gehry designed New World Center is home to the New World Symphony in Miami Beach, FL. This facility includes an intimate 756 seat concert hall and is used as a training platform both for symphonic conductors and musicians. The adjacent park uses an active acoustics system to allow a similar number of people in the park to simultaneously experience the indoor concert experience in an open air environment. The immersive sound of the reproduced concert experience is accompanied by a 7000-square-foot projection wall that carries live video of the performance. The system captures the natural acoustic of the concert hall using microphones distributed throughout, and these signals are processed and then transmitted to the park utilizing a set of 160 distributed loudspeakers. The successful design, commissioning, and tuning of the system relied on a team approach between the architect, consultants, manufacturer, installer, and venue operators. Scope within the team is explored, challenges revealed, and suggestions offered to help ensure the success of new multi disciplinary ventures such as this. Similarities and differences to a surround sound broadcast transmission of the Los Angeles Philharmonic are also reviewed.

Contributed Papers

4:30

4pAAb9. Acoustical design of New World Center, Miami Beach, FL. Daniel F. Beckmann, Kayo Kallas, Motoo Komoda, and Yasuhisa Toyota (Nagata Acoust., 2130 Sawtelle Bl. Ste 308, Los Angeles, CA 90025, beckmann@nagata.co.jp)

The New World Center opened in January 2011 after an 8-year process to design and build the $160 million facility. The Frank Gehry-designed facility for the New World Symphony, "America's Orchestral Academy," has at its heart the 756-seat auditorium, designed in the arena style. Also in the facility are one large orchestral rehearsal room, seven medium-sized Chamber/Ensemble rehearsal rooms, and 24 individual coaching/practice rooms. Administrative offices and production spaces complement the spaces for music to bring the building to 100 641 square feet. The acoustical design of the 756-seat auditorium was performed in close collaboration with Gehry and the founder of the New World Symphony, acclaimed conductor Michael Tilson Thomas. To ensure validity of the acoustical design, a 1:24 scale model test was performed. Flexibility in rehearsal and performance was one of the prime requirements for the space as the program includes much more than just orchestral performance, which is met by a curtain system for variable acoustics, a highly flexible stage lift system, 247 retractable seats, and four alternate "Performance Platforms." Five large "sails," which double as projection surfaces, a proper ceiling above the stage, and a steeply raked audience area are amongst the acoustical design elements that are reported.

4:45

4pAAb10. Acoustical design of Soka University Performing Arts Center. Kayo Kallas, Motoo Komoda, and Yasuhisa Toyota (Nagata Acoust. America, 2130 Sawtelle Blvd., Ste. 308, Los Angeles, CA 90025, kimotsuki@nagata.co.jp)

The Soka University Performing Arts Center and Academic Building will open in September 2011. The $73 million performing arts center is open to the public, hosting various types of performing arts. For lovers of the performing arts, the center will become another choice among the many fine venues in Orange County. The building houses a 1000-seat multipurpose hall, a 150-seat black box theater, support spaces, and classrooms. The multipurpose hall was designed primarily as a concert hall, and later to become suitable for dance, plays and musicals. To satisfy these flexible programs, the design features a curtain system for variable acoustics and an automated stage lift to accommodate concert, thrust, and convocations stage settings. The seating layout of the multipurpose hall was arranged in the arena style. The room shape and interior materials were carefully selected to optimize the acoustics of the space, including the two layered ceiling design: one for aesthetics and the other for room acoustics. Acoustical design and characteristics of the new multipurpose hall will be reported.

5:00

4pAAb11. The effect of reverberation enhancement on the diffusion of the sound field. Hugh Hopper, David Thompson, and Keith Holland (I.S.V.R., University of Southampton, University Rd., Southampton, SO17 1BJ, UK)

Reverberation enhancement is a technology which allows the reverberation time of a room to be increased. It is important to consider the effect of this technology on the other measurable attributes of the room response. The spatial variation of steady state sound pressure level and reverberation time within the room can be used to measure the extent to which the room approximates a diffuse field. A theoretical value of these quantities can be predicted for an ideal diffuse field, and the ratio between the measured and theoretical values gives a normalized measure of the diffusion of the sound field. This work investigates the changes in these measures when reverberation enhancement is applied to a room. Experimental results have shown that the normalized measures of diffusion increase with the introduction of reverberation enhancement. This implies a reduction in the homogeneity and isotropy of the sound field which may be perceived as a reduction in subjective quality.

5:15

4pAAb12. Variable acoustics at hauppauge high school. Richard F. Riedel (Riedel Audio & Acoust., LLC, 443 Potter Blvd., Brightwaters, NY 11718)

The author will discuss how variable acoustic solutions were used to solve room anomalies and provide a means to control room size. Once plagued by a significant rear wall reflection that interfered with stage musicians' timing, due to its later arrival time, the author will explain how a unique custom designed retractable acoustical diffuser reduced this problem. Another recurring issue in the space was the need to reduce the size of the seating area, normally 1100 seats, to one which provided a more intimate setting for dramatic presentations, which were normally not attended by large audiences. Using a commercially available acoustical product, the issue was solved and a means provided to alter the room's acoustical environment. This paper presents details about the methods used to provide variation to the room's acoustics and cites specific measurements of the space with and without the variable acoustical elements.

5:30

4pAAb13. Sound absorbing vertically retracting drapery: A comparative study. Liz L. Lamour (Univ. of Kansas, School of Architecture, Design and Planning, Marvin Hall, 1465 Jayhawk Blvd., Lawrence, KS 66045, lizlamour@gmail.com), Ben Bridgewater (Univ. of Kansas, Lawrence, KS 66045), and Ben Brooks (Univ. of Kansas, Lawrence, KS 66045)

The use of drapery as a variable sound absorbing material is widespread in theaters, university music halls, and other spaces where variable reverberation time is desired. As a class project at The University of Kansas, the authors and their classmates tested two types of sound absorbing vertically retracting drapery manufactured by a theatrical contracting firm using reverberant room methods. Fabric used in these two tests was cotton velour and acrylic velour. Coefficients of absorption were compared with coefficients of absorption published by a firm specializing in the fabrication of vertically retracting sound absorbing velour drapery. The measured and published data allowed both types of vertical retracting drapery to be specified for a multipurpose auditorium renovation project at a college in Kansas. The college auditorium renovation project will use the drapery fabricated by the specialty firm and the on-site determined coefficients of absorption will be presented and compared with the manufacturer's published data and also compared with the data obtained in the reverberant room for the somewhat different retracting drapery design by the theatre contracting firm. And if it proves to be possible, the on-site sound absorption measurements will be made at another college auditorium which uses cotton velour retracting drapery produced by the theatrical contracting firm. These data will be useful in specifying sound absorbing vertically retracting drapery for future projects.

THURSDAY AFTERNOON, 3 NOVEMBER 2011 PACIFIC SALON 1, 1:30 TO 3:15 P.M.

Session 4pAB

Animal Bioacoustics: Long-Term Acoustic Monitoring of Animals II

Simone Baumann-Pickering, Cochair
Scripps Inst. of Oceanography, Univ. of California, San Diego, 9500 Gilman Dr., La Jolla, CA 92093-0238

Marie A. Roch, Cochair
Dept. of Computer Science, San Diego State Univ., 5500 Campanile Dr., San Diego, CA 92182-7720

Invited Papers

1:30

4pAB1. Diel and lunar variations of marine ambient sound in the North Pacific. Simone Baumann-Pickering (Scripps Inst. of Oceanogr., Univ. of California, San Diego, 9500 Gilman Dr., La Jolla, CA 92093, sbaumann@ucsd.edu), Ana Širović (Scripps Inst. of Oceanogr., La Jolla, CA 92093), Marie A. Roch (San Diego State Univ., San Diego, CA 92182), Anne E. Simonis, Sean M. Wiggins (Scripps Inst. of Oceanogr., La Jolla, CA 92093), Erin M. Oleson (Pacific Islands Fisheries Sci. Ctr., NOAA, Honolulu, HI 96822), and John A. Hildebrand (Scripps Inst. of Oceanogr., La Jolla, CA 92093)

Marine ambient sound was recorded on autonomous high-frequency acoustic recording packages (bandwidth 10 Hz to 100 kHz) during long term deployments at multiple sites across the North Pacific, from the high latitude Aleutian Islands to tropical Palmyra Atoll in depths of 600–1 000 m. Most intertropical but no temperate locations showed a distinct diel pattern in ambient sound. The soundscape at each location was unique, yet there was a similar recurring sound of unknown origin in lower latitude locations. This sound had a peak frequency around 3–5 kHz and was recorded only for several hours after sunset. Additionally, at some locations, a broadband acoustic signal with bandwidth up to 60 kHz was recorded at night with crepuscular peaks. Both sound patterns were lunar dependent with lower acoustic levels during full moon phases. Site-specific diel and seasonal acoustic patterns have been observed for various odontocete species. Correlations between odontocete presence and levels of ambient sound are investigated. [Work supported by NOAA-Pacific Islands Fisheries Science Center, US Navy-N45/PACFLT, ONR, Pacific Life, Ocean Foundation, University of California, San Diego.]

1:50

4pAB2. Long-term passive acoustic monitoring of nearshore ecosystems in the Northwestern Hawaiian Islands. Marc O. Lammers, Lisa Munger (Hawaii Inst. of Marine Biology, P.O. Box 1346, Kaneohe, HI 96744, lammers@hawaii.edu), Pollyanna Fisher Pool (Univ. of Hawaii, Honolulu, Hawaii), Kevin Wong (Pacific Islands Fisheries Sci. Ctr., Honolulu, Hawaii), Whitlow W. L. Au (Hawaii Inst. of Marine Biology, Kaneohe, HI 96744), and Russell E. Brainard (Pacific Islands Fisheries Sci. Ctr., Honolulu, Hawaii)

Monitoring the changing state of marine habitats in remote areas is, in most cases, a challenging task due to limited and/or infrequent opportunities to make direct observations. Passive acoustic monitoring is sometimes the best means of establishing long-term biological trends in such areas. Since 2006, an effort has been underway to monitor the neashore ecosystems of the Northwestern Hawaiian Islands (NWHI) using a network of Ecological Acoustic Recorders. A wide range of acoustic signals are being monitored to infer biological trends and to gauge the relative stability of the ecosystem. Among the variables measured are the acoustic activity of snapping shrimp, the incidence of cetaceans and the extent of spectral and temporal partitioning of the acoustic space by different taxa, measured as the "acoustic entropy" of the habitat. Multiyear time series of the different measures provide baseline levels of biological activity at each location and also reveal periods of anomaly. Observed trends are then examined for corollary relationships with oceanographic and meteorological parameters measured both in situ and remotely. The data obtained thus far are providing valuable insights that will help assess the long-term response of ecosystem in the NWHI to both natural and anthropogenic factors

2:10

4pAB3. Eavesdropping on coconut rhinoceros beetles, red palm weevils, Asian longhorned beetles, and other invasive travelers. Richard W Mankin (USDA-ARS-CMAVE, 1700 SW 23rd Dr., Gainesville, FL 32608, richard.mankin@ars.usda.gov)

As global trade increases, invasive insects inflict increasing economic damage to agriculture and urban landscapes in the United States yearly, despite a sophisticated array of interception methods and quarantine programs designed to exclude their entry. Insects that are hidden inside soil, wood, or stored products are difficult to detect visually but often can be identified acoustically because they produce 3–30-ms, 200–5 000-Hz impulses that are temporally grouped or patterned together in short bursts. Detection and analysis of these sound bursts enables scouts or inspectors to determine that insects are present and sometimes to identify the presence of a particular target species. Here is discussed some of the most successful acoustic methods that have been developed to detect and monitor hidden

insect infestations. Acoustic instruments are currently available for use in rapid surveys and for long-term monitoring of infestations. They have been useful particularly for detection of termites, coconut rhinoceros beetles, red palm weevils and Asian longhorned beetles in wood, white grubs and Diaprepes root weevil in soil, and stored product insects.

Contributed Papers

2:30

4pAB4. Acoustic monitoring of dolphin populations in the Gulf of Mexico. Kaitlin E. Frasier (Marine Physical Lab., Scripps Inst. of Oceanogr., Univ. of California, San Diego, 9500 Gilman Dr. La Jolla, CA 92093, kefrasie@ucsd.edu), Melissa S. Soldevilla (Protected Resources and Biodiversity Div, NMFS/SEFSC, Miami, FL 33149), Mark A. McDonald (WhaleAcoust., Bellvue, CO 80512), Karlina P. Merkens, Sean M. Wiggins, John A. Hildebrand (Univ. of California, San Diego, La Jolla, CA 92093), and Marie A. Roch (San Diego State Univ., San Diego, CA 92182)

High-Frequency Acoustic Recording Packages (HARPs) continuously monitored delphinids at five sites in the northeastern Gulf of Mexico during and after the *Deepwater Horizon* oil spill. Surface oil reached two sites, while the three unexposed sites functioned as “controls.” Presence of dolphin vocalizations (clicks, whistles, and burst pulses) was documented at exposed and unexposed sites over the course of a year following the oil spill. These sites are within the known habitat ranges of 11 species of delphinids. Broadband towed array recordings with visual identifications were used to determine species-specific vocalization characteristics, which were then compared with autonomously recorded vocalizations. Two species have distinctive vocalizations that match between towed array and autonomous recordings. At least four more unique vocalization patterns were detected autonomously, which may be species-specific. Both clicks and whistles were explored for identifying features. The data provide a comparative view of delphinid presence relative to the oil spill.

2:45

4pAB5. Passive acoustic monitoring of sperm whales during and after the Deepwater Horizon oil spill. Karlina Merkens (Scripps Inst. of Oceanogr., UCSD, 9500 Gilman Dr. MC 0205, La Jolla, CA 92093-0205, kmerkens@ucsd.edu), Mark A. McDonald (Whale Acoust., Bellvue, CO 80512), Simone Baumann-Pickering, Kaitlin Frasier, Sean M. Wiggins, and John A. Hildebrand (Scripps Inst. of Oceanogr., UCSD, La Jolla, CA 92093-0205)

The Deepwater Horizon oil spill during the summer of 2010 impacted a region of sperm whale habitat along the continental slope and deep waters of the Gulf of Mexico. Passive acoustic monitoring was used to study the potential impact of the oil spill on sperm whales by recording trends in their characteristic sounds, such as echolocation clicks and foraging creaks. High-frequency Acoustic Recording Packages (HARPs) were deployed shortly after the oil spill began; one was located close to the Deepwater Horizon well, above which the sea surface was contaminated by oil throughout the summer of 2010, and another was deployed in a region of sperm whale habitat that remained unexposed to surface oil to function as a “control” site. At both sites, sperm whales were detected on a majority of days during the nearly year-long recording period. Sperm whale presence was evaluated from detected clicks and creaks, and changes in these sounds over time and between sites were compared.

3:00

4pAB6. Long-term acoustic monitoring of marine mammal response to the 2010 oil spill in the Northern Gulf of Mexico. Natalia Sidorovskaia (Dept. of Phys., Univ. of Louisiana at Lafayette, Lafayette, LA 70504-4210, nas@louisiana.edu), Azmy S. Ackleh, Baoling Ma (Univ. of Louisiana at Lafayette, LA 70504), Christopher Tiemann (Univ. of Texas at Austin, Austin, TX 78713), George E. Ioup, and Juliette W. Ioup (Univ. of New Orleans, New Orleans, LA 70148)

The 2010 deep water horizon (DWH) oil spill in the Northern Gulf of Mexico brought a need for assessing spill impact and recovery timeline for the deepwater ecosystem, including marine mammals. Passive acoustics is emerging as a viable technology to monitor short-term and long-term abundance dynamics and to assess different factors that may cause an observed response. Multi-year pre-spill and post-spill acoustic data collected at different distances from the DWH incident site by the Littoral Acoustic Demonstration Center (LADC) are used to compare first-year oil spill response by three different groups of marine mammals: sperm whales, beaked whales, and dolphins. Densities of acoustic phonations by these animals are extracted from collected data and used for point estimates of the resident population density. As an example, a regional abundance estimate shows a decrease in the number of sperm whales at the site nearest to the DWH (9 mi away) which exceeds statistical uncertainties and can be accepted as an existing trend. The use of acoustic data to extract information about environmental factors, such as anthropogenic noise level or food call densities, that may contribute to the explanation of existing trends is also discussed. [Work is partially supported by NSF.]

Session 4pBA

Biomedical Acoustics: Therapy and Applications

Thomas J. Matula, Chair
Applied Physics Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105-6698

Contributed Papers

1:30

4pBA1. Effects of physical properties of the skull on high intensity focused ultrasound for transcranial sonothrombolysis. Prashanth Selvaraj (Dept. of Mech. Eng., Etcheverry Hall, Univ. of California, Berkeley, CA pselvaraj@me.berkeley.edu), Kohei Okita (Ctr. for Intellectual Property Strategies RIKEN 2-1 Hirosawa, Wako, Saitama 351-0198, Japan), Yoichiro Matsumoto (Univ. of Tokyo, 7-3-1 Hongo, Bunkyo, Tokyo 113-8656, Japan), Arne Voie, Thilo Hoelscher (Univ. of California, San Diego, 212 West Dickinson St.), Hope Weiss, and Andrew J. Szeri (Etcheverry Hall, Univ. of California, Berkeley, CA)

The use of high intensity focused ultrasound (HIFU) in transcranial sonothrombolysis is emerging as a promising therapeutic intervention after stroke. Of interest in the present study is the evolution of the wave from transducer to focus, with special attention to two aspects. One is the attenuation of the wave before it reaches the focus, the other is the scattering of the wave at tissue interfaces leading to alteration of the focus. A code developed for tissue ablation (Kohei Okita, Kenji Ono, Shu Takagi, and Yoichiro Matsumoto, Int. J. Numer. Methods Fluids 65:43–66 (2011)), has been modified to study the effect of the physical properties of the skull on the focusing of the HIFU waves. Phase delay of the array transducer is employed to focus the waves. A basic model illustrative of the calvaria of the skull has been used as only the physical properties of the bone are of interest here. Microbubble cavitation has been shown to enhance sonothrombolysis; hence, the altered wave is examined from the point of view of the bubble dynamics it engenders.

1:45

4pBA2. Vascular permeability with targeted contrast agents—The effect of physiologically relevant dynamic shear stress. Pavlos Anastasiadis (Dept. of Mech. Eng., Univ. of Hawaii at Manoa, 2540 Dole St., Holmes Hall 302, HI 96822, pavlos@hawaii.edu), Joshua J. Rychak (Targeson, 3550 General Atomics Court, San Diego, CA 92121), and John S. Allen III (Dept. of Mech. Eng., Univ. of Hawaii at Manoa, HI 96822)

Targeted ultrasound contrast agents (UCAs) may be able to facilitate an early noninvasive diagnosis of atherogenesis. The coronary arterial branches might be routinely scanned for a clinical diagnosis since plaques often first form at disturbed flow regions within bifurcations. However, many outstanding questions exist on the targeting efficacy subject to pulsatile and potentially pathological flow in these areas. Targeted ultrasound contrast agents conjugated to the antibody of the intercellular adhesion molecule-1 (ICAM-1) are injected into y-shaped flow chambers that model the coronary heart artery bifurcations. Simultaneous measurements of the barrier function in physiologically relevant dynamic shear stresses are monitored with electric cell impedance (ECIS) measurements. The study aims at quantifying variations of the barrier function at regions with exposure to different shear stresses and correlating this information to the binding efficacy of targeted UCAs conjugated to the antibody of ICAM-1. These results are discussed with respect to reported targeted ultrasound contrast agent studies of inflammation and plaque in Apolipoprotein E-deficient mice.

2:00

4pBA3. Synergistic interaction between stress waves and cavitation is important for successful comminution of residual stone fragments in shock wave lithotripsy. Jaclyn Lautz, Georgy Sankin, and Pei Zhong (Dept. of Mech. Eng. and Mater. Sci., Duke Univ., Durham, NC 27708, jaclyn.lautz@duke.edu)

To assess the role of stress waves and cavitation in comminuting residual fragments during shock wave lithotripsy (SWL), cylindrical 4×4 mm BegoStone phantoms were treated in an electromagnetic lithotripter either at the focus ($z = 0$, $p_+ = \sim 45$ MPa) or pre-focally ($z = -30$ mm, $p_+ = \sim 24$ MPa). The treatment was performed with the stone immersed either in degassed water or in Butanediol, which has similar acoustic impedance to water but much higher viscosity to suppress cavitation. At the focus, the first fracture was observed after 26 ± 9 shocks, both in water and Butanediol ($p = 0.7$). However, when stones were moved pre-focally where comparable cavitation is produced (based on high speed imaging), the average shock number required for the initial fracture was increased to 66 ± 10 in water and 122 ± 20 in Butanediol ($p = 0.002$). Below -40 mm prefocally ($p_+ < 20$ MPa), stones did not fracture in water even after 2,000 shocks, although cavitation was observed. Furthermore, stone comminution at the focus after 250 shocks was $\sim$35% in water compared to $\sim$5% in Butanediol ($p < 0.001$). Altogether, these findings suggest that a synergistic interaction between stress waves and cavitation is critical in producing effective stone comminution during SWL. [Work supported by NIH and NSF GRFP.]

2:15

4pBA4. Investigation on the effect of specular reflections from stone surface on twinkling artifact. Wei Lu, Oleg A. Sapozhnikov, John C. Kucewicz, Bryan W. Cunitz, Peter J. Kaczkowski, Lawrence A. Crum, and Michael R. Bailey (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle WA 98105)

The twinkling artifact can highlight kidney stones during ultrasound color Doppler imaging with high sensitivity for stone detection. The mechanism of the twinkling artifact is still under debate. It was reported previously that twinkling appeared distal to the echogenic reflection from the stone surface in cases with no signal saturation. [Lu *et al.*, JASA **129**(4), p. 2376]. In this report, the effect of specular reflections on twinkling was investigated. Human kidney stones (5-9 mm in length) were embedded in a polyacrylamide gel phantom. Radio-frequency (RF) data were recorded from pulse-echo ensembles using a software-programmable ultrasound system. The variability within the beamformed Doppler ensemble, which is responsible for twinkling, was traced back to the unbeamformed RF channel data to identify whether variability arose disproportionately on channels receiving the specular reflection. The results showed that the specular reflection did not saturate individual channels and that the variability was observed on most channels with similar magnitude, which indicates that the appearance of twinkling does not rely on the specular reflection from the stone surface. Instead in the beamformer, the varying signals have the appearance of arising from a point source within the stone. [Work supported by NIH DK43881, DK086371, DK092197, and NSBRI through NASA NCC 9-58.]

2:30

4pBA5. Inhibition of breast cancer cell proliferation by low-intensity pulsed ultrasound (LIPUS). Amit Katiyar, Kausik Sarkar (Mech. Eng., Univ. Of Delaware, Newark, DE 19716), and Krishna Sarker (Biological Sci. , Univ. of Delaware, Newark, DE 19716)

Cancer is the second leading cause of death in the United States, preceded only by heart disease. Cancer cells display an uncontrolled proliferation, controlling which has been a big challenge for cancer treatment. Ultrasound is best known for its application in diagnostic imaging; it is also a vehicle for delivering high frequency mechanical stimulation toward beneficial bio-effects. Unlike high intensity focused ultrasound, which is recently being investigated for thermal ablation of solid tumors, low intensity pulsed ultrasound (LIPUS) is directed toward cellular mechanisms. The effects of LIPUS on cancer cell proliferation are not known. Here, we demonstrate that LIPUS dose-dependently inhibits proliferation of breast cancer cell T47D as determined by several biochemical assays such as MTS, Alamar Blue, and BrdU assay. Statistically significant inhibition of T47D cell proliferation is observed when cells are exposed to 50–100 mW/cm^2. For this intensity range, LIPUS excitation inhibits the proliferation of T47D cells upto 50%. We also notice that inhibition of cell proliferation by LIPUS depends on its exposure time on cells. Minimum exposure time of LIPUS excitation for pronounced inhibitory effects on T47D cell proliferation is approximately 10 min.

2:45

4pBA6. Acoustical assessment of body water balance. Armen Sarvazyan (Artann Labs., 1459 Lower Ferry Rd., Trenton, NJ 08618)

A new medical application of acoustics has recently emerged: assessment of body hydration status by ultrasonic measurement of muscle water content. The need for an easy-to-perform method for the detection of water imbalance is of the utmost clinical importance. Body hypohydration may cause severe health and performance problems, decreasing cognitive and physical work capabilities, while excessive hydration is a common symptom of many other diseases. The speed of longitudinal acoustic waves in muscle, as well as in other soft tissues, is defined by tissue molecular composition because both the density and bulk compressibility of tissue depend mainly on short range molecular interactions. Skeletal muscle is the largest water compartment in the body; it comprises 40% of body mass and 75% of muscle is water. The ultrasound velocity in muscle is a linear function of water content with the slope of about 3.0 m/s per 1% change in water content. We will describe the design, measurement principles, and testing results for new acoustic devices for assessment of hydration status of elderly and infants, two most vulnerable groups of population. Advantages and disadvantages of acoustical method of hydration over currently available methods are discussed.

3:00

4pBA7. Use of highly nonlinear solitary waves for the assessment of dental implants. Bruk Berhanu (942 Benedum Hall, Dept. of Civil and Environ. Eng., 3700 O'Hara St., Pittsburgh, PA 15261, bruk.berhanu@gmail.com)

This paper presents a noninvasive technique based on the propagation of highly nonlinear solitary waves (HNSWs) to monitor the stability of dental implants. HNSWs are mechanical waves that can form and travel in highly nonlinear systems, such as one-dimensional chains of contacting spherical particles (i.e., granular crystals). In this study, a granular crystal-based actuator/sensor, designed and built at the University of Pittsburgh, was used to introduce HNSWs into dummy implants that were inserted into either hardened plaster or treated beef bones. The waves reflected at the interface between the particle and implant were monitored to estimate the change in stiffness of the material. The hydration of the plaster was monitored because it can be considered largely similar to the osseointegration process that occurs in the oral connective tissue once a dental-endosteal threaded implant is surgically inserted. In the experiment using bone, the implant-bone system was immersed in an acid bath causing decalcification of the bone and, therefore, reduced stiffness of the bone itself, simulating the inverse of osseointegration. Positive correlations were found, in both experiments, between certain properties of the HNSWs and the stiffness of the test object, demonstrating that HNSWs show promise for use in assessment of dental implants.

3:15

4pBA8. Artificial flesh material selection for hearing protection evaluation system. Mehmet M. A. Bicak, Josiah M. Oliver, and Kevin R. Shank (Adaptive Technologies. Inc., 2020 Kraft Dr. Ste. 3040, Blacksburg, VA 24060)

Developing an advanced hearing protection evaluation system (HPES) in the form of an acoustic test fixture (ATF) allows for characterizing either circumaural or insert-type hearing protection devices (HPDs) in both impulse and continuous noise environments across the dynamic range of human hearing. This is a challenging task since the acoustical transfer paths through flesh contribute to the dynamic response of the system. Current ATFs do not account for the transfer paths through flesh to ear canal. In this study, we investigated several visco-elastic flesh materials numerically using coupled vibro-acoustic simulations, and experimentally using vibration and acoustic excitation methods. Geometrically representative prototypes are being developed using volume computed tomography (VCT) that include detailed features of the skull and flesh structure, so that flesh conducted sound transmission paths can be physically modeled. The HPD on ATF dynamic behavior is compared with the HPD on subject behavior using finite element simulation models developed using the VCT images. The material selection is validated using noise reduction and vibration experiments on the subjects.

3:30

4pBA9. Cough count as a marker for patient recovery from pulmonary tuberculosis. Brian H. Tracey (ECE Dept., Tufts Univ., 196 Boston Ave., Rm., 4330 Medford, MA 02155, brian.tracey@tufts.edu), German Comina (Laboratorio de Ingeniería Fsica, Facultad de Ciencias, Universidad Nacional de Ingeniería, Rimac, Lima, Per), Sandra Larson (Michigan State Univ., College of Osteopathic Medicine, East Lansing, MI), Marjory Bravard (Massachusetts General Hospital, Boston, MA 02114), Jose W. Lopez (Unidad de Epidemiología, Hospital Nacional Dos de Mayo, Lima, Per), and Robert H. Gilman (Johns Hopkins Bloomberg School of Public Health, Baltimore, MD)

In regions of the world where tuberculosis (TB) poses the greatest disease burden, clinicians often lack access to skilled laboratories. This is particularly problematic for patients with drug-resistant tuberculosis, as these patients will otherwise receive standard TB medication and will not respond to treatment. Thus, a lab-free method for assessing patient recovery during treatment would be of great benefit. We hypothesize that cough analysis may provide such a test, and have carried out a pilot study to record coughs from a cohort of patients in Lima, Peru. We describe algorithm development for cough data analysis and compare several event detection and classification strategies. Results from our ongoing validation efforts suggest that cough count (cough/hour) decreases noticeably after the start of treatment in drug-responsive patients. Our long-term goal is development of a low-cost ambulatory cough analysis system that will help identify patients with drug-resistant tuberculosis.

Session 4pEA

Engineering Acoustics and Underwater Acoustics: Vector Sensors, Projectors, and Receivers II: Receivers, Reception, and Transmission

Stephen C. Butler, Cochair
Naval Undersea Warfare Center, 1176 Howell St., Newport, RI 02840

Roger T. Richards, Cochair
Naval Undersea Warfare Center, 1176 Howell St., Newport, RI 02840

Invited Papers

1:05

4pEA1. Acoustic intensity vector probes. Gary W. Elko (Mh Acoust. LLC, 25A Summit Ave., Summit, NJ 07901)

Modern acoustic intensity measurement techniques began in the early 1980's with the realization that the imaginary part of the cross-spectral density was directly related to the active intensity component along the axis between two closely-spaced pressure microphones. Since acoustic intensity is a vector quantity, it was obvious that one would like to measure all three orthogonal components and to graphically represent acoustic power flow through space. Needing an interesting topic for a Ph.D. thesis at Penn State (and a way of funding it ..., thank-you US Navy), the task of investigating and developing the estimation of the acoustic intensity vector field fortunately came my way. This talk will present some of the early interesting and fun things that came out of working on in-air acoustic vector sensors. It will conclude with some more recent developments that have direct connections to the early acoustic vector probes that were built, tested, and used at Penn State.

1:25

4pEA2. Vector sensors for airborne surveillance applications. James McConnell, Scott Jensen, Thomas McCormick, and Brendan Woolrich (Appl. Physical Sci. Corp., 475 Bridge St., Groton CT 06340)

The use of vector sensors for airborne surveillance applications (e.g., frequencies below 500 Hz) is discussed with emphasis on transducers that measure the acoustic pressure-gradient. Traditional approaches such as ribbon microphones, hot-wire anemometers, and finite-difference techniques will be reviewed. The crux of the presentation concerns a discussion of diffraction type pressure-gradient microphones that go beyond the classic ribbon microphone and utilize large format membrane transduction elements comprised of piezoelectric and electret materials. The results of analytical, numerical, and experimental evaluations will be presented.

1:45

4pEA3. Estimation of sea floor properties using acoustic vector sensors. Steven E. Crocker (Naval Undersea Warfare Ctr., 1176 Howell St., Newport, RI 02841), James H. Miller, John C. Osler (NATO Undersea Res. Ctr., La Spezia, Italy), Gopu R. Potty (Univ. of Rhode Island, Narragansett, RI 02882), and Paul C. Hines (DRDC Atlantic, Dartmouth, NS B2Y3Z7, Canada)

Information in the acoustic vector field can be used to estimate properties of the environment with which the field interacts. A study was performed to understand the value of vector field data when inverting for sea floor geoacoustic properties. The study compared results obtained with new inverse methods based on measurements of complex acoustic transfer functions and specific acoustic impedance. Acoustic field data consisted of gated continuous wave transmissions acquired with four acoustic vector sensors that spanned the water-sediment interface during the Sediment Acoustics Experiment 2004 (SAX04). Motion data provided by the buried vector sensors were affected by a suspension response that was sensitive to the sediment density and shear wave speed. The suspension response for the buried vector sensors included a resonance within the analysis band of 0.6–2.4 kHz. The response was sufficiently sensitive to the local geoacoustic properties, that it was exploited by the inverse methods developed for this study. Inversions of real and synthetic data sets indicated that information about sediment shear wave speed was carried by the suspension response of the buried sensors, as opposed to being contained inherently within the acoustic vector field. [Work supported by ONR.]

2:05

4pEA4. Comparing vector- and pressure-sensor arrays: The 2009 cooperative array performance experiment. Daniel Rouseff, Russell Light (Appl. Phys. Lab., Univ. of Washington, Seattle, WA 98105), Zhongkang Wang, and Shihong Zhou (Hangzhou Appl. Acoust. Res. Inst., Hangzhou, China)

The 2009 cooperative array performance experiment (CAPE'09) was designed to compare performance between vector- and pressure-sensor arrays. The experiment was a joint effort of Chinese and American investigators; both arrays were designed and assembled by the Hangzhou Applied Acoustics Research Institute (HAARI), while the source systems and signal processing/recording systems were designed and assembled by Applied Physics Laboratory, University of Washington (APL-UW). The two arrays, both approximately 7 m in length, were deployed vertically off the stern of the APL-UW's R/V Robertson in Lake Washington, Seattle. Various transmitted signals in the 1.5–4 kHz band were recorded simultaneously on the two arrays at ranges between 10 m and 4 km. The signals included repeated linear frequency-modulated chirps and communications sequences. The pressure- and vector-sensor arrays had 32 and 8 uniformly spaced elements, respectively. Because each element in the vector-sensor array recorded both pressure and the three components of particle velocity, the two arrays made the same number of measurements over a similar vertical aperture. In the present talk, the design features of the vector-sensor array are emphasized. Sample results for both arrays are presented. [Work supported by ONR.]

2:25

4pEA5. Acoustic particle velocity amplification with horns. Dimitri M. Donskoy (Stevens Inst. of Technol., 711 Hudson St., Hoboken, NJ 07030)

Previously [Donskoy and Cray, J. Acoust. Soc. Am., **129**(4), Pt. 2, 2644 (2011)] the authors numerically investigated an acoustic particle velocity amplification effect with conical open-ended horns. Here, using Webster's approach, an analytical solution for the particle velocity response of conical horns (single and double) is derived and analyzed. The solutions are verified by comparison with direct numerical computations and supported with experimental measurements. It is shown that small horns, compared to the acoustic wavelength, are capable of providing substantial particle velocity amplification. For example, a 20 cm horn, in-water, can deliver nearly 10 dB of amplification over a very broad frequency range (from zero to 1000 Hz) without significant amplitude and phase distortion. Another unique feature of the velocity horn is its dipole directionality. The paper presents a thorough analysis of horn's amplification versus geometrical parameters. [This work is supported in part by the ONR summer fellowship program].

Contributed Papers

2:45

4pEA6. Analysis of the advantages and complexities of acoustic vector sensor arrays. Andrew J. Poulsen (Appl. Physical Sci. Corp., 49 Waltham St., Ste. 4, Lexington, MA 02421, apoulsen@aphysci.com) and Arthur B. Baggeroer (MIT, Cambridge, MA 02139)

The hydrophone, an omnidirectional underwater microphone, is the most common sensor for listening to underwater sound. Directional sensors, however, have many important applications. Acoustic vector sensors, one important class of directional sensors, measure acoustic scalar pressure along with acoustic particle motion. With this additional vector measurement, vector sensors feature many advantages over conventional omnidirectional hydrophone sensors: improved array gain/detection performance, enhanced bearing resolution, the ability to "undersample" an acoustic wave without spatial aliasing, and the capability of attenuating spatial ambiguity lobes, e.g., left/right ambiguity resolution for a linear array. Along with their advantages, however, vector sensors also pose additional practical complexities: greater sensitivity to non-acoustic, motion-induced flow noise at low frequencies, requisite knowledge/measurement of each sensor's orientation, management of different sensor types (pressure and particle motion) that each with different noise properties/calibration requirements, and adaptive processing can become difficult in a snapshot limited regime since each vector sensor is made up of up to four data channels. This paper will explore the virtues and limitations of vector sensor arrays in the presence of realistic ocean noise fields and system imperfections, including their effects on array performance (gain, beampatterns, etc.) supported by theoretical analysis and illustrative examples.

3:00

4pEA7. Vector wave measurements on landmine detection with an array of loudspeakers focused on the ground. Martin L. Barlett, Justin D. Gorhum, Wayne M. Wright, Mark F. Hamilton, and Thomas G. Muir (Appl. Res. Labs., The Univ. of Texas at Austin, P.O. Box 8029, Austin, TX 78713-8029)

An array of 16 loudspeakers, deployed along a segment of the base of a right circular cone, was used to focus sound on soils overlying buried targets lying along the conical axis of the source. Measurements were made at several incident angles tending toward grazing to examine long range detection for humanitarian de-mining applications. Targets and soils were instrumented with triaxial geophone and accelerometer sensors. The transmission of airborne sound into the soils produced vertical and radial vibrations in both the soil and the targets, which included rigid and compliant mine cases. Several waveform transmission types were utilized. The compliant targets provided resonances amenable to optical detection, depending on the acoustic, geometric and environmental parameters, which are discussed. [Work supported by the IRD program at ARL:UT Austin, in cooperation with the National Center for Physical Acoustics.]

3:15

4pEA8. Capacity and statistics of measured underwater acoustic particle velocity channels. Huaihai Guo, Chen Chen, Ali Abdi (Elec. & Comput. Eng. Dept., New Jersey Inst. of Technol., Newark, NJ 07102, hg45@njit.edu), Aijun Song, Mohsen Badiey (College of Earth, Ocean, and Environment, Univ. of Delaware, Newark, DE 19716), and Paul Hursky (Heat, Light, and Sound Res., Inc., La Jolla, CA 92037)

Acoustic particle velocity channels can be used for communication in underwater systems [A. Abdi and H. Guo, IEEE Trans. Wireless Communi. **8**, 3326–3329, (2009)]. In this paper, the information (Shannon) capacity of underwater acoustic particle velocity channels is studied using measured data. More specifically, the maximum achievable data rates of a compact vector sensor communication receiver and another communication receiver with spatially separated scalar sensors are compared. Some statistics of particle velocity channels such as amplitude distribution and power delay profile are investigated using measured data and proper models are suggested as well. The results are useful for design and simulation of vector sensor underwater communication systems in particle velocity channels. The work is supported in part by the National Science Foundation (NSF), Grant CCF-0830190.

Session 4pNS

Noise and Physical Acoustics: Launch Vehicle Noise II

R. Jeremy Kenny, Cochair
Marshall Space Flight Center, Bldg. 4203, Huntsville, AL 35812

Kent L. Gee, Cochair
Dept. of Physics and Astronomy, Brigham Young Univ., N243 Eyring Science Center, Provo, UT 84602

Invited Papers

1:10

4pNS1. Overview of the Ares I scale model test program. Douglas D. Counter (Bldg. 4203, M.S. ER42, Marshall Space Flight Ctr., Huntsville, AL 35812, douglas.d.counter@nasa.gov) and Janice Houstion (Jacobs Eng., Huntsville, AL 35812)

Launch environments, such as lift-off acoustic (LOA) and ignition overpressure (IOP), are important design factors for any vehicle and are dependent upon the design of both the vehicle and the ground systems. LOA environments are used directly in the development of vehicle vibro-acoustic environments and IOP is used in the loads assessment. The NASA Constellation Program had several risks to the development of the Ares I vehicle linked to LOA. The risks included cost, schedule, and technical impacts for component qualification due to high predicted vibro-acoustic environments. One solution is to mitigate the environment at the component level. However, where the environment is too severe for component survivability, reduction of the environment itself is required. The Ares I scale model acoustic test (ASMAT) program was implemented to verify the Ares I LOA and IOP environments for the vehicle and ground systems including the mobile launcher (ML) and tower. An additional objective was to determine the acoustic reduction for the LOA environment with an above deck water sound suppression system. ASMAT was a development test performed at the Marshall Space Flight Center (MSFC) East Test Area (ETA) Test Stand 116 (TS 116). The ASMAT program is described in this presentation.

1:30

4pNS2. Ares I Scale Model Acoustic Tests Instrumentation for Acoustic and Pressure Measurements. Magda B. Vargas (Bldg. 4203, M.S. ER42, MSFC, Huntsville, AL 35812, magda.b.vargas@nasa.gov) and Douglas D. Counter (MSFC, Huntsville, AL, 35812)

The Ares I Scale Model Acoustic Test (ASMAT) was a development test performed at the Marshall Space Flight Center (MSFC) East Test Area (ETA) Test Stand 116. The test article included a 5% scale Ares I vehicle model and tower mounted on the Mobile Launcher. Acoustic and pressure data were measured by approximately 200 instruments located throughout the test article. There were four primary ASMAT instrument suites: ignition overpressure (IOP), lift-off acoustics (LOA), ground acoustics (GA), and spatial correlation (SC). Each instrumentation suite incorporated different sensor models which were selected based upon measurement requirements. These requirements included the type of measurement, exposure to the environment, instrumentation check-outs and data acquisition. The sensors were attached to the test article using different mounts and brackets dependent upon the location of the sensor. This presentation addresses the observed effect of the sensors and mounts on the acoustic and pressure measurements.

1:50

4pNS3. Measurements of the ground acoustic environments for small solid rocket motor firings. Bruce T. Vu (NASA Kennedy Space Ctr., NE-M1, KSC, FL 32899, Bruce.T.Vu@nasa.gov) and Kenneth J. Plotkin (Wyle Labs., Arlington, VA 22202, Kenneth.Plotkin@wyle.com)

During the ground launch of a space vehicle, the mobile launcher deck and tower are exposed to severe acoustic environments. These environments, if not properly managed, can weaken ground support equipment and result in structure failure. The ground acoustic environments are different than the vehicle acoustic environments. They are typically more severe because of the close proximity of the rocket plume, which often involves direct impingement. They are more difficult to predict, and their measurement and data reduction remain challenging. This paper discusses these challenges and describes the methods of processing ground acoustic data during a series of static firings of a 5-percent scale solid rocket launch vehicle and mobile launcher, known as the Ares Scale Model Acoustic Test.

2:10

4pNS4. Ares I scale model acoustic test lift-off acoustics. Douglas D. Counter (MSFC, M.S. ER42, Huntsville, AL, 35812, douglas.d.counter@nasa.gov) and Janice Houston (Jacobs Egnineering, Huntsville, AL, 35812)

The lift-off acoustic (LOA) environment is an important design factor for any launch vehicle. For the Ares I vehicle, the LOA environments were derived by scaling flight data from other launch vehicles. The Ares I LOA predicted environments are compared to the Ares I scale model acoustic test (ASMAT) preliminary results.

2:30

4pNS5. Ares I Scale Model Acoustic Test above deck water sound suppression results. Douglas D. Counter (MSFC, M.S. ER42, Huntsville, AL 35812, douglas.d.counter@nasa.gov) and Janice Houston (Jacobs Eng., Huntsville, AL 35812)

The Ares I Scale Model Acoustic Test (ASMAT) program test matrix was designed to determine the acoustic reduction for the LOA environment with an above deck water sound suppression system. The scale model test can be used to quantify the effectiveness of the water suppression system as well as to optimize the systems necessary for LOA noise reduction. Several water flow rates were tested to determine which rate provides the greatest acoustic reductions. Preliminary results are presented.

2:50

4pNS6. Recovering the spatial correlation of liftoff acoustics from the Ares Scale Model Acoustics Test. Brian Prock, Paul Bremner (ATA Eng., Inc.), and Thomas L. Philley (NASA JSC)

Accurately predicting structural vibrations due to acoustic loads requires knowledge about the overall sound pressure levels, the amplitude of the sound pressure levels as a function of frequency (auto-spectra), and the spatial correlation of the sound pressure levels (cross-spectra). When dealing with the liftoff acoustics of launch vehicles, a large amount of historical data is available in terms of overall levels and auto-spectra, but only a limited amount of data exists for cross-spectra. ATA Engineering has used data taken during NASAs Ares Scale Model Acoustic Test to recover the spatial correlation of liftoff acoustics for a typical launch vehicle. By assuming the measured liftoff acoustics that are a combination of propagating waves and diffuse acoustic fields, curve-fitting algorithms are used to recover spatial correlation parameters required by modern vibro-acoustic analysis software.

3:10–3:30 Break

3:30

4pNS7. 5% Ares I scale model acoustic test: Overpressure characterization and analysis. David Alvord (Jacobs Eng., M.S. ER42, MSFC, Huntsville, AL 35812, david.alvord@nasa.gov), Matthew Casiano, and David McDaniels (MSFC, Huntsville, AL, 35812)

During the ignition of a ducted solid rocket motor (SRM), rapid expansion of injected hot gases from the motor into a confined volume causes the development of a steep fronted wave. This low frequency transient wave propagates outward from the exhaust duct, impinging the vehicle and ground structures. An unsuppressed overpressure wave can potentially cause modal excitation in the structures and vehicle, subsequently leading to damage. This presentation details the ignition transient findings from the 5% Ares I scale model acoustic test (ASMAT). The primary events of the ignition transient environment induced by the SRM are the ignition overpressure (IOP), duct overpressure (DOP), and source overpressure (SOP). The resulting observations include successful knockdown of the IOP environment through use of a space shuttle derived IOP suppression system, a potential load applied to the vehicle stemming from instantaneous asymmetrical IOP and DOP wave impingement, and launch complex geometric influences on the environment. The results are scaled to a full-scale Ares I equivalent and compared with heritage data including Ares I-X and both suppressed and unsuppressed space shuttle IOP environments.

3:50

4pNS8. Simulation of acoustics for Ares I scale model acoustic tests. Gabriel C. Putnam and Louise L. Strutzenburg (Marshall Space Flight Ctr., MSFC / ER42, Huntsville, AL, 35812)

The Ares I scale model acoustics test (ASMAT) is a series of live-fire tests of scaled rocket motors meant to simulate the conditions of the Ares I launch configuration. These tests have provided a well documented set of high fidelity acoustic measurements useful for validation including data taken over a range of test conditions and containing phenomena like ignition over-pressure and water suppression of acoustics. To take advantage of this data, a digital representation of the ASMAT test setup has been constructed and test firings of the motor have been simulated using the LOCI/CHEM computational fluid dynamics software. Results from ASMAT simulations with the rocket in both held down and elevated configurations, as well as with and without water suppression have been compared to acoustic data collected from similar live-fire tests. Results of acoustic comparisons have shown good correlation with the amplitude and temporal shape of pressure features and reasonable spectral accuracy up to approximately 1000 Hz. Major plume and acoustic features have been well captured including the plume shock structure, the igniter pulse transient, and the ignition overpressure.

4:10

4pNS9. Hybrid computational fluid dynamics and computational aero-acoustic modeling for liftoff acoustic predictions. Louise L. Strutzenberg (MSFC, M.S. ER42, MSFC, Huntsville, AL 35812, louise.s@nasa.gov) and Peter A. Liever (CFD Res. Corp., Huntsville, AL 35812)

This paper presents development efforts at the NASA Marshall Space flight Center to establish a hybrid computational fluid dynamics and computational aero-acoustics (CFD/CAA) simulation system for launch vehicle liftoff acoustics environment analysis. Acoustic prediction engineering tools based on empirical jet acoustic strength and directivity models or scaled historical measurements are of limited value in efforts to proactively design and optimize launch vehicles and launch facility configurations for liftoff acoustics. CFD based modeling approaches are now able to capture the important details of vehicle specific plume flow environment, identify the noise generation sources, and allow assessment of the influence of launch pad geometric details and sound mitigation measures such as water injection. However, CFD methodologies are numerically too dissipative to accurately capture the propagation of the acoustic waves in the large CFD models. The hybrid CFD/CAA approach combines the high-fidelity CFD analysis capable of identifying the acoustic sources with a fast and efficient boundary element method (BEM) that accurately propagates the acoustic field from the source locations. The BEM approach was chosen for its ability to properly account for reflections and scattering of acoustic waves from launch pad structures. The paper will present an overview of the technology components of the CFD/CAA framework and discuss plans for demonstration and validation against test data.

4:30

4pNS10. Validation study on computational aeroacoustics of acoustic waves from sub-scale rocket plumes. Seiichiro Morizawa (Dept. of Aerosp. Engineering, Tohoku Univ., 3-1-1 Yoshinodai, Chuo-ku, Sagamihara 25205210, morizawa@edge.ifs.tohoku.ac.jp), Taku Nonomura (Inst. of Space and Astronautical Sci., Sagamihara 2525210), Seiji Tsutsumi (JAXA's Eng. Deigital Innovation, JAXA, Sagamihara 2525210), Nobuhiro Yamanishi, Keita Terashima (Space Transportation Mission Directorate, Tsukuba 3058505), Shigeru Obayashi (Tohoku Univ., 2-1-1 Katahira Aoba-ku Sendai, 9808577), and Kozo Fujii (Inst. of Space and Astronautical Sci., Sagamihara 2525210)

In this paper, the comparative study of prediction based on computational aeroacoustics (CAA) and experimental results for acoustic waves from modeled rocket motors is conducted, and prediction accuracy of CAA is discussed in the framework of JAXA-CNES collaboration. Experimental data of flow and acoustic fields of solid motor by JAXA and H2-AIR liquid motor by CNES are used as the reference. Two types of computational codes are adapted in this study. The predictions of sound pressure level by both computational codes agree reasonably with corresponding experimental data, whereas the errors are approximately less than 5 dB. In addition, each aeroacoustic field of CAA results in this study is discussed in detail.

Contributed Papers

4:50

4pNS11. Measurement and propagation of supersonic aeroacoustic noise sources using continuous scanning measurement technologies and the fast multipole boundary element method. Michael Y. Yang, Havard Vold, and Parthiv N. Shah (11995 El Camino Real, San Diego, CA 92130)

ATA Engineering has developed a technique which uses a continuous scanning robot to take high-resolution measurements of supersonic jet plumes. The jet noise was modeled using a reduced-order model and propagated to far field microphone locations in the free-field. It is shown that the pressure at these microphones was successfully reconstructed across a range of frequencies. The capability to make predictions when scattering surfaces are present is also demonstrated using the fast multipole boundary element method in VA One. This work was originally designed for supersonic jets but can also be used for static firing tests of launch vehicle engines. The measured data could then be used for analytic predictions of the liftoff environment.

5:05

4pNS12. On the jet-wake similarity. Ballard W. George (1367 Boblink Circle, Sunnyvale, CA 94087)

This paper is concerned with similarities and differences between jets and wakes, as indicated by a sampling of the literature, including what tests and studies have been made in each case. As noted by Franken, jets and wakes are both characterized by a region of shear and (depending on the Reynolds number) high turbulence. Jet noise has been extensively studied under the impetus of a strong concern with community noise. Jet noise studies have examined numerous quantities including, for example, spectra, directivity, source location, and total power. Lighthill published widely quoted papers on sound generated aerodynamically, primarily in the context of jet noise and without solid boundaries. Literature referred to for this paper was primarily for airborne sound and included underwater sound, in which case cavitation plays a significant role and tends to mask turbulent wake noise due to the vehicle body. Propellers are a source of wake as well as lift noise. A readily noticeable difference between jets and wakes involves the fact that jets can be tested statically, while for wakes there has to be relative motion. Also jets, though not jet engines, can be tested "cold."

THURSDAY AFTERNOON, 3 NOVEMBER 2011 — ESQUIRE, 1:55 TO 4:45 P.M.

Session 4pPP

Psychological and Physiological Acoustics: Perceptual Aspects of Sound

Elizabeth A. Strickland, Chair
Dept. of Speech, Language, and Hearing Sciences, Purdue Univ., 500 Oval Dr., West Lafayette, IN 47907

Chair's Introduction—1:55

Contributed Papers

2:00

4pPP1. Mistuned harmonic detection and the role of tonotopically local neural synchrony. William M. Hartmann (Depatment of Phys. and Astronomy, Michigan State Univ., East Lansing, MI 48824)

A mistuned harmonic in an otherwise periodic complex tone is a simple stimulus for the study of segregation and integration of complex tones by human listeners. In a mistuned harmonic *detection* experiment, a listener must discriminate between two tones—one perfectly periodic and the other with a mistuned harmonic. This detection paradigm is subject to known artifacts. However, with careful experiment design the artifacts can be controlled, and the paradigm becomes an efficient way to explore the emergence of segregated mistuned spectral components. The results of mistuned harmonic detection experiments support a tonotopically local model in which detection is mediated by the dephasing of a neural spike sequence in a tonotopically tuned channel. Evidence for this model comes from (1) the functional dependences of detectability on the amount of mistuning and the tone duration, (2) the non-monotonic level dependence of detectability, (3) the lowpass character of detection—indicating an essential role for neural synchrony, and (4) the need for tonotopically local interaction as evidenced by experiments using mistuned distracters, mistuned harmonics in spectral gaps, and dichotic presentation. [Work supported by the NIDCD and the AFOSR.]

2:15

4pPP2. Parametric issues in measuring the olivocochlear reflex with a masking technique. Elin M. Roverud and Elizabeth A. Strickland (Dept. of Speech, Lang., and Hearing Sci., Purdue Univ., 500 Oval Dr. West Lafayette, IN 47907, eroverud@purdue.edu)

Previous studies in this laboratory have suggested that forward masking occurs by two mechanisms, excitatory masking and gain reduction via the medial olivocochlear reflex (MOCR), that operate over different time courses. In this laboratory, a forward masking technique is used in which a precursor, intended to elicit the MOCR, is followed by a fixed off-frequency forward masker and signal. A prior study [Roverud and Strickland, J. Acoust. Soc. Am. **128**, 1203–1214] found that for short precursors, signal threshold increased then decreased (buildup) as the signal and masker were delayed from the precursor. This result could be consistent with the sluggish onset of the MOCR. When the masker is removed, however, buildup is not observed as the signal is delayed from a precursor, even though the precursor should still be eliciting the MOCR. This raises the question, what is the role of the masker and why is it necessary to observe buildup? The current study examines the frequency and level characteristics of the masker required to observe the buildup effect. Results will be discussed in terms of gain reduction and temporal window models of forward masking. [Research supported by a Grant from NIH(NIDCD) R01 DC008327.]

2:30

4pPP3. Perceiving auditory distance using level and direct-to-reverberant ratio cues. Andrew J. Kolarik, Silvia Cirstea, and Shahina Pardhan (VERU, Eastings 204, Anglia Ruskin Univ., East Rd., Cambridge, CB1 1PT, United Kingdom, andrew.kolarik@anglia.ac.uk)

The study investigated how level and reverberation cues contribute to distance discrimination, and how accuracy is affected by reverberation cue strength. Sentence pairs were presented at distances between 1 and 8 m in a virtual room simulated using an image-source model and two reverberation settings (lower and higher). Listeners performed discrimination judgments in three conditions: level cue only (Level-Only), reverberation only (Equalized), and both cues available (Normal). Percentage correct judgment of which sentence was closer was measured. Optimal distance discrimination was obtained in the Normal condition. Perception of the difference in distance between sentences had a lower threshold (i.e., performance was significantly better, $p<0.05$) for closer than further targets in Normal and Level-Only conditions. On the contrary, in the Equalized condition, these thresholds were lower for further than closer targets. Thresholds were lower at higher reverberation in the Equalized condition, and for further targets in the Normal condition. Data indicate that level generally provided more accurate discrimination information than direct-to-reverberant ratio. Direct-to-reverberant ratio provided better information for sounds further from the listener than for nearer sounds, and listeners were able to use direct-to-reverberant ratio as effectively as level in highly reverberant rooms when discriminating far sound sources.

2:45

4pPP4. Infants' vowel discrimination in modulated noise. Lynne A. Werner (Dept. Speech & Hearing Sci., Univ. Washington, 1417 NE 42nd St., Seattle WA 98105-6246)

Adult listeners detect and discriminate target sounds better in amplitude modulated noise than in unmodulated noise. This study examined infants' ability to take advantage of masker modulation to improve sensitivity to a target. Listeners were 7–9-month-old infants and 18–30-year-old adults. Vowel discrimination in noise was tested. Listeners learned to respond when a repeated vowel changed from /a/ to /i/ or from /i/ to /a/. An observer-based method was used to assess sensitivity to the vowel change. The maskers were speech-spectrum noise either unmodulated, amplitude modulated with the envelope of single-talker speech or sinusoidally amplitude modulated at 8 Hz with a 75% modulation depth. The overall level of the maskers was 60 dB SPL. The level of the vowels, chosen to yield an average d' of 1 in the unmodulated masker, was 46 dB SPL for adults and 58 dB SPL for infants. Adults' d' was substantially and significantly higher in both modulated maskers than in the unmodulated masker. Infants' d' was also significantly higher in the two modulated maskers than in the unmodulated masker, but the improvement due to modulation was significantly smaller for infants than for adults. [Work supported by NIDCD R01DC00396 and P30DC04661.]

3:00

4pPP5. Effects of silent interval on human frequency-following responses to voice pitch. Fuh-Cherng Jeng and Ronny P. Warrington (Commun. Sci. and Disord., Ohio Univ., 1 Ohio Univ. Dr. Athens, OH 45701, jeng@ohio.edu)

Human frequency-following responses (FFRs) to voice pitch have provided valuable information on how the human brain processes speech information. Recordings of the FFR to voice pitch, however, may overlap when insufficient silent intervals are used. To determine the shortest silent interval that can be used with no overlap between adjacent response waveforms, FFRs were recorded from 12 Chinese adults using a wide range of silent intervals. The stimulus token was a Chinese monosyllable with a rising pitch of 117–166 Hz and a duration of 250 ms. A high stimulus intensity at 70 dB SPL was used to maximize overlaps in the response waveforms. A total of seven silent intervals, ranging from the full length of the stimulus duration down to approximately half period of the fundamental frequency of the stimulus token, were administered at a random order across participants. Two distinct methods (Hilbert transform and root-mean-square amplitudes) were used to delineate the envelopes and overlaps of the response waveforms. A one-way repeated measures analysis of variance was significant ($p = 0.038$) in defining the magnitude of overlaps for the 10 ms pre-stimulus interval. The results indicated the shortest silent interval that could be used without compromising the response is between 35 and 45 ms.

3:15

4pPP6. Psychoacoustics of chalkboard squeaking. Christoph Reuter (Musicological Inst., Univ. of Vienna, Vienna, Austria) and Michael Oehler (Univ. of Cologne, Cologne, Germany)

At least since 1975 the "pleasantness" of a sound is discussed from many different angles (Ely 1975; Aures 1984; Halpern *et al*. 1986; Vaschillo 2003; Neumann & Waters 2006; Cox 2008), but often chalkboard squeaking or scratching a chalkboard with finger nails tops the list of unpleasant sounds. The aim of the presented study is to detect specific parts of the sounds that make chalkboard squeaking particularly unpleasant. With a combination of perception experiments and electro-physiological measurements, it was analyzed to what extent the knowledge about the sounds influenced the subjects' judgments and/or the physiological reactions. Basically the study is a replication of Halpern *et al*. (1986), whose methods were extended by several sophisticated sound analysis and re-synthesis techniques and the measurement of some electro-physiological parameters (heart rate and skin resistance) during listening. First results show that especially the modification of the tonal parts as well as applying a filter between 2000 and 4000 Hz led to a more pleasant sound perception. Almost all stimuli were rated more unpleasant if the subjects knew about the nature of the sounds.

3:30–3:45 Break

3:45

4pPP7. The shopping sound experience. Ralf Jung, Gerrit Kahl, and Lbomira Spassova (DFKI, Campus D 3 4, 66123 Saarbrücken 66123 Germany, Ralf.Jung@dfki.de)

The attempt to influence the shopping behavior of customers in a supermarket through music often fails due to the different music preferences. In this work, a method to bring personalized music in the supermarket is presented by providing a location-aware playback and notification service. By using a web interface, customers are able to create an electronic shopping list, associate list items with music tracks, and thus create an individual shopping playlist. These product-associated music tracks start playing at the customer's instrumented shopping cart when he enters specific product departments where an item from the shopping list is located. Additionally, a service to provide product-awareness through non-speech audio cues is presented. This location-aware service notifies customers when they come closer to products that are listed on their shopping list. A two-stage notification approach is used to create user-centric notification zones. Depending on the customer's distance to the product, he gets notified by an ambient or arousal noise that is mixed into the background music. In addition, further information about the detected product is displayed on the screen that is mounted at our instrumented shopping cart.

4:00

4pPP8. Psychological factors influencing the evaluation of electric vehicle interior noise. Jochen Steffens (Inst. of Sound and Vib. Eng., Duesseldorf Univ. of Appl. Sci., Josef-Gockeln-Strasse 9, 40474 Dsseldorf, Germany, jochen.steffens@fh-duesseldorf.de), Thomas Kueppers (Daimler AG, 70372 Stuttgart, Germany), and Sabrina Skoda (Duesseldorf Univ. of Appl. Sci., 40474 Dsseldorf, Germany)

Increasing global awareness of the benefits of electromobility has brought about the need for new concepts in terms of the acoustic design of future vehicle generations. This includes both the creative design process and the development of suitable methods for the subjective evaluation of target sounds. The main difference between e-car sound surveys and those carried out on familiar sound categories is the potential consumer's lack of experience with electric vehicles. Thus, the consumer has no, or very unspecific, expectations in this regard. Several studies have shown that many subjects have to construct their personal frame of reference for evaluation within the listening experiment. However, this is possibly at odds with experience-based expectations relating to sounds of conventional combustion engines. The result is a conflict of objectives between the traditional and the modern, familiarity and strangeness, and not least between driving freedom and ecological awareness. In this context, the authenticity of the sound and the subjective interpretability of the sound information also appear as moderator variables. Moreover, associations with other vehicle categories, for example, streetcars, also influence the perceived sound quality. Within this contribution, these factors will be expounded and their influence on the evaluation of interior noise discussed.

4:15

4pPP9. Auditory-proprioceptive interaction—How do acceleration forces influence the evaluation of driving sounds? Sabrina Skoda, Jochen Steffens (Inst. of Sound and Vib. Eng., Duesseldorf Univ. of Appl. Sci., Josef-Gockeln-Strasse 9, 40474 Duesseldorf, Germany, sabrina.skoda@fh-duesseldorf.de), Thomas Kueppers (Daimler AG, Mercedesstrasse 122, 70372 Stuttgart, Germany), and Joerg Becker-Schweitzer (Inst. of Sound and Vib. Eng., Duesseldorf Univ. of Appl. Sci., Josef-Gockeln-Strasse 9, 40474 Duesseldorf, Germany)

One fundamental requirement for a widespread use of electric powered vehicles is a high degree of social acceptance of alternative drive concepts. As part of this, perception and evaluation of comfort and quality in a vehicle become increasingly important. The customers' judgment on these factors is strongly influenced by the noise and vibration behavior of the vehicle and always is passed in the context of multiple sensory impressions which are processed in the human brain consciously and unconsciously. The interaction mechanisms of sensory perception are highly complex and raise several scientific questions. During the past years, valuable insights about the interaction of visual and auditory perception have been obtained and there are also a number of theories about the auditory-tactile interaction. In contrast, the connection between auditory and proprioceptive perception remains largely unexplored. This paper deals with the question how acceleration forces influence human auditory perception. In order to make evident possible crossmodal effects, a listening experiment was conducted in different kinds of driving simulators. The results of this study will be presented and discussed.

4:30

4pPP10. Structurized sound design process of electric vehicle interior sound. Thomas Kueppers (Daimler AG, NVH Electric Powertrain, Stuttgart, D-70372 Germany, thomas.kueppers@daimler.com), Jan-Welm Biermann (IKA, RWTH Aachen Univ.), and Jochen Steffens (SAVE Inst., FH Duesseldorf Univ.)

The sound character of electric vehicles extremely differs from internal combustion engine vehicles. While the electric vehicle development is still in the beginning, there is potential to consider customer expectations very early in the development process, to have an impact on the typical desired electric vehicle sound on market launch and to generate unique selling propositions in competition between vehicle manufacturers. The current electric powertrain noise is perceived as inconvenient and nonaesthetical especially on long journeys. A complete reduction of this powertrain noise emphasizes the perception of wind and rolling noise and avoids feedback of velocity and load dependency, which decreases emotional impression. Futuristic sounds by sound design systems have to be evaluated by customers and revised in their acceptance and authenticity. Besides the improvement of the sound character, sound design systems can additionally be used to influence the informative acoustic feedback of driving and vehicle parameters. This article introduces the upcoming scenarios for interior sound development and focuses the contradiction between a great amount of freedom in design and a structurized workflow for designing electrical powertrain sounds.

THURSDAY AFTERNOON, 3 NOVEMBER 2011 — PACIFIC SALON 4/5, 1:30 TO 3:20 P.M.

Session 4pSCa

Speech Communication: Forensic Acoustics—On the Leading Edge of the Tidal Wave of Change About to Hit Forensic Science in the US? II

Geoffrey Stewart Morrison, Chair
School of Electrical Engineering, Univ. of New South Wales, Sydney, NSW, 2052, Australia

Chair's Introduction—1:30

Contributed Papers

1:35

4pSCa1. Selection of speech/voice vectors in forensic voice identification. James Harnsberger (Dept. Linguist., U. Florida, Gainesville, FL 32607) and Harry Hollien (Instit. Adv. Study Comm. Proc., U. Florida, Gainesville, FL 32611)

The case for the use of speech/voice vectors in speaker identification was made by Hollien and Harnsberger (2010). Those vectors found most robust in capturing speaker-specific characteristics were voice quality, vowel quality, speaking fundamental frequency, and temporal features. In this study, the speech cues for each of the four vectors will be compared to each other with respect to their predictive power. In addition, different vector algorithms and/or processing approaches for each will be contrasted in terms of their effects on identification robustness. One example is conversion of the vowel formant frequencies and bandwidth measurements to geometric scaling (semits). Finally, a second dataset of 18 male voices obtained from evidence recordings, paired with exemplars recorded from a speaker pool, were used to test a modified form of this speech/voice vector approach. The data from these subjects will be compared to those from the 1993 study (substantial improvement) and those from the 2010 experiment (confirmation).

1:50

4pSCa2. When to punt on speaker comparison? Reva Schwartz (U.S. Secret Service, Forensic Services Div. 950 H St. NW, Ste. 4200, Washington, DC 20223), Joseph P. Campbell, and Wade Shen (MIT Lincoln Lab., Lexington, MA, 02420)

In forensic speaker comparison, it is crucial to decide when completion of the examination may not possible (punt). We explore the factors that make speaker comparison decisions difficult or impossible. These factors may include duration, noise, speaking style, language/dialect, mental state, number of speakers, type and quality of recording, and deception. The analyst needs criteria to decide to reject case work. We present analysis of some of these factors and their impact on automatic speaker recognition systems. We propose a methodology for setting objective thresholds by which comparison examples can be rejected. This methodology could be used by forensic analysts to decide whether or not to proceed with speaker comparisons involving these factors.

2:05

4pSCa3. Defining the default defense hypothesis in likelihood-ratio forensic voice comparison. Felipe Ochoa and Geoffrey Stewart Morrison (Forensic Voice Comparison Lab., School of Elec. Eng. & Telecom., Univ. of New South Wales, Sydney, New South Wales, Australia)

In forensic DNA comparison, the person submitting samples for evaluation does not know what properties the samples will have when they are analyzed at the laboratory; samples are submitted as a matter of routine. In contrast, in forensic voice comparison the decision to submit samples for evaluation is based on prior screening: Typically a police officer, a lay person with respect to forensic voice comparison, has listened to the questioned-speaker recording and the known-speaker recording and decided that they sound sufficiently similar that they could be the same speaker and merit evaluation by a forensic scientist. If they do not sound sufficiently similar they are not submitted for evaluation. Unless the defense proposes a more restrictive hypothesis, the forensic scientist should therefore adopt the following as the default defense hypothesis and select a background database accordingly: The known speaker is not the same person as the questioned speaker but is one member of a population of speakers whom to a lay person sound sufficiently similar to the voice on the questioned-voice recording that they would submit recordings of these speakers for forensic comparison with the questioned-voice recording. Examples of how this theory might be applied are discussed.

2:20–2:35 Break

2:35

4pSCa4. Human error rates for speaker recognition. Wade Shen, Joseph P. Campbell (MIT Lincoln Lab., 244 Wood St, C-290A, Lexington, MA, 02420), and Reva Schwartz (U.S. Secret Service, Forensic Services Div. Ste. 4200, Washington, DC 20223)

It is commonly assumed that speaker identification by human listeners is an innate skill under certain conditions. As such, human listening tests have served as the benchmark for automatic recognition systems. In recent evaluations comparing human and machine performance on a speaker comparison task, error rates of naïve human listeners far exceed those of machines [special session on Human Assisted Speaker Recognition, IEEE ICASSP, Prague, 2011]. In this presentation, we quantify the performance of naïve listeners in a variety of challenging channel conditions and we compare these results against automatic systems and trained human listeners. The results of these experiments impact the admissibility of both forensic voice analysis and courtroom testimony by human listeners.

2:50

4pSCa5. Investigating the acoustic and phonetic correlates of deceptive speech. C. Kirchhuebel (Dept. of Electronics, Audio Lab., Univ. of York, Heslington, York, UK YO10 5DD, ck531@york.ac.uk)

The following study describes an initial investigation into the acoustic and phonetic correlates of deceptive speech using auditory and acoustic analysis. Due to the lack of extant data suitable for acoustic analysis, a laboratory-based experiment was designed which employed a mock-theft paradigm in conjunction with a “security interview” to elicit truthful and deceptive speech as well as control data from a total of ten male native British English speakers. Using PRAAT, the control, truthful, and deceptive speech samples were analyzed on a range of speech parameters including f0 mean and variability, intensity, vowel formant frequencies, and speaking/articulation rate. Preliminary analysis suggests that truth-tellers and liars cannot be differentiated based on these speech parameters. Not only was there a lack of significant changes for the majority of parameters investigated but also, if change was present it failed to reveal consistencies within and between speakers. The remarkable amount of inter and intra-speaker variability underlines the fact that deceptive behavior is individualized and very multifaceted. As well as providing a basis for future research programs, the present study should encourage researchers and practitioners to evaluate critically what is (im)possible using auditory and machine based analyses with respect to detecting deception from speech.

3:05

4pSCa6. Progress toward a forensic voice data format standard. James L. Wayman (Office of Graduate Studies and Res., San Jose State Univ., San Jose, CA 95192-0025), Joseph P. Campbell, Pedro Torres-Carrasquillo (MIT Lincoln Lab., Lexington, MA 02420-9108), Peter T. Higgins (Higgins and Assoc., Int., 3900-A, #7E, Watson Pl. NW, Washington, DC 20016), Alvin Martin (Information Access Div. Natl. Inst. of Standards and Technol., Gaithersburg, MD 20899-8940), Hirotaka Nakasone (Operational Technol. Div. Federal Bureau of Investigation, Quantico, VA 222135), Craig Greenberg, and Mark Pryzbocki (Information Access Div. Natl. Inst. of Standards and Technol., Gaithersburg, MD 20899-8940)

The de facto international standard for the forensic exchange of data for biometric recognition is ANSI/NIST ITL-1/2, “Data Format for the Interchange of Fingerprint, Facial, and Other Biometric Information.” This format is used by law enforcement, intelligence, military, and homeland security organizations thoughout the world to exchange fingerprint, face, scar/mark/tatoo, iris, and palmprint data. To date, however, there is no provision within the standard for the exchange of audio data for the purpose of forensic speaker recognition. During the recent 5-year update process for ANSI/NIST ITL-1/2, a consensus decision was made to advance a voice data format type under the name “Type 11 record.” Creating such an exchange format type, however, is far from straight forward—the problem being not the encoding of the autio data, for which many accepted standards exist, but rather in reaching a consensus on the metadata needed to support the varied mission requirements across the stakeholder communities. In this talk, we’ll discuss the progress that has been made to date, the questions that remain, and the requirements for additional input from the broader stakeholder communities.

Session 4pSCb

Speech Communication: Prosody, Articulation, and Articulatory Modeling (Poster Session)

Meghan Sumner, Chair
Dept. of Linguistics, Stanford Univ., Stanford, CA 94305-2150

Contributed Papers

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.

4pSCb1. Reduction of consonants and vowels in the course of discourse. Michael McAuliffe and Molly Babel (Dept. of Linguist., Univ. of British Columbia, 2613 West Mall, Vancouver, BC V6T 1Z4, mcauliff@interchange.ubc.ca)

There is a clear link between the discourse status of a word and the degree of reduction. For instance, Gregory [Dissertation (2002)] provided evidence that hearer knowledge affected reduction in production for discourse-old items. Lexical information, such as word frequency, also plays a crucial role in the degree of reduction [Fosler-Lussier and Morgan, Speech Commun. (1999)]. These previous studies either looked at short discourses or words isolated from context. Therefore, the current study investigates longer discourses, using the VIC Corpus [Pitt *et al.*, Corpus (2007)]. The primary question is to what degree do repeated uses cause further reductions, and if the reductions are syllabically and segmentally uniform across a word. Many studies on reduction rely on the intuition that reductions occur when information load is light, such as when the word was repeated recently or has a high probability of occurrence, so the prediction is that unstressed syllables would show more reduction than stressed syllables, due to their lighter load of information. Likewise, as vowels vary more than consonants across dialects, the informational load of vowel quality may be less than that of consonant quality, so the prediction is that vowels would show greater reductions than consonants.

4pSCb2. Automatic analysis of constriction location in singleton and geminate consonant articulation using real-time magnetic resonance imaging. Christina Hagedorn, Michael Proctor, Louis Goldstein, and Shrikanth Narayanan (Dept. of Eng., USC, Los Angeles, CA 90089, shri@sipi.usc.edu)

Research on geminate consonants has attempted to establish whether the control of their articulation differs from that of corresponding singletons in temporal parameters, spatial parameters, or both. One piece of evidence supporting spatial control in Italian geminates is EPG results revealing that the location of maximal constriction (CL) of coronal geminates along the palate exhibits small differences from the CL of singletons (Payne 2005). However, our recent work investigating Italian using real-time magnetic resonance imaging (MRI) has shown that when measuring CL dynamically, the CL of singletons and geminates are identical. Dynamic CL is defined as the region of the image that exhibits maximum intensity change during constriction formation and release. These results can be reconciled with the EPG findings if we hypothesize that differences in CL at the moment of maximal constriction are due to compression effects (in the longer geminates) involving the tongue tip sliding along the palate during the closure duration. We evaluate this hypothesis by testing whether CL differences between singletons and geminates can also be found in MRI data, when CL is measured statically (region on the palate contacted in the most constricted frame) in the same utterances in which dynamic CL is invariant. [Work supported by NIH.]

4pSCb3. Development of speech motor control: Lip movement variability. Anders Lfqvist (Dept. Logopedics, Phoniatrics and Audiol., Lund Univ., Lund, Sweden), Johan Frid, and Susanne Schötz (Humanities Lab., Lund Univ., Lund, Sweden)

This study examined variability of lip movements across repetitions of the same utterance as a function of age in Swedish speakers. Subjects were 37 typically developed Swedish children and adults (19 females, 18 males, aged 5–31 yr). Lip movements were recorded during 15–20 repetitions of a short Swedish phrase using articulography, with a sampling rate of 200 Hz. After correction for head movements, the kinematic records were expressed in a maxilla-based coordinate system. Movement onset and offset of the utterance were identified using kinematic landmarks. The Euclidean distance between receivers on the upper and lower lips was calculated and subjected to functional data analysis [Ramsay *et al.*, J. Acoust. Soc. Am. **99**, 3718–3727 (1996)] to assess both temporal and spatial variabilities. Results show a decrease in both indices as a function of age, with a greater reduction of amplitude variability. [Work supported by grant 349-2007-8695 from the Swedish Research Council.]

4pSCb4. Imaging and quantification of glottal kinematics with ultrasound during speech. Benjamin Parrell (Dept. of Linguist., Univ. of Southern California, Grace Ford Salvatori 301, Los Angeles, CA 90089), Adam Lammert (Univ. of Southern California, Dept. of Comput. Sci. 3740 McClintock Ave., Rm. 400, Los Angeles, CA 90089), Louis Goldstein, Dani Byrd (Dept. of Linguist., Univ. of Southern California, Grace Ford Salvatori 301, Los Angeles, CA 90089), and Shrikanth Narayanan (Dept. of Elec. Eng., Univ. of Southern California, Los Angeles, CA 90089)

Most examinations of glottal abduction and adduction during speech have employed laryngoscopic video or transillumination (Hoole, 2006). While these provide accurate information about timing of glottal movements, they are invasive and cannot provide absolute measurements about glottal width. At the same time, recent medical studies have used ultrasonic imaging to accurately capture glottal movements and laryngeal anatomy [Hu, J. Ultrasound Med. (2010); Jadcherla *et al.*, Dysphagia (2006)]. We demonstrate novel methods of using ultrasound to measure both temporal and spatial aspects of glottal movements during speech. While previous work on glottal ultrasound has been limited by the need to manually analyze each acquired frame, we present methods to automatically quantify glottal aperture in ultrasound images. Finally, since glottal ultrasound does not interfere with the acquisition of supra-laryngeal articulatory data, we present the results of preliminary experiments that record laryngeal and supra-laryngeal speech movements simultaneously using ultrasound concurrently with electromagnetic articulometry. This allows the analysis of relative timing between movements in the two systems as well as examination of changes in timing or magnitude due to variables such as prosodic structure or speech rate. [Work supported by NIH.]

4pSCb5. Detailed study of articulatory kinematics of critical articulators and dependent articulators of emotional speech. Jangwon Kim (Univ. of Southern California, 3740 McClintock Ave., EEB 400, Los Angeles, CA 90089, jangwon@usc.edu), Sungbok Lee (Univ. of Southern California Los Angeles, CA 90089, sungbokl@usc.edu), and Shrikanth Narayanan (Univ. of Southern California, Los Angeles, CA 90089)

This study investigates the articulatory kinematics of critical articulators and dependent articulators as a function of emotion. Our hypothesis is that critical articulators and dependent articulators are utilized differently for achieving distinctive emotion goals that overlay linguistic goals. For example, speakers may use variability of dependent articulators distinctively to that of critical articulators in achieving emotion goals. Distinctive articulatory movements for different emotions have been observed (Lee *et al.* 2005). This study uses a database of three speakers (2 female and 1 male) collected with electromagnetic articulography to collect kinematic information. Articulatory trajectories are aligned by dynamic time warping. Linguistically identical syllable-level segments are analyzed based on detailed aspects of articulatory movements (e.g., position, velocity and phase), after sampling with a 20-ms window and 10-ms shifting. The emotion-specific patterns that emerge for critical (i.e., goal-directed) articulators are compared to those for articulators that are not under overt control for a particular phone. [Work supported by NIH and NSF Grants.]

4pSCb6. Statistical estimation of speech kinematics from real-time MRI data. Adam Lammert, Vikram Ramanarayanan (Univ. of Southern California, 3740 McClintock Ave., Rm. 400, Los Angeles, CA 90089), Louis Goldstein, Khalil Iskarous (Univ. of Southern California, Los Angeles, CA 90089), Elliot Saltzman (Boston Univ., Boston, MA 02215), Hosung Nam (Haskins Labs., New Haven, CT 06511), and Shrikanth Narayanan (Univ. of Southern California, Los Angeles, CA 90089)

The human speech production system can be fundamentally characterized by the kinematic relationships between low-level articulator variables and relatively high-level tasks. Such kinematics can be illuminating about many system aspects from degrees of freedom and redundancy to dynamics and even control. Since these relationships are generally complex and infeasible to express in closed form, recent work has focused on statistical methods for estimating the relevant relationships from data (Saltzman, 2006, in Dynamics of Speech Production and Perception; Lammert, 2010, Proc. INTERSPEECH). Such methods have been applied to synthetic speech data in order to evaluate their effectiveness, but they have not yet been demonstrated on real data. Here, we apply these methods to real speech data acquired from real-time magnetic resonance imaging. We extract articulator variables that are consistent with those chosen by various articulatory models, and we relate them to high-level task variables such as constriction locations and degrees and formant frequencies. This is done to facilitate an analysis of articulatory motor control during speech production. [Work supported by NIH.]

4pSCb7. Semi-automatic modeling of tongue surfaces using volumetric structural MRI. Daniel K. Bone, Michael I. Proctor, Yoon Kim, and Shrikanth S. Narayanan (Univ. of Southern California, Signal Analysis and Interpretation Lab., Los Angeles, CA 90089)

Although volumetric magnetic resonance imaging has proven to be a valuable tool in the study of consonant production (Narayanan *et al.*, 1995; Kröger *et al.*, 2000), its utility is limited by the difficulty and laboriousness of reliably extracting tissue boundaries from imaging data. Current methods typically involve manual segmentation of air-tissue boundaries (e.g., Birkholz, 2006). Conventional automated (Atkins, 1998) and semi-automated (Ashton *et al.*, 1995) methods used for the segmentation of brain MRI datasets may not be directly applicable to lingual segmentation because they are designed to work with different anatomical features. We present a method for extracting tongue surfaces from hi-resolution volumetric MRI data with limited user intervention. For each vocal tract volume to be analyzed, a lingual bounding box and search seed was first specified by an expert user, and voxel intensity was normalized across the region of interest. Lingual surfaces were automatically identified using a multi-pass region-growing algorithm operating over coronal planes. Thresholding was performed asymmetrically to allow for differential detection of air, teeth, and palatal boundaries, in opposition to adjacent lingual tissue. Smoothed tongue surfaces were fit to the resulting volumes by incorporating prior knowledge of intrinsic lingual musculature. [Work supported by NIH.]

4pSCb8. Enhancement of laryngeal features under segmental and prosodic conditioning. Indranil Dutta (Dept. of Computational Linguisitics, EFL Univ., Tarnaka, Hyderabad, Andhra Pradesh, 500007 India)

Evidence in support of enhancement features (Keyser and Stevens, 2006) is presented from an acoustic study of laryngeal contrasts in Hindi. Segmental contrasts where defining features and their corresponding acoustic outcomes are attenuated are said to be made acoustically salient by enhancing gestural features (Stevens and Keyser, 2010). In this study, the four-way laryngeal contrasts in Hindi are examined under varying prosodic and segmental contexts. In segmental contexts where a defining distinctive feature (slack vocal folds) is attenuated in its acoustic manifestation, namely, closure duration (CD), an enhancing gesture (spread glottis) (Avery and Idsardi, 2001) is added to increase the saliency of contrasts between voiced aspirated stops (VAS) and voiced stops (VS). The acoustic consequence of this enhancement is f_0 lowering in the following vowel. Similar results are obtained under weak prosodic conditions where both f_0 lowering and increase in spectral tilt result from the enhancing gesture (spread glottis). In addition, in segmental contexts where gestural overlap compromises the feature (slack vocal folds), (spread glottis) by way of f_0 lowering in the acoustic dimension enhances the laryngeal contrast between VAS and VS. These results lend support to the theory of enhancement as proposed by Stevens and Keyser (2010).

4pSCb9. Statistical analysis of constriction task and articulatory posture variables during speech and pausing intervals using real-time magnetic resonance imaging. Vikram Ramanarayanan, Louis Goldstein, Dani Byrd, and Shrikanth Narayanan (Univ. of Southern California, Electrical Eng., Los Angeles 90089)

We have shown previously using real-time magnetic resonance imaging data (Ramanarayanan *et al.*, 2011, ISSP Montreal) that it is more likely that articulatory posture variables (such as jaw angle) are controlled to achieve articulatory settings during pausing intervals in read speech than constriction task variables (such as lip aperture, tongue tip constriction degree, etc.). In this study, we extend this work to examine correlations between constriction task variables and articulatory posture variables during both speech and pause intervals. This work serves to deepen our understanding of the differences in postural motor control of the vocal tract observed during speaking and pausing. [Work supported by NIH.]

4pSCb10. The scope of phrasal lengthening: Articulatory and acoustic evidence. Daylen Riggs and Dani Byrd (Dept. of Linguist., USC, 3601 Watt Way, GFS 301, Los Angeles, CA 90089)

The temporal lengthening that occurs at phrase edges is known as phrasal lengthening. The scope of phrasal lengthening refers to the distance both before and after a phrase edge that phrasal lengthening can occur. This paper examines the influence of prosodic prominence on the scope of phrasal lengthening in articulation. Pitch accent was placed immediately adjacent to the phrase edge and at varying distances before and after the phrase edge. Articulatory durations of gestures were measured in the pitch-accented syllables and in the syllables intervening between the phrase edge and the pitch-accented syllables. Acoustic measurements of consonant, syllable, and vowel duration were also examined. These constriction durations were compared to those for gestures in phonologically parallel control sentences that lacked a phrase boundary. The results indicate that phrasal lengthening is most systematic immediately adjacent to the phrase edge. However, pitch accent can attract phrasal lengthening. One subject showed phrasal lengthening in a pitch-accented syllable, three syllables away from the boundary. Finally, when the remote pitch-accented syllable showed phrasal lengthening, gestures intervening between the phrase edge and the pitch-accented syllable also showed phrasal lengthening. These patterns are evaluated in the context of the prosodic gestural model of Byrd & Saltzman (2003).

4pSCb11. An ultrasound study of Canadian French rhotic vowels. Jeff Mielke (Arts 401, 70 Laurier Ave East, Ottawa, ON K1N6N5, Canada, jmielke@uottawa.ca)

Some speakers of Canadian French produce the vowels /ø/ and /œ/ with a rhotic perceptual quality, leading *pneu*, *docteur*, and *brun* to sound like /pnr/, /dOktarR/, and /bRr/. English /r/ can be produced with a variety of tongue shapes (including bunched and retroflex variants; Delattre and

Freeman 1968, etc.), raising the question of whether French rhotic vowels are also produced with categorically different tongue shapes. Mid-sagittal ultrasound video was recorded for three native speakers of Canadian French producing words containing /œ/, /œR/, and /ø/ in a carrier phrase. Acoustic analysis of rhotic- and non-rhotic-sounding vowels reveals that the rhotic perceptual quality is associated with a low third formant, which is an important acoustic cue for English /r/. Two of the subjects produced rhotic vowels with a tip-down bunched tongue shape closely resembling Delattre and Freeman's type 4 and one subject produced them with a retroflex tongue shape. The study provides articulatory data on retroflex vowels, which are typologically rare, and on bunched-retroflex variation in a language other than English. The parallel cases of articulatory variability in French and English raise opportunities for investigating articulatory-acoustic mapping in bilinguals.

4pSCb12. Can you say [ṽ] or [x̃]? Aerodynamics of nasalized fricatives in Scots Gaelic. Daniel Brenner, Andrea Davis, Natasha Warner, Andrew Carnie, Muriel Fisher, Jessamyn Schertz, Michael Hammond, and Diana Archangeli (Dept. Ling., Univ. of Arizona, Box 210028, Tucson, AZ 85721-0028, dbrenner@u.arizona.edu)

Scots Gaelic is said to have nasalized fricatives ([ṽ, x̃, h̃] etc.). Nasalized fricatives may be aerodynamically impossible (except [h̃]), because opening the velum would vent the high oral pressure required for frication (Ohala 1975). However, phonologically nasalized fricatives which are realized some other way phonetically are possible, and [h̃] is possible because the frication occurs behind the velic opening. The current work presents oral and nasal airflow data from 14 native speakers of Scots Gaelic, primarily from the Skye dialect, to investigate the nasalized fricative distinction. Results indicate that the most common solution to the aerodynamic problem is to neutralize the distinction: most phonological nasalized fricatives, from most speakers, are simply not nasalized at all ([v] for [ṽ]. Some tokens show nasalization during the preceding vowel ([ãv] for [aṽ]). Some tokens in which the expected fricative is pronounced as an approximant (common in Scots Gaelic) show nasalization ([w̃] for [ṽ]). Furthermore, [h̃] occurs. Very rarely, there may be slight nasalization overlapping part of an oral fricative. Thus, the data show that conflicting aerodynamic demands are resolved variably: in this dialect, the distinction is usually neutralized; when it is not, several alternatives that avoid the aerodynamic conflict are produced.

4pSCb13. An articulatory and acoustic investigation of Mandarin apical vowels. SANGIM LEE (Dept. of Linguist., New York Univ., 10 Washington Pl. NY 10003, sangim@nyu.edu)

The present study investigates the articulatory and acoustic properties of apical vowels in Mandarin Chinese using ultrasound and acoustic measures. Instead of the high front vowel, dental and retroflex sibilants are known to be followed by apical vowels made with tongue tip/blade, rather than the tongue body. The ultrasound images of one Beijing speaker confirmed that the tongue tip/blade gesture indeed remains unreleased during the production of these vowels. However, it also revealed that the tongue body for apical vowels is significantly retracted with a relative order of the vowel following dentals being more retracted than the vowel following retroflexes. This gave rise to a consistent acoustic consequence; the second formant of the vowel after dentals of six Mandarin speakers was significantly lower than that of the vowel after retroflexes. In addition, the third formant of the vowel after a retroflex was significantly lowered and nearly merged with the second formant, indicating that the vowel is also strongly retroflexed. The correct analysis is thus claimed to include the apical gestures as secondary features added to the main articulation of the tongue body as in the following notation: a dentalized back vowel and a retroflexed mid vowel.

4pSCb14. Diphthong centralization and reduction in constriction degree. Fang-Ying Hsieh, Louis Goldstein, and Khalil Iskarous (Dept. of Linguist., Univ. of Southern California, Grace Ford Salvatori 301, Los Angeles, CA 90089-1693, fangyinh@usc.edu)

Diphthongs in Mexian Hakka Chinese have been divided into two categories (Cheung 2007): Endpoints of the falling (in vowel space) diphthongs have formant values equivalent to those of the monophthongal vowels, while the endpoints of the rising diphthongs generally have more centralized formants. This categorization provides a chance to test the hypothesis of Iskarous *et al*. (2010) that in speech production the location of constriction (CL) changes discretely in transitions between successive targets, while the degree of constriction (CD) changes continuously. We therefore hypothesize that the centralization in these diphthongs results from a reduction in CD (possibly due to undershoot) rather than CL, thus making it parallel to common consonant reduction changes in languages, such as spirantization of stops. The current study tests this hypothesis using TADA (TAsk-Dynamic) modeling, in which CD and CL in speech production can be manipulated. Diphthongs [ia], [ua], [ai], and [au] were modeled, and the preliminary results indicate that a change in target CD alone yields formant patterns that are more similar to those reported compared with a change in CL. Results of additional studies will be presented that analytically derive CD and CL in these diphthongs from new data collected from speakers of Hakka.

4pSCb15. When inferring lingual gestures from acoustic data goes wrong: The case of high vowels in Canadian French. Will Dalton (Dept. of Linguist., Univ. of Ottawa, 70 Laurier East, Ottawa, ON K1N6N5, wdalt017@uottawa.ca)

Canadian French (CF) is distinguished from other dialects partly by the presence of lax high vowel allophones in closed syllables (Walker, 1984). Acoustically, tense high vowels are characterized by a lower F1 than their lax counterparts, which could be the result of tongue root advancement, tongue body raising, or both (Ladefoged & Maddieson, 1996). High vowel allophony in CF therefore represents a case in which articulatory gestures cannot reliably be inferred from acoustic data alone. Nevertheless, the literature discussing the phonetic properties of high vowels in CF commonly assumes tongue root position to be the parameter that distinguishes between tense and lax vowels, despite an absence of empirical evidence. The purpose of this experiment is to test this assumption and provide articulatory evidence using ultrasound imaging to examine tongue position during speech production by CF speakers. Results indicate that an advanced tongue root gesture is not used to distinguish between high vowels in CF; no significant difference in tongue root position was found between tense and lax allophones. Rather, tongue body height was found to be the distinguishing feature. These findings contribute to our knowledge of the typology of articulatory gestures used to distinguish between so-called tense and lax vowels.

4pSCb16. Identifying relevant analysis parameters for the classification of vocal fold dynamics. Daniel Voigt (Dept. of Linguist., Max Planck Inst. for Evolutionary Anthropology, Deutscher Platz 6, D-04103 Leipzig, Germany, daniel_voigt@eva.mpg.de) and Ulrich Eysholdt (Univ. Hospital Erlangen, Bohlenplatz 21, D-91054 Erlangen, Germany)

In the previous work, a computer-based analysis framework was proposed, which is capable of objectively and automatically classifying vocal fold vibrations as captured by high-speed videoendoscopy during phonation. The method is based on quantitative feature extraction from Phonovibrograms combined with nonlinear machine learning techniques, allowing for the discrimination of normal and pathological laryngeal movement patterns. The diagnostic reliability and potential of this analysis approach were demonstrated. However, the practically relevant question, whether certain control parameters of the procedure can lead to increased classification accuracy, remained partially unanswered. In this study, the following parameter sets of the analysis framework were investigated in a systematic manner: method of feature extraction, type of feature aggregation and normalization, number of considered oscillation cycles, feature laterality, classification task, and employed machine learning algorithm. For this purpose, more than 150 000 experiments were conducted using a data set of 105 laryngeal high-speed video recordings, comprising various clinical cases with non-organic findings and subjects from a healthy control group. The results of this extensive study show the particular suitability of certain parameter combinations, helping to further improve the practical application of the automated classification framework for vocal fold dynamics.

4pSCb17. A two-dimensional/three-dimensional hybrid structural model of the vocal folds. Douglas Cook, Pradeep George, and Margaret Julias (Div. of Eng., New York Univ. Abu Dhabi, Abu Dhabi, United Arab Emirates)

The spatial dimensionality of the vocal fold vibration is a common challenge in creating parsimonious models of vocal fold vibration. The ideal model is one that provides acceptable accurate with the lowest possible computational expense. Inclusion of full three-dimensional (3D) flow and structural vibration typically requires massive amounts of computation, whereas reduction of either the flow or the structure to two dimensions eliminates certain aspects of physical reality, thus making the resulting models less accurate. Previous two-dimensional (2D) models of the vocal fold structure have utilized a plane strain formulation, which is shown to be an erroneous modeling approach since it ignores influential stress components. We herein present a 2D/3D hybrid vocal fold model that preserves three-dimensional effects of length and longitudinal shear stresses, while taking advantage of a two-dimensional computational domain. The resulting model exhibits static and dynamic responses comparable to a 3D model and retains the computational advantage of a two-dimensional model.

4pSCb18. Nonlinear viscoelastic properties of human vocal fold tissues under large-amplitude oscillatory shear. Elhum Naseri, Mindy Du, and Roger W. Chan (Otolaryngol.—Head and Neck Surgery, Biomedical Eng., Univ. of Texas Southwestern Medical Ctr., Dallas, TX 75390-9035)

Traditionally, viscoelastic shear properties of human vocal fold tissues have been described by the linear viscoelastic moduli G' and G" at small strain amplitudes. However, once the mechanical behavior becomes nonlinear these moduli are no longer sufficient for viscoelastic characterization. MITlaos, a rheological framework developed for better describing such behavior, was used to characterize the nonlinear viscoelastic properties of the vocal fold lamina propria when subjected to large-amplitude oscillatory shear (LAOS). MITlaos involved Fourier transform processing to convert raw stress–strain signals into a filtered/smoothed total stress signal based on the odd-integer harmonic components, with nonlinear viscoelastic parameters simultaneously derived. Rheometric testing of vocal fold cover specimens was performed with increasing strain amplitudes using a controlled-strain, simple-shear rheometer. Smoothed Lissajous–Bowditch curves generated from MITlaos were plotted in Pipkin space, and the nonlinear analysis was summarized by variations of rheological fingerprints. Pipkin diagrams served as a geometric means for identifying the onset of nonlinear behavior, and any distortions due to noise. Results showed that the human vocal fold cover when subjected to LAOS demonstrated intracycle strain stiffening and intercycle strain softening with increasing strain amplitude, as well as shear thinning with increasing strain-rate amplitude. [Work supported by NIH.]

4pSCb19. Recovery of articulatory information from acoustics using linear mixed effects models. Sean Martin (Dept. of Linguist., New York Univ., 10 Washington Pl. New York, NY 10003)

The current study attempts to establish a lower bound on the degree to which articulatory information can be recovered from the speech signal. The recovery of articulatory information is modeled with comparatively simple linear mixed-effects models using synchronized articulatory and acoustic data from the University of Wisconsin X-Ray Microbeam Speech Production Database (Westbury, 1994). The performance of this model is compared with other speech inversion models. While previous work [Mitra *et al.* (2010) and others] demonstrates that a more sophisticated model can perform speech inversion quite accurately, the results here show that even a strongly constrained linear model performs better than might be expected given that the well-known non-linearity of the acoustic-articulatory mapping. More detailed investigation of the results indicates that while non-linearity in the acoustic-articulatory mapping affects the recovery of articulatory information for all tongue regions, it contributes considerably more to error in recovering tongue-tip position. For the tongue body, the correlation between predicted and observed tongue body position is approximately 0.9 while the correlation for tongue tip position was approximately 0.65. A linear approximation seems to be sufficient to recover tongue body position with average error of approximately 12%, even with limited acoustic information.

4pSCb20. Emotion effects on speech articulation: Local or global? Sungbok Lee, Jangwon Kim, and Dany Byrd (Dept. of EE, Univ. of Southern California, Los Angeles, CA 90089, sungbokl@usc.edu)

Although differences in overall articulatory behaviors in emotional speech production have been well acknowledged in the literature, it is not well known whether such articulatory differences are distributed or localized along the time axis. In this study, we examine the lower lip and tongue tip trajectories of a perceptually evaluated EMA (electromagnetic articulography) emotional speech database of one male and two female subjects. The speech material is a set of sentences repeated at least three times, and the emotion types investigated are neutral, angry, and happy. Timing differences between repetitions as well as across emotions are normalized by the FDA (functional data analysis, Ramsay and Silverman, 1997) time- alignment technique so that important kinematic land-marks (e.g., velocity maxima) are aligned with respect to each other. Therefore, not only differences in timing but also differences in movement range and velocity can be investigated along the time line. The normalized quantities are sampled along the time line every 10 ms, with a 20-ms window, and their differences along the time line are compared as a function of emotion. Final results as well as their relation to pitch patterns and other acoustic measures will be reported. [Work supported by NIH and NSF.]

4pSCb21. Suprasegmental features in Northern Paiute. Joseph D. Brooks (josephdbrooks@umail.ucsb.edu)

The relationship between intonation and word-level stress in Northern Paiute (Western Numic; Uto-Aztecan) is described in order to support the level of phonology as a basis for the lexical, syntactic, and ideational integration of connected speech into intonation units (IUs). Word-level stress in Northern Paiute, intrinsic and in most cases peninitially fixed, is manifested by vowel lengthening and a rise in pitch. Intonation depends largely on the word-level pitch rises, minimalizing the frequency range for non-tonic accents and increasing it for most rising tonic accents of most IUs. The opposite pattern also occurs, whereby non-stressed syllables of individual words have an intonation pattern superimposed upon them; most tonic accents with falling pitch and a few with rising pitch over a non-stressed syllable distinguish this second category as independent from word-level stress. Although these two patterns can explain how suprasegmental features integrate words into larger, cohesive units, the deictic word class is an exception. Deictics in Northern Paiute lack intrinsic pitch, have extra-long vowels, and occur overwhelmingly in IU-final position. It is proposed that suprasegmental features serve as a cohesive force between the areas of linguistic structure by varying degrees of phonological integration.

4pSCb22. The prosodic structure of Koasati. Matthew Gordon (Dept. of Linguist., UC Santa Barbara, Santa Barbara, CA 93106, mgordon@linguistics.ucsb.edu), Linda Langley (McNeese State Univ., Lake Charles, LA 70609), and Jack Martin (College of William & Mary, Williamsburg, VA 23187)

This paper presents results of the first systematic acoustic investigation of the prosodic system of Koasati, an endangered Muskogean language of Louisiana. Koasati is prosodically complex, featuring lexically marked pitch accents in nouns and verbs, as well as phonologically predictable pitch accents and boundary tones assigned by the intonational system. While most nouns are characteristically associated with a final F0 rise attributed to a phrase-level tone, some have an additional lexically marked high pitch accent on a non-final syllable. The pitch accent system in verbs is more intricate and consists of different types of aspectually determined accents (termed 'grades' in the Muskogeanist literature) and concomitant segmental changes including lengthening, nasalization, or aspiration. Koasati also has a rich intonation system featuring at least two levels of prosodic constituents, the accentual phrase and the intonational phrase, which are associated with boundary tones and/or pitch accents that often temporally overlap with lexically specified pitch accents. Our results will be compared with those for Creek (Johnson and Martin 2001, Martin and Johnson 2002) and Chickasaw (Gordon 1999, 2003, 2004, 2005, 2007, Gordon *et al.* 2000, Gordon and Munro 2007), Koasati's prosodically best described relatives. [Work supported by NSF.]

4pSCb23. Effects of morpheme boundaries in /n/ palatalization in Korean. Jae-Hyun Sung (Dept. of Linguist., Univ. of Arizona, 1100 E Univ. Blvd, Tucson, AZ 85721, jhsung@u.arizona.edu)

Korean palatalization is known to be one of the rare palatalization phenomena where the morpheme boundary plays a role. For example, /mat+i/ 'the elderly', where a morpheme boundary exists between /t/ and /i/, becomes [madi] by the palatalization process, while a tautomorphemic word /mati/ 'joint' is realized as [madi] (*[madi]), with no palatalization. The present study investigates whether the effect of morpheme boundaries is

also at play in allophonic /n/ palatalization in Korean as well as in /t/ palatalization using ultrasonic analysis, and shows that morpheme boundaries contribute to allophonic variation. This study uses ultrasound imaging and audio recordings of 6 native speakers of Korean to examine the difference in palatalization between two morphologically different words in Korean—tautomorphemic words (e.g., [koni] 'swan') and heteromorphemic words (e.g., [muni] (/mun+i/) 'door + nominative marker'). Comparison of the degree of palatalization before [i] shows there is stronger /n/ palatalization in tautomorphemic /n/ + [i] words than that in heteromorphemic /n/ + [i] words, in contrast to /t/ palatalization. The results from this study are in line with the reported language-universal tendency for greater palatalization in tautomorphemic environments.

4pSCb24. Louder is longer: Amplitude conditioned lengthening in diasporic Siraiki. Aditi Arora (Dept. of Phonet. and Spoken English, EFL Univ., Tarnaka, Hyderabad, Andhra Pradesh, India 500007) and Indranil Dutta (EFL Univ., Tarnaka, Hyderabad, India 500007)

Based on a longitudinal study of three generations of diasporic Siraiki speakers in India, it is shown that first generation (Gen1) implosives develop into lengthened plosives, intervocalically, in the subsequent two generations (Gen2/Gen3). Historically, Prakrit geminates developed into Siraiki and Sindhi implosives. Ohala (1992a) observes that maintaining voicing during intervocalic geminates is facilitated by increased oral cavity volume, achieved by lowered larynx (LL). The consequence of this cavity enlargement is listener misperception of the accompanying cues that lead to the genesis of implosives in the intervocalic position in Sindhi (Ohala, 1993). Closure voicing durations, root mean square (rms) amplitude during closure, and f_0 following release are measured in this study. It is observed that in contexts where the Gen1 speakers produce increased rms amplitude at the release of implosives, the subsequent Gen2/Gen3 produce lengthened plosives. Increased peak-amplitude has been shown to be a weak cue for initial voiceless long consonants in Pattani Malay (Abramson, 1998). Based on the evidence from this study and Pattani, it is argued that the acoustic consequence of the feature LL, mainly increased amplitude, allows a Gen2/Gen3 mapping of the Gen1 implosives onto lengthened plosives. This mapping suggests that in addition to listener misperception, sound changes result from articulatory-acoustic mismatch.

4pSCb25. Phonetic vowel hiatus in Spanish. Erika Varis (Dept. of Linguist., Univ. of Southern California, Grace Ford Salvatori 301, Los Angeles, CA 90089, varis@usc.edu)

Phonologically, Spanish tolerates sequences of adjacent vowels at a word boundary, but coalescence and deletion have been impressionistically identified at this juncture as phonetic resolution strategies (Alba, 2006). In other languages, phonetic measurements of vowels in hiatus show gradient distinctions between different prosodic environments, supporting a gestural overlap analysis that produces the perception of deletion or coalescence (for example, Igbo, Catalan, and Greek). The current experimental study investigates phonetic manifestations of vowel hiatus in four prosodic environments (IP, PP, Pwd, and clitic) in Spanish to determine whether hiatus resolution is accomplished by vowel deletion or gestural overlap. The study was conducted with nine native speakers of Peninsular Spanish. The duration and formant frequencies of vowel sequences was measured in read sentences that manipulated boundary strength separating the two vowels. Higher-level prosodic boundaries IP and PP produced no significant hiatus effects, but vowels separated by lower-level Pwd and clitic boundaries showed formant influences of one vowel on the other. The durations of the two-vowel sequences were longer than singleton controls across all conditions, indicating no vowel deletion. The results support an analysis of vowel gesture overlap and reject deletion, contrary to auditory impressions but consistent with the findings in other languages.

4pSCb26. Acoustic cues to prominence in children's speech. Irina A. Shport and Melissa A. Redford (Dept. of Linguist., Univ. of Oregon, Eugene OR 97403-1290)

Previous work indicates that phrasal prominence patterns are not yet adult-like in 6-year-old children's speech. Perception of prominence in English is influenced by temporal patterning, amplitude changes, and fundamental frequency variations across the phrase. This study examined the relative weightings of these cues to prominence in adult judgments of 6-year-old children and college-aged adults' speech. Eleven adult judges listened to two—word phrases produced by 25 children and 25 adults in a counting task designed to induce stress shift (*thirtéen banana* versus *thirtéen barbeque*). The judges decided which word was most prominent in the phrase: number, noun, or equal prominence. Although the number word was always judged to be prominent regardless of speakers' ages or context, initial results indicate that agreement between judges varied systematically with different cues in children and adults' speech. More judges agreed that the number word was prominent in children's speech when the first syllable of this word was produced with an especially high F0 and long duration, but only the relative intensity of the initial syllable predicted inter-judge agreement for adult speech. The results have implications for understanding the development of cue integration in the prominence production. [Work supported by NIH/R01HD061458.]

4pSCb27. Prosodic pattern detection based on fuzzy set theory. Nicholas Bacuez (Dept. of French and Italian, Univ. of Texas, 1 Univ. Station B7600 Austin, Texas 78712-0224, nicholasbacuez@gmail.com)

This research utilizes fuzzy logic and fuzzy set theory to experimentally abstract the mental representations of prosodic contours otherwise not accessible to speakers or researchers. I have developed a pattern recognition system to extract approximations of these representations and their range of variation from speech datasets. Following the principles of fuzzy set theory, a prosodic contour is defined as a category and the variation range of its components is expressed in terms of degree of membership to the category, from 0 (excluded) to 1 (included). Subsequently, the system analyzes the prosodic contour of new sentences by assessing through fuzzy matching their degree of membership to previously identified categories: 1 is perfect, 0.5 is borderline, and 0.1 is improbable. In the current study, I have used closed questions obtained experimentally from native French speakers. The prosodic contour of closed questions is non problematic and ensures the controllability of the output. The model output is an array storing the ideal curve and its degrees of variation. Fuzzy matching reveals that most closed questions from the dataset are in medium range (0.5). This materializes the idea that vagueness (Russell, Wittgenstein), the linguistic correlate to fuzzy logic, is a natural property of language.

4pSCb28. Pragmatically determined variation in Greek wh-question intonation. Stella Gryllia (Univ. Potsdam, Inst. fuer Linguistik/Allgemeine Sprachwissenschaft, Komplex Golm, Haus 35, 0.02, Karl-Liebknecht-Str. 24-25, D-14476 Golm, Germany, gryllia@uni-potsdam.de), Mary Baltazani (Univ. of Ioannina, Ioannina 45110, Greece), and Amalia Arvaniti (UC San Diego, La Jolla, CA, 92093-0108)

This paper presents production data testing the analysis of Arvaniti and Baltazani (2005) and Arvaniti and Ladd (2005) according to which the default melody used with Greek wh-questions is L*+H L- !H% (showing a delayed accentual peak on the utterance-initial wh-word, a low stretch, and a final curtailed rise), with !H% sometimes being replaced by L%. Here it was hypothesized that the melodies also differ in pitch accent and are used in different contexts. Four speakers, two male and two female, took part in reading a varied corpus of questions in contexts that lead to the use of a wh-question either in order to seek information or in order to politely register disagreement (a function of wh-questions peculiar to Greek). Our results confirmed that there are two different melodies: L*+H L- !H%, with a delayed accentual peak and a final rise, and L+H* L- L%, with an early peak and no final rise. The former is used for requesting information and the latter when questions function as dissenting statements. In addition to leading to a revision of the existing analysis, these results show that distinctions such as statement versus question are too coarse-grained for the analysis of intonational meaning and function.

4pSCb29. Acoustic correlates of narrow focus in Turkish. Canan Ipek (Dept of Linguist., Univ. of Southern California Grace Ford Salvatori 301 Los Angeles, CA 9008, ipek@usc.edu9-1693)

Acoustic correlates of narrow focus in Turkish focus is known to affect the acoustic properties of the lexical element under focus. Its effect on the pre-focus and post-focus domains has attracted much less attention. This study aims to investigate the acoustic changes as a function of focus in the pre-focus and post-focus domains as well as in the on-focus domain in Turkish and listeners' sensitivity to those changes in retrieving information from

the acoustic signal. For this purpose, a production and a perception experiments have been conducted. For the production study, speakers read sentences in which the location of focus was manipulated via wh-questions preceding those sentences. For the perception study, listeners heard selected sentences recorded for the production study and were asked to judge the prominent word in the sentence. Results showed that focused words had increased duration and intensity, and post-focus words had reduced F0, duration and intensity. Interestingly, pre-focus words were found to have increased F0 and duration. Listeners' identification of focus was positively affected by the presence of acoustic changes especially in the post-focus domain. These findings have implications for speech perception and modelling prosody.

4pSCb30. Modeling imperatives in Spanish. Sergio Robles-Puente (Dept. of Linguist., Univ. of Southern California, 3601 Watt Way GFS 301, Los Angeles, CA 90089-1693, roblespu@usc.edu)

The intonation of imperatives in Spanish has traditionally been considered to not differ systematically from that of declaratives. This study shows that given the appropriate contexts, imperatives can exhibit unique phonetic properties. Nine speakers of Peninsular Spanish produced imperatives in response to instructions that elicited different levels of imperativity, along with control declarative items. Results show that while imperatives may fail to differ from declaratives in some conditions, when the context requires a stronger imperative, speakers use intonational configurations not found in declaratives. These include higher F0 values and changes in the overall pitch contour with higher F0 values toward the end of the sentence, different boundary tones and different F0 peak alignments. A perceptual experiment with 13 speakers confirmed the relevance of these intonational modulations by demonstrating that the strategies that were more commonly used in the production experiment were preferred over others to express imperativity. Results can be modeled within the framework of grammar dynamics (Gafos and Benus, 2006. Cognitive Science, **30**, 837–862). [This research was supported by the University of Southern California Del Amo Foundation.]

4pSCb31. Intonation of Mandarin speakers in their English as a Foreign language. Karen Barto-Sisamout (SLAT Program, 1423 E. Univ. Blvd., P.O. Box 210067, Tucson, AZ 85721, kabarto@email.arizona.edu)

Does the prosody of speakers' first language (L1) influence their prosody in their second language (L2)? The current work investigates this for tone languages (Beijing Mandarin and Taiwanese Mandarin) as L1, and an intonation/stress language, English, as L2. English uses F0 contours in the intonation system, to signal syllabic prominence in a word, word prominence in a phrase, and the difference between questions and statements. In English, there is a pitch peak delay, where the F0 peak occurs after the stressed syllable in two-syllable stress-initial words. Conversely, tone languages use F0 contours lexically, and in the case of Mandarin, the F0 peak is on the stressed syllable. Thus, F0 measurements were taken from three subject groups (Beijing Mandarin L1, Taiwanese Mandarin L1 and Native English) who produced narrow and broad focus statements and questions, to learn if the Mandarin speakers lack pitch peak delay in English like in their L1, or delay some of their pitch peaks beyond the corresponding syllable offset, like English speakers. Further, all groups produced contrastive focus statements and questions, to see to what degree differences in the L1 system impact L2 production, as Beijing and Taiwanese Mandarin differ in these structures. Data analysis is ongoing.

4pSCb32. Prosodic characterization of reading styles using audio book corpora. Michael I Proctor and Athanasios Katsamanis (Signal Anal. and Interpretation Lab., Univ. of Southern California, Los Angeles, CA 90089)

Although native English speakers have strong intuitions about the felicity of different reading styles, it is unclear which properties of read speech contribute to these reactions. Although the prosodic structures of read speech and spontaneous speech have been shown to differ (Howell and Kadi-Hanifi, 1991; Blaauw, 1994), it is not clear whether similar prosodic factors contribute to the perception of different reading styles as more felicitous nor even whether such differences can be systematically quantified. A large-scale corpus analysis of read speech was conducted to shed more light on the prosodic characteristics of those reading styles preferred and dispreferred by native speakers of American English. Audio book recordings of classic works of English literature by male and female American readers were rated by native speakers. The two most and least preferred renditions were transcribed at lexical and phonemic levels using SailAlign (Katsamanis, 2011). A variety of metrics were calculated to characterize prosodic properties of each of the readers, including %V, VarCoV, VarCoC, and nPVI (Grabe, & Low 2002; Stojanovic, 2009). The results suggest that although listeners exhibit a preference for syllabic regularity, the perceived felicity of reading styles results from a combination of factors. [Work supported by NIH].

4pSCb33. Acoustic correlates of spanish speech rhythms. Michael J. Harris (Dept. of Spanish and Portuguese, UCSB, Phelps Hall 4206, Santa Barbara, CA 93106-4150, michaelharris@umail.ucsb.edu) and Stefan Th. Gries (UCSB, CA 93106-3100, stgries@linguistics.ucsb.edu)

This paper describes a study of the acoustic correlates of speech rhythms of Hispanic Bilinguals living in California and Mexican Monolinguals living in Mexico City in order to study the effect of bilingualism on language, especially on rhythm classes, and the reliability of acoustic correlates in distinguishing these classes. This study addresses stress-timing versus syllable-timing as described in Pike's pioneering work (1945). Ten monolingual speakers from Mexico and ten bilingual Spanish–English speakers born and raised in California to Mexican parents were recorded speaking spontaneously. Fifty vowel durations per speaker were collected from phrases in these recordings and explored statistically and graphically with R.2.12.2 (R Development Core Team, 2011) in order to determine the reliability of various acoustical correlates of language rhythms in differentiating speech rhythms between varieties of Spanish. Specifically the Pairwise Variability Index, introduced by Low and Grabe (1995), and interval measures, such as the standard deviation and normalized standard deviation of vowel durations, were explored. The effects of word frequencies, as determined by relevant files from the Corpus del Espa'ol (Davies, 2002), were also considered in the analysis of the data. [Work supported by a grant for the University of California Institute for Mexico and the United States (UC Mexus).]

4pSCb34. Effect of linguistic background on convergence of prosodic rhythm. Gayatri Rao (Dept. of Psych., University of Texas, 1 University Station A8000, Austin, TX 78712, raog@mail.utexas.edu), Rajka Smiljanic (Univ. of Texas, 1 University Station B5100, Austin, TX 78712), and Randy Diehl (Univ. of Texas, 1 University Station A8000, Austin, TX 78712)

Speech patterns of the interlocutors become more similar to each other over the course of an interaction. These spontaneous speech adaptations, or phonetic convergence (PC), have been demonstrated for segmental features, such as vowels and voice onset times (VOT) and for suprasegmental features, such as stress. In this study, speaker adaptations to speech rhythm are examined before and after an interactive map task. Using American English and Indian English speakers, convergence was measured using the centroid of the envelope modulation spectrum (EMS + centroid, Rao & Smiljanic, 2011). This spectral measure of rhythm goes beyond considering consonantal and vocalic duration variability, as used in the traditional rhythm measures, and includes information about syllable prominence, stressed and unstressed syllable variation and distribution, and pauses and disfluencies. This research will allow us to examine whether language background has an effect on convergence of global speech properties, such as linguistic rhythm. The results of this study add to our current knowledge of features that are subject to imitation in the speech of dialogue partners.

4pSCb35. A biologically inspired neural network for modeling phrase-final lengthening. Erin C. Rusaw (Dept. of Linguist., Univ. of Illinois, 4080 Foreign Lang. Bldg., 707 S Mathews Ave. Urbana, IL 61801, erusaw2@illinois.edu)

This work proposes a central-pattern-generator-inspired neural network model for the interaction between phrase-final lengthening and stress. Recent work in the area of speech prosody has been concerned with the mechanisms involved in phrase-final lengthening, and specifically how phrase-final lengthening interacts with stress or prosodic prominence. The current study investigates the interaction of stress and lengthening at the end of English phrases. Adult American English speakers were recorded reading aloud sentences in which phrase boundaries had been manipulated so that the target words were either phrase-final or phrase-medial, and the durations of syllables in the target words were compared between the two conditions.

Results so far support previous findings that phrase-final lengthening in English affects stressed syllables near phrase boundaries and phrase-final syllables while leaving unstressed syllables between the two unaffected [Turk and Shattuck-Hufnagel, 2007]. Domain-based models of prosodic lengthening have so far been unable to provide a unified account of this phenomenon. A biologically plausible artificial neural network is shown which provides a model of the mechanism behind this interaction using three oscillators with differing periods which input to three interconnected thresholded integrate-and-fire artificial neurons, the output of which determines the timing of the syllables, stress feet, and phrase.

THURSDAY AFTERNOON, 3 NOVEMBER 2011 PACIFIC SALON 3, 1:00 TO 5:00 P.M.

Session 4pUWa

Underwater Acoustics: Volume Scattering From Objects, Bubbles, or Internal Waves

R. Lee Culver, Chair

Applied Research Lab., Pennsylvania State Univ., P.O. Box 30, State College, PA 16804-0030

Contributed Papers

1:00

4pUWa1. Boundary enhanced scattering by solid metallic simple geometric shapes: Experiments and modeling. Jon R. La Follett (Naval Surface Warfare Ctr., Panama City Div., Panama City, FL 32407, jon.lafollett@navy.mil) and Philip L. Marston (Washington State Univ., Pullman, WA 99164-2814)

The presence of a flat reflecting boundary can enable target scattering mechanisms that are not possible for objects in the free field. Experimental results demonstrating a strong boundary related backscattering feature for a solid aluminum cylinder near an air–water interface will be presented. This effect has been modeled previously [J. R. La Follett, Ph.D. thesis, WSU (2010)] by treating the volume of water bounded by the top surface of a solid cylinder and the air–water interface as a waveguide; qualitative agreement between the model and experimental results was demonstrated. In the present work that model is extended to incorporate aspects of the scatterer geometry. Monostatic and bistatic experimental measurements were obtained by suspending a solid aluminum cylinder, solid steel sphere, and a rectangular aluminum bar through the air–water interface of a tank. Model predictions for a solid cylinder are in good agreement with the observed dependence of the feature on the distance from the cylinder to the air–water interface. [Research supported by the NSWC PCD In-house Laboratory Independent Research program.]

1:15

4pUWa2. Bistatic scattering by scaled solid metallic objects: Circular line-scan measurements and modeling. Jon R. La Follett, Patrick C. Malvoso, and Raymond Lim (Naval Surface Warfare Ctr., Panama City Div., Panama City, FL 32407, jon.lafollett@navy.mil)

Multistatic sonar systems can be used to obtain target scattering information that cannot be measured monostatically. This information has potential benefits for detection and classification schemes. Bistatic and monostatic scattering measurements have been performed simultaneously on several scaled targets using the circular line-scan system of the small scale test bed facility at NSWC PCD. Water tank measurements were made for targets in the free-field and resting proud on or buried in simulated scaled sediment composed of spherical glass beads. Targets were placed in the center of the circular scan line. Circular synthetic aperture sonar and frequency-aspect target strength (acoustic color) results will be presented for a solid steel sphere, solid aluminum cone, and solid aluminum cylinders. Results for cylinders and the sphere are compared with T-matrix simulations to facilitate interpretation of features observed. Comparisons between back and forward scattering results demonstrate particularly strong features for each target in the forward scattering direction. Forward scattering by the sphere and cylinders also exhibits responses that arrive earlier in time than sound that travels directly from the source to the receiver. In the case of the sphere, this is attributed to elastic target responses involving leaky Rayleigh waves. [Research supported by ONR.]

1:30

4pUWa3. Scattering from highly extended acoustical objects using multiple precision computation. C. Feuillade (Facultad de Física, Pontifícia Universidad Católica de Chile, Av. Vicuña Mackenna 4860, Santiago, Chile)

The extended boundary condition (or "T-matrix") technique of Waterman [J. Acoust. Soc. Am., **45**, 1417-1429 (1969)] has recently been used to study scattering from highly non-spherical axisymmetric air-filled acoustical objects in water. Evaluation of the T-matrix for extended objects is frequently problematical, typically requiring the inversion of very large and highly ill-conditioned matrices. A second issue is computation of the matrix elements themselves, requiring accurate numerical quadrature of the surface integrals around the scattering object. In this work, a computational scheme was implemented using MATLAB®, and a freely downloadable software package "mptoolbox," which enables numerical computations to be performed to arbitrary degrees of precision. Here, it was implemented with 250 bits of precision specified for the mantissa part of the number, leading to computations with about 75 decimal places of accuracy. Using a multiple precision method significantly improved the stability and accuracy of both types of operations and indicates a powerful tool for the successful implementation of many scattering and other acoustical problems demanding high precision computational schemes. [Work supported by ONR.]

1:45

4pUWa4. On Rayleigh and Mie scattering. Jerald W. Caruthers (Dept. of Marine Sci., Univ. of Southern Mississippi, 1020 Balch Blvd., Stennis Space Ctr., MS 39560, jerald.caruthers@usm.edu)

Scattering described today as "Rayleigh scattering" represents something that is far short of what Rayleigh actually contributed to the topic in both optics and acoustics. This limited view seems to lie in a few papers in which he truncates series solutions for practical computations, thus leading to scattering of the form $(ka)^4$ for $ka \ll 1$, where k is the wavenumber and a is the radius of the sphere and for selected limitations on index of refraction. These approximations led optical scientists to equating "Rayleigh scattering" to little more than "the blue sky." In 1908, Gustav Mie developed a theory for plane-wave scattering from a sphere to which the names "Mie theory" and "Mie scattering" have been indelibly attached to many applications in optics. It is virtually unknown, especially in optics, that Rayleigh actually developed the full theory of plane-wave scattering from a sphere in 1878 (primarily Section 334, Vol. **2**, *The Theory of Sound*, Macmillan),

including original contributions in the concurrently developing mathematics of Bessel functions. The motivation of this presentation is to establish a means of treating weak scattering from bubbles based on their contribution as a distribution of spheres by combining Rayleigh and Mie.

2:00

4pUWa5. Sound speed, pulse spreading, and reverberation in muddy bubbly sediments. William M. Carey and Allan D. Pierce (Dept. Mech. Eng., Boston Univ., Boston, MA 02215)

The sound speed characteristic of the high-porosity mud has been found to have sonic speeds lower than expected. Since the presence of bubbles is known to be an important factor in decreasing the sound speed, these low sound speeds are attributed to methane microbubbles that result from biological decay. A theoretical treatment of "muddy sediments," the Card House Theory (Pierce and Carey, POMA (5), 7001, 2009), estimated the slow sound speed and frequency dispersion proportional to mud porosity, $C_{mud} \sim (0.91 - 0.97)C_w$. The presence of microbubbles can lower the sound speed consistent with the Mallock-Wood equation [Carey and Pierce, A 1aUW6, J. Acoust. Soc. Am. 129(4,Pt2) April 2011]. The recent Dodge Pond experiment found low sound speed estimates consistent with bubble volume fractions between 10^{-4} to 10^{-5}. The experiment has also produced estimates of pulse time spreading and reverberation. This paper interprets these results in terms of a three-component mixture with the bubbles distributed in a random Poisson process. Since measurement of the bubble size distribution within the mud is difficult, limits on the distribution may be obtained by the frequency dependent nature of the sound speed, pulse spreading, and reverberation characteristics. [Work sponsored by ONR OA and NSWC PCD.]

2:15

4pUWa6. Acoustic scattering by bubbles in naturally occurring mud sediments. Allan D. Pierce and Willam M. Carey (Dept. Mech. Eng., Boston Univ., Boston, MA 02215, adp@bu.edu)

Naturally occurring sediment mud contains bubbles created by decaying vegetable matter. Work reported by Preston Wilson *et al*. (ca. 2007) has determined via x-ray tomography systems that mud bubbles are not spherical in shape, but resemble oblate spheroids and are "inhomogeneously distributed." These features are explained in terms of the card-house structure of mud with an adaptation of the fracture mechanics ideas of Boudreau *et al*. (ca. 2002). The scattering of sound at low frequencies by such non-spherical bubbles has both monopole and dipole components. The scattered wave associated with the monopole term is proportional to the bubble volume. The dipole term involves an effective entrained mass tensor, which is found by a solution of Laplace's equation. All bubbles, regardless of shape, have a smallest resonance frequency, and the scattered radiation near the resonance frequency is monopole in character. Example solutions for the resonance frequencies and the scattering near resonance are given for oblate spheroidal bubbles, and a suggested interpolation from low frequencies to resonance frequencies is given. A discussion is also given of how one can make use of the range-evolving form of compact-source generated pulses to infer information about the bubbles near the propagation path.

2:30

4pUWa7. Affects of nearby bubbles on underwater array gain. R. Lee Culver and J. Daniel Park (Appl. Res. Lab, Penn State Univ., P.O. Box 30, State College, PA 16804)

Combining multiple sensor signals coherently (i.e., beamforming) improves spatial or angular resolution and increases signal to noise ratio (SNR). When the array is steered, signals arriving from the steering direction add in phase, while signals arriving from other directions do not (proper choice of signal frequency assumed). Array gain (AG) is a measure of how much the SNR at the array output is increased relative to array input SNR. The degradation in underwater acoustic array AG by scattering from nearby bubbles was measured at the AB Wood tank located at the Institute of Sound and Vibration Research (ISVR), University of Southampton, in June 2008. AG degradation is separate from the effects of bubbles in water to attenuate acoustic signals. Measured statistics of signal phase at the individual sensors show that as bubble density increases, phase differences between the elements increase and AG is degraded. We present a theory and numerical simulation that attributes the phase shifts to scattering from nearby bubbles and provides a way to predict AG degradation from the bubble density. Work sponsored by ONR Undersea Signal Processing.

2:45

4pUWa8. Transport equations for cross-frequency transport equations for acoustic intensity moments. Dennis B. Creamer (P.O. Box 660537, Arcadia, CA 91066)

The second and fourth moment mode-amplitude statistics for ocean sound propagation through random sound-speed perturbations are investigated using exact transport theory for the cross-frequency cross-mode coherence matrix. These exact equations are derived using the method of successive approximations, originally developed by Klyatskin and Tartarskii. These equations allow the determination of the validity of the usual transport equation (involving the Markov approximation), which is the first order approximation (in a infinite sequence of approximations to the exact equations). The range scales for the approach to the asymptotic behavior of the intensity moments, and the decay of the cross-modal coherence is easily determined at all frequencies.

3:00

4pUWa9. Transport equation approach for second order mode statistics in an ocean with random sound speed perturbations: Coherence. John Colosi and Tarun Chandrayadula (Dept. of Oceanogr., Naval Postgrad. School, Monterey, CA 93943)

A transport equation has been derived to describe the range evolution of the single frequency cross mode coherence matrix so that acoustic field coherence functions with temporal as well as depth and transverse separations can be easily computed. The theory assumes 2-D propagation in the depth range plane, small angle weak multiple forward scattering, and the Markov approximation, and it has been previously shown to accurately predict the observable of mean intensity for both deep and shallow water environments. This talk will address the issues of: the accuracy of the approximations, relative contributions from coupling and adiabatic effects, scaling with range and frequency, and the functional form of the coherence with regards to lag.

3:15–3:30 Break

3:30

4pUWa10. Investigating sources of variability of the range and structure of the low frequency shallow convergence zone. Stephen D. Lynch, Gerald L. D'Spain (Marine Physical Lab.—SIO, 291 Rosecrans, San Diego, CA 92106), Kevin D. Heaney (OASIS, Lexington, MA 02421), Arthur B. Baggeroer (MIT, Cambridge, MA 02139), Peter Worcester (SIO, La Jolla, CA, 92093), James Mercer (APL-UW, Seattle, WA, 98105), and James Murray (OASIS, Lexington, MA 02421)

During an experiment in the northern Philippine Sea in 2009, a ship towing Penn State's Five-Octave Research Array (FORA) at approximately 120 m depth drove counter-clockwise in an arc, maintaining constant range at one convergence zone (CZ) from a second ship holding station with an acoustic source deployed at 15 and 60 m. In addition, the FORA was towed at various depths in a star pattern about the station-keeping source ship, thereby sampling the first CZ in range, depth, and azimuth. Throughout the experiment, sound speed profiles were measured using expendable bathythermographs, expendable sound velocimeters, and conductivity/temperature versus depth sensors, and detailed bathymetric data were collected using the multibeam systems aboard these and other ships. By incorporating this extensive environmental information into numerical models, variability observed in these measurements of the range and structure and asymmetry of the distribution of received levels of the first CZ resulting from a shallow source to shallow receiver are attributed to variability in the sound speed of the upper-most water column and bathymetry (especially bathymetric features mid-way between the source and receiver). It is found that the sensitivity of the CZ to bathymetry is dependent on the sounds speed in the upper water column.

3:45

4pUWa11. Spectral effects of nonlinear internal waves on narrowband shallow-water signal propagation. Chad M. Smith and David L. Bradley (Appl. Res. Lab., Penn State Univ., P. O. Box 30, State College, PA 16804)

Water column, bathymetry, and acoustical sediment properties collected during the transverse acoustic variability eXperiment (TAVEX) of 2008 are used to create a 30 km computational environment for propagation models. These models are used to display internal-wave-induced acoustic variability which is characteristic of the shallow-water northern East China Sea environment where the experimental work took place. Analyses of computational models are then compared to narrowband acoustic recordings made during the experiment using a spectral and time-of-arrival analysis approach. The expected acoustic effects and computational impact of specific internal wave characteristics on acoustic arrivals will be discussed along with recorded data comparison. [Work supported by the Office of Naval Research.]

4:00

4pUWa12. Variability of horizontal interference structure of the sound field in the presence of moving nonlinear internal waves. Mohsen Badiey (Univ. of Delaware, College of Earth, Ocean, and Environment, Newark, DE 19716, badiey@udel.edu), Boris Katsnelson, and Andrey Malikhin (Voronezh State Univ., Voronezh 394006, Russia)

Variations of the low-frequency sound field when a train of nonlinear internal wave (NIW) crosses the acoustic track are considered. Three positions of moving NIW train, during 10:30 to 11:37 GMT on 19 August 2006 are examined. During this time, three periods of sound transmission took place, each lasting about 7.5 min. These periods referred to as starting time (10:30–10:37.5 GMT), mid-time (11:00–11:07.5 GMT), and the end-time (11:30–11:37.5 GMT). Low frequency modulated signals centered at 300 Hz, and bandwidth of 60 Hz with duration of about 2 s were transmitted. During this time, NIW consisting of 6-7 separate solitons was shifted in horizontal position by 2.5 to 2.8 km, moving toward the coast at the velocity of about 0.75 m/s. Fluctuations of the sound field in horizontal plane are described in detail for three periods of the sound pulses radiation: near the forward front, in the middle of the train, and near the back front using the fluctuations of an angle of horizontal refraction. Estimation of this angle using experimental data and those of the corresponding theory show the same value. [Work supported by ONR.]

4:15

4pUWa13. Observations of Philippine Sea sound-speed perturbations, and the contributions from internal waves and tides, and spicy thermohaline structure. John Colosi (Dept. of Oceanogr., Naval Postgrad. School, Monterey, CA 93943), Brian Dushaw (Univ. of Washington, Seattle Washington, 98105), Lora Van Uffelen (Univ. of Hawaii, Honolulu, HI, 96822), Matt Dzieciuch, Bruce Cornuelle, and Peter Worcester (Scripps Inst. of Oceanogr., La Jolla, CA 92093)

In the PhilSea09 pilot study two moorings equipped with temperature (T), conductivity (C), and pressure sensors, along with upper ocean ADCP, monitored ocean variability for a month in the Spring. The measurements reveal an energetic and nonlinear mixed diurnal-semidiurnal internal tide, a diffuse Garrett-Munk (GM) type internal wave field at or above the reference GM energy level, and a strong eddy field. One mooring, which was equipped with pumped sensors for enhanced salinity (S) resolution, was able to accurately quantify T and S variability along isopycnals (spice). The spice contribution to sound-speed fluctuation is observed to be strong near the mixed layer but significantly weaker than the other contributions in the main thermocline. Frequency spectra as well as vertical covariance functions will be presented to quantify the temporal and vertical spatial scales of the observed fluctuations.

4:30

4pUWa14. Refraction of horizontal rays and vertical modes from receding solitary internal wave front in shallow water. Mohsen Badiey (Univ. of Delaware, College of Earth, Ocean, and Environment, Newark, DE 19716, badiey@udel.edu), Valery Grogorev, and Boris Katsnelson (Voronezh State Univ., Voronezh 394006, Russia)

Previously, it was shown that on the approach of an internal wave to an acoustic source-receiver track in shallow water, interference between a direct and a horizontally refracted acoustic path can occur (J. Acoust. Soc. Am. **129**(4), EL141, 2011). This phenomenon that is dependent on the angular geometry between the solitary internal wave (IW) front and the acoustic track is similar to the well known Lloyd mirror interference phenomenon in optics. In particular, for a specific time during the SW06 experiment, it was shown that the refracted pulse propagating along horizontal ray, corresponding to the fourth vertical mode arrives with temporal delay relative to the direct horizontal ray. In this paper, we show this phenomenon occurring on the back front of the solitary IW. On the receding front, the forth mode of the refracted signal occurs with an increasing delay in the modal arrival time as the IW leaves the acoustic track. This separation between the direct and the refracted path continues while it gets larger until the IW is far enough from the track for refraction not to occur. A theoretical description of this phenomenon in support of the experimental observation is also presented. [Work supported by ONR.]

4:45

4pUWa15. Variations in the active and reactive intensity components of the sound field due to nonlinear internal waves. Robert J. Barton (Naval Undersea Warfare Ctr., Div. Newport, 1176 Howell St., Newport, Rhode Island 02841), Georges A. Dossot (Dept. of Ocean Eng., Univ. of Rhode Island, Narragansett Bay Campus, Narragansett, RI 02882), and Kevin B. Smith (Dept. of Phys., Naval Postgrad. School, Monterey, CA 93943)

Shallow water acoustic energy propagation influenced by nonlinear internal waves is investigated by examining complex acoustic intensity vector fields. The acoustic field is modeled using the three-dimensional Cartesian version of the Monterey-Miami parabolic equation (MMPE) algorithm, which relies upon a split-step Fourier approach. The modeled internal wave is approximated using environmental mooring data from the Shallow Water ’06 (SW06) field experiment, interpolated into a representative three-dimensional sound speed profile, and incorporated into the PE model. The soliton wavecrests are oriented such that they are parallel to the direction of forward acoustic propagation and variations along their length (such as curvature) are neglected. Both pressure and particle velocity fields are computed in a self-consistent manner, allowing a full description of the three-dimensional acoustic intensity field which describes the flow of energy in the presence of the solitons. The complex intensity field is separated into its active and reactive (real and imaginary) and spatial components and presented in the form of energy plots. Specific modeled examples showing horizontal refraction, focusing, and defocusing effects on the structure of the acoustic intensity field are illustrated.

Session 4pUWb

Underwater Acoustics: Measurement, Characterization, and Mitigation of Underwater Anthropogenic Noise

John A. Hildebrand, Cochair
Scripps Inst. of Oceanography, Univ. of California, San Diego, 9500 Gilman Dr., La Jolla, CA 92093-0238

Gerald D'Spain, Cochair
Scripps Inst. of Oceanography, Univ. of California, San Diego, 9500 Gilman Dr., La Jolla, CA 92093-0238

Contributed Papers

1:00

4pUWb1. Statistical analysis of northern Philippine Sea underwater sounds. Brianne Moskovitz, Gerald D'Spain (Marine Physical Lab., Scripps Inst. of Oceanogr., 291 Rosecrans St., Bldg. 4, San Diego, CA 92106, bmoskovi@ucsd.edu), Peter Worcester, Matt Dzieciuch (Scripps Inst. of Oceanogr., San Diego, CA, 92037), Kevin D. Heaney (OASIS Inc., Fairfax Station, VA 22039), Jim Mercer (Univ. of Washington, Seattle, WA 98105), and Art Baggeroer (MIT, Cambridge, MA 02139)

The deep ocean experiment, PhilSea09, was conducted April–May, 2009, in the central part of the northern Philippine Sea. The deep 1000-m section of the distributed vertical line array (DVLA), which was composed of 30 elements (upper 10 at 90 m spacing starting at 4285 m depth, and 5 m spacing for the deepest 20 elements over the depths 5185–5280 m), recorded primarily four types of sounds not created as part of the experiment: wind noise, ship noise, airgun signals, and earthquake T-phases. The statistical properties of these sounds are examined quantitatively using non parametric statistical tests operating on the single element and beam level narrowband envelope time series. These statistical tests include the Wald-Wolfowitz runs test for mutual independence of the data samples, the Kolmogorov-Smirnov two-sample test for stationarity, and the Lillifors test for Gaussianity. One result is that these sounds, except for wind-dominated noise, fail the test for Gaussianity. Higher-order spectral analysis is performed to quantify the degree of non-Gaussianity. In addition, analytical probability density functions are fit to the histograms of the envelope values. Physics-based models are developed to predict the statistical characteristics of some of these sounds. [Work supported by the Office of Naval Research.]

1:15

4pUWb2. Ambient noise bathymetric domains. Donald Ross (Wesley Palms Rm 325, 2404 Loring St., Box 101 San Diego, CA 92109, donaldnmiross@mac.com), Megan F. McKenna, Sean M. Wiggins, and John A. Hildbrand (Univ. of California San Diego, La Jolla, CA 92093)

For the purposes of describing and understanding ambient sea noise for frequencies below about 300 Hz, most of the world's seas can be classified as belonging to one of three bathymetric domains. These domains are distinguished by their proximity to shipping lanes and by the degree to which they are exposed to noises originating at long distances. The three domains display different short-term characteristics as well as different historical patterns. In this paper, the three domains are described and typical ambient noise characteristics for each are shown, including changes which are attributable to increased ocean commerce.

1:30

4pUWb3. Unintended consequences of recent changes in ship traffic. Megan F. McKenna, Sean M. Wiggins, John A. Hildebrand (Scripps Inst. of Oceanogr., Univ. of California San Diego, Ritter Hall 200E, 8635 Kennel Way, La Jolla, CA 92093-0205, megan.mckenna@gmail.com), and Donald Ross (San Diego, CA 92109, USA)

Underwater ambient noise levels measured off the coast of southern California were correlated with regional changes in commercial shipping trade. Between 2007 and 2010, two events occurred that resulted in a decrease in ship traffic in the Santa Barbara Channel: the economic recession and a coastal air-quality improvement rule. From October 2005 to June 2010, monthly low-frequency ambient noise levels at a site 3 km from a major shipping route were compared to regional traffic levels. Two different metrics of ship traffic showed that on average a 1 dB reduction in low-frequency noise levels resulted from a decrease in traffic by one ship passage per day in a coastal basin.

1:45

4pUWb4. Passive acoustic monitoring of vessel presence at Rose Atoll and the National Park of American Samoa. Pollyanna Fisher-Pool (Joint Inst. for Marine and Atmospheric Res., Univ. of Hawaii, Honolulu, Hawaii, pollyanna.fisher-pool@noaa.gov), Marc O. Lammers, Lisa M. Munger (Univ. of Hawaii, Kaneohe, Hawaii), Kevin Wong, and Russell E. Brainard (Pacific Islands Fisheries Sci. Ctr., Honolulu, Hawaii)

American Samoa is in the process of evaluating the development of a network of marine protected areas (MPAs) to preserve coral reef environments and to prevent the decline of fish populations. Two long-standing MPAs in American Samoa are Rose Atoll Marine National Monument (RAMNM) and coastal marine regions within the National Park of American Samoa (NPSA) system. NPSA includes areas on the populated island of Tutuila, while RAMNM is approximately 130 miles away from the nearest population. Both are protected marine reserves where commercial and public recreational fishing are restricted, although the size and remoteness of the locations create a challenging task for observation and enforcement. A growing management concern over the decline of large fishes and possible illegal fishing has prompted interest in vessel incidence within the two MPAs. We gathered evidence of vessel presence with the use of long-term, autonomous passive acoustic monitoring within the two MPAs. Here we present results of vessel detection within acoustic recordings collected 2009–2010 at RAMNM and 2006–2007 and 2008–2009 at the NPSA Tutuila location. Results from this study highlight the patterns and seasonality of vessel incidence and provide managers with information to assist enforcement.

2:00

4pUWb5. A high-fidelity model for mitigating underwater pile driving noise in a shallow ocean waveguide. Kevin L. Cockrell, Ann Stokes, and Dwight Davis (Appl. Physical Sci. Corp., 2445 Truxtun Rd. #200, San Diego, CA 92106)

Driving large piles into the seafloor, as is done when constructing offshore wind farms, produces high level underwater noise that can have an adverse effect on local marine life. This talk reviews an investigation of methods to mitigate this pile driving noise. A numerical model was used to simulate the structural vibrations in the pile and its coupling to the acoustic field in surrounding air, water and sediment. The simulated acoustic field in the immediate vicinity of the pile was then coupled into an ocean waveguide propagation model using a virtual-source technique to match the boundary conditions. These numerical models were used to assess the relative contribution of the air-borne, water-borne, and sediment-borne acoustic radiation to the noise level in the water-column at ranges up to several hundred meters from the pile. Various noise mitigation methods were simulated and compared. It was determined that a dewatered cofferdam, which places a layer of air between the vibrating pile and the seawater, has the potential to reduce the far-field noise level by approximately 20 dB. A practical method for creating a dewatered cofferdam during construction is the subject of ongoing work.

2:15

4pUWb6. A model for noise radiated by submerged piles and towers in littoral environments. Todd A. Hay, Yurii A. Ilinskii, Evgenia A. Zabolotskaya, Preston S. Wilson, and Mark F. Hamilton (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX 78713-8029)

Pile driving in shallow water during the construction of bridges and other structures can produce transient broadband noise of sufficient intensity to kill fish and disturb marine mammals. Sustained tonal noise radiated by towers supporting offshore wind turbines contains energy in frequency bands that may inhibit detection of coastal activities via passive sonar and seismic sensors. Understanding the generation and propagation of underwater noise due to pile driving and wind farms is important for determining the best strategies for mitigating the environmental impact of these noise sources. An analytic model, based on a Green's function approach, is presented for the sound radiated in the water column by a submerged cylindrical structure embedded in horizontally stratified layers of sediment. The sediment layers are modeled as viscoelastic media and the Green's function is derived via angular spectrum decomposition. Noise radiation due to both vibration of the structure and impulses delivered to the sediment is considered. Contributions to the pressure field in the water column due to radiation directly into the water, radiation from the sediment into the water, and Scholte waves propagating along the sediment-water interface will be discussed. [Work supported by the ARL:UT IR&D program.]

2:30

4pUWb7. Numerical analysis of sound radiation underwater from a fully submerged pile. Shima Shahab, Katherine F. Woolfe, and Mardi C. Hastings (George W. Woodruff School of Mech. Eng., Georgia Inst. of Technol., Atlanta, GA 30332-0405)

The study of acoustic radiation from pile driving is essential because the impact between the hammer and pile causes extremely high underwater sound pressure levels that are potentially harmful to the marine environment. When the hammer strikes a cylindrical steel pile, movement of the pile wall in circumferential, longitudinal, and radial directions can excite many modes of vibration. The radial expansions of the pile that propagate along the pile after impact create sound waves in the surrounding water. A finite-difference time-domain (FDTD) formulation is presented to analyze the sound produced from a fully submerged pile with simply supported ends. The pile considered is a cylindrical shell of finite length, bounded by sediment at the bottom end, and surrounded by and filled with water. Three coupled partial differential equations govern the vibration of the shell, and the effects of acoustic media including water and sediment are added in the radial direction. Results of a correlation between the radiated sound field predicted by the FDTD model and acoustic data from a scaled physical model of a fully submerged pile hit with an impact hammer are presented. [Work supported by the Oregon Department of Transportation and Georgia Institute of Technology.]

2:45

4pUWb8. A scaled physical model to study underwater noise from impact pile driving. Katherine F. Woolfe, Shima Shahab, Juan Morales, and Mardi C. Hastings (George W. Woodruff School of Mech. Eng., Georgia Inst. of Technol., Atlanta, GA 30332-0405, mardi.hastings@gatech.edu)

Development of computational models to predict underwater noise from impact pile driving is limited because of the difficulty and cost involved in collecting acoustic field data for model verification during construction activities. To alleviate this situation, a scaled physical model for marine pile driving was designed and implemented in a 500-gallon shallow water tank, 3.5 m long and 0.85 m wide. The scaled piles are steel pipes having lengths up to 1 m, and radius-to-wall thickness and length-to-radius ratios similar to large cast-in-shell steel (CISS) piles. The wavelength-to-depth ratio for the primary breathing mode of the fully submerged scaled piles and a fully submerged CISS pile of diameter 2.4 m and length 30 m is between 2.0 and 2.5. The impact force is generated and measured with an impulse hammer, and sound field data are collected using a small 2-D hydrophone array. Data are correlated with the results of numerical and analytical models developed to predict sound radiation from CISS piles to verify that the scaled physical model accurately represents their structural acoustics behavior. [Work supported by the Georgia Institute of Technology and the Oregon Department of Transportation.]

3:00

4pUWb9. Underwater structure-borne noise analysis using a finite element/boundary element coupled approach. Dooho Lee (Dept. of Mech. Eng., Dongeui Univ., Busan, 614-714, South Korea, dooho@deu.ac.kr), Hyun-Sil Kim, Bong-Ki Kim, and Seong-Hyun Lee (Acoust. Team, KIMM, Daejeon, South Korea)

Radiated noise analysis from ship structure is a challenging topic due to difficulties in accurate calculation of fluid-structure interaction as well as massive degree of freedoms of the problem even in the case of finite element/boundary element coupled formulations. To reduce the severity of the problem, a new added mass and damping approach is proposed in this paper. The complex frequency-dependent added mass and damping matrices are calculated by using the high-order Burton-Miller boundary element formulation in order to obtain accurate values over all frequency bands. The calculated fluid-structure interaction effects are added to the structural matrix calculated by commercial finite element software, MSC/NASTRAN. An iterative solver is introduced to solve the eigenvalue problem because the combined system matrices are complex and frequency-dependent. The calculated eigenfrequencies for a submerged cylindrical shell show good agreement on reference data. The accuracy and efficiency of the present formulation is compared with those from conventional finite element/boundary element formulation. The comparison results show that the present formulation has better accuracy than the conventional one because of accurate added mass and damping calculation.

3:15

4pUWb10. Non-cavitating propeller noise source inversion. Woojae Seong, Jaehyuk Lee, Ara Hyun (Dept. of Ocean Eng., Seoul Natl. Univ., Seoul 151-742, Korea, wseong@snu.ac.kr), and Junghoon Lee (Samsung Heavy Industry, Taejon, Korea)

Ship's propeller cavitation generates major inboard noise and vibration over the aft body surface of a ship. During the last decade, cavitation-induced hull pressure forces have been reduced considerably leading to reduced noise and vibration levels owing to cavitation, in which case the non-cavitating components of propeller excitations have to be considered: pressure fluctuations due to blade loading and blade thickness. In this work, an algorithm to invert for the non-cavitating propeller noise source parameters based on the experimental data in a cavitation tunnel is shown. The fluctuating hull pressure is estimated by forward modeling based on acoustic boundary element method (BEM). The propeller blade loading and thickness noise sources are modeled using rings of dipoles and quadrupoles, respectively. The inversion for these pseudo sources are carried out by matching the data obtained from a cavitation tunnel experiment, where several hydrophone are flush mounted on the ship's surface near the propeller. Proper inversion results are obtained when six or more dipole and quadrupole rings are placed at each blade. Using the inverted source parameters, hull pressures were estimated, which show good agreement with the experiment data.

THURSDAY EVENING, 3 NOVEMBER 2011 — 7:30 TO 9:30 P.M.

OPEN MEETINGS OF TECHNICAL COMMITTEES

The Technical Committees of the Acoustical Society of America will hold open meetings on Tuesday, Wednesday, and Thursday evenings beginning at 7:30 p.m.

These are working, collegial meetings. Much of the work of the Society is accomplished by actions that originate and are taken in these meetings including proposals for special sessions, workshops, and technical initiatives. All meeting participants are cordially invited to attend these meetings and to participate actively in the discussion.

Committees meeting on Thursday are as follows:

Animal Bioacoustics	Pacific Salon 1
Noise	Royal Palm 1/2
Speech Communication	Royal Palm 5/6
Underwater Acoustics	Pacific Salon 3