Session 2pAA

Architectural Acoustics, Noise, and ASA Committee on Standards: Innovations and Challenges in Healthcare Acoustics

Erica Ryherd, Cochair
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David M. Sykes, Cochair
The Remington Group LP, 23 Buckingham St., Cambridge, MA 02138

Kenric D. Van Wyk, Cochair
Acoustics By Design, Inc., 124 Fulton St. East, Second Fl., Grand Rapids, MI 49503

Chair's Introduction—1:00

Invited Papers

1:05

2pAA1. Research methods: Beta-testing Facilities Guideline Institute's acoustical criteria—A case study of continual improvement in healthcare facilities. Kurt Rockstroh (Board of Directors, Facility Guidelines Inst., Falcon Pier, Boston, MA 55801, kurtr@steffian.com)

Continual improvement processes (CIP) present opportunities and challenges for researchers. In healthcare, FGI manages a four-year building-code-development cycle that is continually improved by research and sets standards that others follow (USGBC-LEED, ICC). Innovative, accelerated research methods are needed to continually improve healthcare environments during a period of rapid and profound change, so FGI's CIP method seeks to synchronize the work of code writers and independent researchers. One useful research method is to subject draft requirements to "beta-testing." FGI leaders (led by the presenter) did this to test new acoustical criteria on a 641,000 "hospital of the future" wing at Tufts University's Baystate Medical Center in Springfield MA (a major hospital and Level 1 trauma center). Planning began in 2006 (using draft #1 of the acoustical guidelines); occupancy began in late 2012 (after the guidelines' publication, Jan 2010). But only now, after a full year of occupancy, can the full cost and benefit of the improvements be measured using Federal HCAHPS (a new procedure required by the Affordable Care Act that impacts hospital reimbursements). "Beta test" results were factored into the forthcoming FGI 2014 edition. The "beta test" is the subject of independent, funded research currently underway.

1:25

2pAA2. New healthcare safety risk assessment toolkit includes acoustics and noise control design measures for error prevention in medication safety zones. Mandy Kachur (Soundscape Engineering LLC, 317 S Div. St. #170, Ann Arbor, MI 48104, mkachur@soundscapeengineering.com), Xiaobo Quan (The Ctr. for Health Design, Concord, CA), Daniel M. Horan (Cavanaugh Tocci Assoc., Inc., Sudbury, MA), and David M. Sykes (Acoust. Res. Council, Lincoln, MA)

The Facilities Guidelines Institute (FGI) manages the *Guidelines for Design and Construction of Hospitals and Outpatient Facilities*, which is used by the Joint Commission, many federal agencies, and authorities having jurisdiction in 42 states. The forthcoming 2014 edition calls for completion of a Safety Risk Assessment (SRA) to guide healthcare facility planning and design teams through systematic evaluation of safety issues in the built environment. In 2012, FGI and the federal Agency for Healthcare Research and Quality commissioned The Center for Health Design to oversee a three year effort to develop, test, and disseminate an SRA toolkit. The toolkit will consist of an online facility design questionnaire, supporting whitepapers and guidelines, and education for the healthcare community. The Acoustics Research Council was invited to contribute by integrating noise control language into the medication safety segment of the online questionnaire tool. The result is a question that specifically addresses acoustics and noise control design for error prevention in medication safety zones. In May 2013, the questionnaire content was finalized, and it will be validated, integrated, and evaluated during the subsequent two years. The co-authors are project participants.

1:45

2pAA3. New healthcare acoustics subcommittee: Overview and call for participation. Gary Madaras (Making Hospitals Quiet, Chicago Metallic, 4849 S. Austin Ave., Chicago, IL 60638, DoctorSonics@aol.com)

The Acoustical Society of America (ASA) Technical Committee on Architectural Acoustics (TCAA) formed a new Healthcare Acoustics subcommittee at the 165th meeting in Montreal. The 1st meeting of the Healthcare Acoustics subcommittee will be held at the 166th meeting in San Francisco. This presentation will provide an overview of the purpose of the subcommittee and introduce possible initiatives

on which the subcommittee may want to initially focus. *Examples*: 1. Despite financial incentive to make their hospitals quieter, most healthcare providers cannot significantly increase their "quiet-at-night" (HCAHPS) scores in their older or newer inpatient facilities. The national "quiet-at-night" (HCAHPS) score remains the lowest of all patient experience quality indicators. Changing things may require a significant paradigm shift in the architectural and acoustical communities. 2. The Facility Guidelines Institute (FGI) has developed its new *Guidelines for Design and Construction of Residential Health, Care, and Support Facilities* to be published in early 2014. Those Guidelines for eldercare facilities will include terms and concepts such as "nature sounds," "positive auditory distraction," "auditory landmark," "music," "quiet rooms," "quietly operating," etc., that have yet be fully defined acoustically. Please attend and participate.

2:05

2pAA4. Value based education for healthcare design. Edward Logsdon (D. L. Adams Associates, Inc., 1536 Ogden St., Denver, CO 80218, elogsdon@dlaa.com and Kenric Van Wyck (Acoustics By Design, Grand Rapids, MI 49503)

Accelerating the understanding and adoption of the FGI Guidelines is an essential part of the continuous improvement process—especially at a time when healthcare is coping with so much rapid change, and while the FGI Guidelines are becoming more accepted internationally. As part of the process to improve the acoustical environment in healthcare facilities, there is also the continual need to educate architects and designers concerning sound and vibration design criteria included in the FGI Guidelines. Designers need to understand how to apply the criteria, proposed changes to the guidelines, and how the changes will benefit the hospital by improving the patient experience. Using professional social networking, electronic, and face-to-face continuing education combined with research based design, professionals and the public can be educated on the merits of improvement in the acoustical design of healthcare facilities. Web-based education will attract a much wider audience worldwide and further international acceptance. Project case studies where FGI Guidelines were applied will identify the cost impact while increasing awareness and understanding.

2:20

2pAA5. Top ten research needs in the decade ahead. David M. Sykes and William J. Cavanaugh (Architectural Acoust., Rensselaer Polytechnic Inst., 31 Baker Farm Rd., Lincoln, MA 01773, david.sykes@remington-partners.com)

Independent, third-party research co-led by teams including medical personnel, engineers, and scientists is key to raising the bar on acoustical performance in healthcare facilities. While links between noise and health have been officially ignored in the United States for three decades, research over the past eight years in healthcare environments has significantly advanced understanding and created a small community of funding organizations and interested researchers. To make further progress, researchers must demonstrate clear links between noise/sound and patient outcomes as well as the performance effectiveness of healthcare professionals on critical factors such as error rates. Achieving useful, translational results requires: innovative research methods, collaborative funding mechanisms, and a clear focus on outcomes. With no Federal agency providing organized peer-review, it is essential for the acoustics research community to both agree on a research agenda and organize peer-review procedures that will enable continued progress among a widely distributed population of researchers working within the constraints of limited funding. The presenter worked with FGI to draft a "top 10" research agenda. He co-chairs the FGI Acoustics Working Group, edits the Springer Verlag "Quiet series," and has experimented with collaborative research funding mechanisms over the past three decades.

Contributed Paper

2:40

2pAA6. Acoustic quality of buildings: Contributions to the workers health of the health area. Marta R. Macedo (Coordenação de Saúde do Trabalhador, Fundação Oswaldo Cruz, Av. Brasil 4365, Pavilhão Carlos Augusto da Silva, Sala 202, Rua Comandante Rubens Silva 90, ap. 203, bl 1, Rio de Janeiro, Rio de Janeiro 21040-360, Brazil, mribeiro@fiocruz.br), Marcia S. Almeida, Liliane R. Teixeira (Escola Nacional de Saúde Pública, Fundação Oswaldo Cruz, Rio de Janeiro, Rio de Janeiro, Brazil), Ana Paula Gama, Stephanie Livia S. Silva, Olga Dick (Coordenação de Saúde do Trabalhador, Fundação Oswaldo Cruz, Rio de Janeiro, Brazil), Isabele C. Costa, Denise Torreao (Escola Nacional de Saúde Pública, Fundação Oswaldo Cruz, Rio de Janeiro, Brazil), Paulo Roberto L. Jorge (Coordenação de Saúde do Trabalhador, Fundação Oswaldo Cruz, Rio de Janeiro, Brazil), Paulo Marcelo S. Dias, Daniel Valente, and Diane R. Valente (Escola Nacional de Saúde Pública, Fundação Oswaldo Cruz, Rio de Janeiro, Brazil)

This work presents a case study carried through in a research center, education, assistance, and technological development in the areas of the health of the woman, child, and adolescent, situated in Rio de Janeiro, where pathologies of average and high complexity are taken care of. It intended detect environmental problems that could affect the health of workers of the institution, in order to support the elaboration of the action plan to mitigate urgent questions and to develop an architectonical project for a new center. The environments and processes of work had been evaluated, through a participative approach, being visited all the sectors of the center, carried through interviews half-structuralized with workers of all the sectors and promoted debates that had pointed the noise as one of the main factors of bother and stress. The observation in loco and analyze of the project allowed to detect that the architectural design and the disrespect of basic acoustics recommendations had contributed for this picture. Measurements carried through in some pointed areas as uncomfortable indicated to have exposition to sound pressure level above 75 dB(A). However, the resolution of the many of identified acoustic problems will have to wait the construction of the new headquarters.

2:55–3:10 Break

Invited Papers

3:10

2pAA7. Why alarm fatigue is a pivotal issue that affects the acoustical design of healthcare facilities. Paul Barach MD (Res. Committee, Facility Guidelines Inst., 31 Baker Farm Rd., Lincoln, MA 01773, pbarach@gmail.com)

The U.S. FDA and Joint Commission designated "alarm fatigue" the "#1 priority in healthcare technology" in 2011–2012, acknowledging that this acoustical problem results in hundreds of patient deaths and thousands of injuries. The healthcare facilities industry has been slow to recognize that "alarm fatigue" is partly a facility design issue: i.e., a cacophony of recurrent noises from myriad uncorrelated medical devices, set at maximum loudness, occurring in hard-walled, reverberant spaces (such as patient rooms, ORs, and ICUs) produce elevated stress, sleep impairment, disorientation, and dangerously irrational, potentially deadly behavior. "Alarm fatigue" has been addressed as a human factors problem elsewhere: e.g., nuclear plant control rooms (after Three-Mile Island) and aircraft cockpits. In healthcare, it is imperative to engage architects, designers, acoustical engineers facility engineering staffs, and clinicians, who represent the "first line of defense" as the medical device industry requires 5–10 years to implement solutions. The presenter co-led a delegation of 12 distinguished members of the acoustics profession to the national summit on "alarm fatigue," Washington DC, 2011 and has co-authored peer-reviewed medical journal articles and a forthcoming FGI white paper on the subject. This presentation focuses on solutions, challenges, and the research roadmap needed to address "alarm fatigue."

3:30

2pAA8. The healthcare acoustics research team: Bridging the gap between architecture, engineering, and medicine. Erica Ryherd (Woodruff School of Mech. Eng., Georgia Inst. of Technol., Mech. Eng., Atlanta, GA 30332-0405, erica.ryherd@me.gatech.edu), Jeremy Ackerman (Dept. of Emergency Medicine, Emory Univ., Atlanta, GA), Craig Zimring (College of Architecture, Georgia Inst. of Technol., Atlanta, GA), and Kerstin Persson Waye (Occupational and Environ. Medicine, Gothenburg Univ., Gothenburg, Sweden)

Healthy soundscapes are paramount to the missions of hospitals: patients need to sleep and heal without environmental stressors; staff, patients, and family need to communicate accurately but privately; staff need to be able to localize alarms and calls for help. This talk discusses recent findings from the Healthcare Acoustics Research Team (HART), an international, interdisciplinary collaboration of specialists in architecture, engineering, medicine, nursing, and psychology. Members of the HART network are actively engaged in research in the United States and Sweden, having worked in more than a dozen hospitals and a broad range of unit types including adult and neonatal intensive care, emergency, operating, outpatient, long-term care, mother-baby, and others. Highlights will include projects relating noise and room acoustic measures to staff and patient response in addition to studies evaluating impacts of acoustic retrofits. Results show that effective hospital soundscapes require a complex choreography of architectural layout, acoustic design, and administrative processes that is only beginning to be fully understood.

3:50

2pAA9. Acoustic comfort in healthcare facilities—What is it? and What does it mean to both patients and medical staff? Kenneth P. Roy (Bldg. Products Technol. Lab, Armstrong World Industries, 2500 Columbia Ave., Lancaster, PA 17603, kproy@armstrong.com)

Acoustic comfort is all about meeting the acoustic needs of the patient, family, professional staff in hospitals, clinics, pharmacies, etc. These needs include low distraction and annoyance for patients to fully enjoy a "healing environment", high levels of speech intelligibility to support communications and to help reduce medical errors, and adequate speech privacy to meet HIPAA and other privacy needs. How to design for these, and how to verify performance—these are the key issue to be addressed. Architectural choices need to be made with a view for acoustic comfort in addition to all the other relevant design factors. Facility evaluations as part of the commissioning process need to include both objective and subjective surveys to push forward evidence based design approaches for others to take advantage of in future designs. All of these issues are part of the ongoing discussions and field studies that we have been having, and will continue to have.

4:10

2pAA10. Analysis of granular hospital sound level measurements. Benjamin Davenny (Acentech Inc., Cambridge, MA), Aaron Betit (Acentech Inc., 601 South Figueroa St., Ste. 4050, Los Angeles, CA 90017, abetit@acentech.com), and William Yoder (Acentech Inc., Cambridge, MA)

Remote sound monitoring systems were deployed in several locations in a hospital to measure sound levels due to building system and activity noise. In addition to 5 min interval data, granular sound levels were recorded several times per second to provide flexibility in data analysis. Analysis of this information will be presented.

4:30

2pAA11. Estimation of noise induced hearing loss because of indoor and outdoor environment noise factors in Turkish healthcare facilities: A survey of hospitals in Turkey. Filiz Kocyigit (Architecture, Atilim Univ., İncek, Ankara, Turkey, filizbk@gmail.com)

This article aims to evaluate effect of indoor and outdoor environment noise factors in healthcare facilities which affect the work area due to noise induced hearing loss, and also to determine their relationship with the architectural design of the building. As a case study; noise levels, in five state and private healthcare centers, including medical schools, research hospitals, and state hospitals in Turkey were measured. They were compared with similar healthcare centers in the United States. Results include equivalent sound pressure levels (Leq for 5 min from 20 spots in each area), and Lmax–Lmin evaluated as a function of location, frequency, time, and days of the week. Research showed that no location was in compliance with current World Health Organization Guidelines, and a review of

objective data indicated that this was true of hospitals throughout the world. Noise induced hearing loss on continuously users had been estimated with these results in selected hospitals. Data gathered at various hospitals for last decay indicate a trend of increasing noise levels during daytime and nighttime hours. The implications of these results were significant for patients, visitors, and hospital staff.

4:50

2pAA12. Renovation of neonatal intensive care unit per the Facilities Guideline Institute's 2010 Acoustical Design Guidelines. Erik Miller-Klein (SSA Acoust., LLP, 222 Etruria St, Ste. 100, Seattle, WA 98109, erik@ssaacoustics.com)

We did an evaluation of the acoustical conditions of the existing neonatal intensive care unit at the St. Joseph's Hospital in Tacoma, Washington, and issued detailed renovation recommendations that were integrated into the final design. At the conclusion of construction, our team was able to evaluate the improvements of the acoustical conditions per the 2010 FGI Guidelines and the original performance. These included noise impacts from door closures, curtains, alarms, and the conversations.

5:10–6:10 Panel Discussion

TUESDAY AFTERNOON, 3 DECEMBER 2013 UNION SQUARE 23/24, 1:00 P.M. TO 3:20 P.M.

Session 2pABa

Animal Bioacoustics and Acoustical Oceanography: Broadening Applications of Tags to Study Animal Bioacoustics II

Marla M. Holt, Cochair
NOAA NMFS NWFSC, 2725 Montlake Blvd. East, Seattle, WA 98112

Alison K. Stimpert, Cochair
Dept. of Oceanogr., Naval Postgrad. School, Monterey, CA 93943

Invited Papers

1:00

2pABa1. Conservation applications of baseline acoustic tag data from right whales. Susan Parks (Dept. of Biology, Syracuse Univ., 114 Life Sci. Complex, Syracuse, NY 13244, sparks@syr.edu), Douglas P. Nowacek (Nicholas School of the Environ. and Pratt School of Eng., Duke Univ. Marine Lab., Beaufort, NC), Mark Johnson, and Peter L. Tyack (Sea Mammal Res. Unit, Scottish Oceans Inst., School of Biology, Univ. of St. Andrews, St. Andrews, Scotland, United Kingdom)

Data collected from archival acoustic tags have dramatically improved our understanding of marine mammal subsurface behavior. Some of the earliest tag deployments were made on the endangered North Atlantic right whale (*Eubalaena glacialis*) to better understand their behavior and to contribute to their conservation. Over the past 15 years, the database of acoustic tag records collected from a diverse cross section of the population has served as a valuable resource for research. Acoustic tag data have provided critical behavioral information including details on individual calling behavior in designated critical habitat feeding grounds, behavioral responses to controlled exposures of acoustic signals including conspecific and man-made signals, acoustic responses to vessel noise, and ground truth data for passive acoustic monitoring efforts. Current studies continue to utilize tag data to shed light on the individual, seasonal, and regional variation in acoustic behavior of right whales, particularly focused on the shallow water coastal habitats of the calving grounds off of Florida and Georgia and in the migration corridor off the Eastern United States. These studies have contributed data that are crucial for targeted conservation efforts and highlight the value of long-term databases of tag data in baleen whale research.

1:20

2pABa2. Controlled sound exposure experiments to measure marine mammal reactions to sound: Southern California behavioral response study. Brandon L. Southall (SEA, Inc., 9099 Soquel Dr., Ste. 8, Aptos, CA 95003, Brandon.Southall@sea-inc.net), John Calambokidis (Cascadia Res. Collective, Olympia, WA), Moretti David (Naval Undersea Warfare Ctr., Newport, RI), Jay Barlow (Southwest Fisheries Sci. Ctr., La Jolla, CA), Stacy DeRuiter (CREEM The Observatory, St. Andrews, Scotland, United Kingdom), Jeremy Goldbogen (Cascadia Res. Collective, Olympia, WA), Ari Friedlaender (Long Marine Lab., Univ. of California, Santa Cruz, Santa Cruz, CA), Elliott Hazen (NOAA-PFEL, Pacific Grove, CA), Alison Stimpert (Naval Postgrad. School, Monterey, CA), Arranz Patricia (Sea Mammal Res. Unit, Scottish Oceans Inst., St. Andrews, Scotland, United Kingdom), Erin Falcone, Greg Schorr, Annie Douglass (Cascadia Res. Collective, Olympia, WA), Chris Kyburg (SPAWAR Systems Ctr., San Diego, CA), and Peter Tyack (Sea Mammal Res. Unit, Scottish Oceans Inst., St. Andrews, Scotland, United Kingdom)

SOCAL-BRS is an inter-disciplinary collaboration designed to increase understanding of marine mammal behavior and provide a robust scientific basis for estimating risk and minimizing effects of mid-frequency military sonar systems. Data were collected using visual observations, passive acoustic monitoring, animal-attached acoustic and movement tags, photo ID, biopsy, and controlled sound exposure experiments on over 20 cetacean species in biologically important areas throughout the southern California Bight. Ninety-six individuals of ten species were tagged with six tag types, including two species [Baird's beaked whale (*Berardius bairdii*), Risso's dolphin (*Grampus griseus*)] not previously studied with such tools. Fifty-six controlled CEEs were conducted using protocols and protective measures to ensure animals were not harmed. Simulated sonar signals were projected through a deployed sound source and changes in vocal, diving, and horizontal movement behavior were measured. Results demonstrate that Cuvier's beaked whales (*Ziphius cavirostris*) react most strongly to simulated sonar exposures with clear changes in vocal and diving behavior and avoidance responses at low received sound levels. Blue whale (*Balaenoptera musculus*) responses are more variable, depending on complex interactions of exposure and behavioral conditions. Ongoing efforts include expanding sample sizes in other species using simulated sounds and the novel inclusion of operational mid-frequency sonars.

1:40

2pABa3. Noise design tradeoffs for a general-purpose broadband acoustic recording tag. William C. Burgess (Greeneridge Sci., Inc., 6060 Graham Hill Rd. Stop F, Felton, CA 95018, burgess@greeneridge.com), Susanna B. Blackwell (Greeneridge Science, Inc., Aptos, CA), and Patrick Dexter (Greeneridge Science, Inc., Ojai, CA)

To design instruments for general-purpose use invokes both blessings and curses; blessings because a flexible product may leverage its development effort across many different applications, but curses because it may not be perfect for any of them. Designing for noise performance of broadband acoustic recording tags epitomizes this tradeoff. Lower self noise typically requires more power and a larger transducer, and may come at the expense of clipping strong sounds, all of which impact tag applications. Higher self noise, however, reduces detection range under quiet conditions and diminishes utility for noise monitoring. The electronic design of the Acousonde$^{\text{TM}}$ acoustic/ultrasonic recording tag navigates this challenge using two acoustic channels with very different gains (29–49 versus 14–34 dB), noise (minimum 40 versus 70 dB re 1 $\mu Pa^2/Hz$ at 1 kHz), and bandwidths (42 versus 9.3 kHz). Mechanically, design for hydrodynamics reduces turbulent flow noise especially once the tag aligns with flow direction due to self-orienting. As a result of these design elements, the Acousonde's noise performance is comparable to that of a much larger recording instrument while preserving capability as an animal tag. [Development supported by ONR.]

2:00

2pABa4. The next generation of multi-sensor acoustic tags: Sensors, applications, and attachments. Douglas Nowacek (Nicholas School of the Environ. and Pratt School of Eng., Duke Univ. Marine Lab., 135 Duke Marine Lab Rd., BEAUFORT, NC 28516, dpn3@duke.edu), Matthew Bowers (Marine Sci. and Conservation, Duke Univ. Marine Lab, BEAUFORT, NC), Andrew Cannon (1900 Eng., Greenville, SC), Mark Hindell (Inst. of Marine and Antarctic Sci., Univ. of Tasmania, Hobart, TAS, Australia), Laurens E. Howle (Mech. Eng. and Mater. Sci., Duke Univ., Durham, NC), Mark M. Murray (Mech. Eng., US Naval Acad., Annapolis, MD), Dan Rittschof (Marine Sci. and Conservation, Duke Univ. Marine Lab, Beaufort, NC), K. Alex Shorter, and Michael Moore (Biology Dept., Woods Hole Oceanographic Inst, Woods Hole, MA)

From Kooyman's 1963 wind-up kitchen timer TDR, multi-sensor tags have evolved significantly over the last twenty years. These advancements, including high fidelity acoustics, have been driven by improved sensing and electronics technology, and resulted in highly integrated mechatronics systems for the study of free ranging animals. In the next decade, these tags will continue to improve, and promising work has begun in three key areas: (i) new sensors; (ii) expanding uses of existing sensors; and (iii) increasing attachment duration and reliability. The addition of rapid acquisition GPS and the inclusion of gyroscopes to separate the dynamic acceleration of the animal from gravitational acceleration, are underway but not widely available to the community. Existing sensors could be used for more and different applications, e.g., measuring ambient ocean noise. Tags attached to pinnipeds in the Southern Ocean, for example, could provide noise measurements from remote areas. Finally, attachment duration has been limiting for cetaceans because the suction cups typically used do not reliably stay attached for more than a day. We will present data on engineering efforts to improve attachments: (i) improved tag hydrodynamics; (ii) incorporating bio-compatible glues; and (iii) micro structuring tag components to utilize hydrostatic forces and enhance adhesion.

2:20

2pABa5. The relationship between vessel traffic and noise levels received by killer whales. Juliana Houghton (School of Aquatic & Fishery Sci., Univ. of Washington, Box 355020, Seattle, WA 98195, stephj5@uw.edu), Marla Holt (NOAA/NMFS/Northwest Fisheries Sci. Ctr., Seattle, WA), Deborah Giles (Dept. of Wildlife, Fish, and Conservation Biology, Univ. of California, Davis, CA), Candice Emmons, Brad Hanson (NOAA/NMFS/Northwest Fisheries Sci. Ctr., Seattle, WA), Jeff Hogan (Cascadia Res. Collective, Olympia, WA), Trevor Branch, and Glenn VanBlaricom (School of Aquatic & Fishery Sci., Univ. of Washington, Seattle, WA)

Cetaceans that rely on their acoustic environment for key life history strategies are susceptible to noise effects from anthropogenic use such as ecotourism. Endangered Southern Resident killer whales (SRKW) are the primary target for vessel-based whale-watching in the Salish Sea. Vessel interactions and associated noise have been identified as potential stressors for SRKW. Previous research has indicated that both stressors negatively impact SRKW; however, there is a missing link between vessel characteristics/behavior and noise levels actually received by individual whales. To investigate this relationship, data were collected concurrently using mobile remote sensing survey equipment packages and digital acoustic recording tags. This allowed us to obtain precise geo-referenced vessel data and noise levels received by the whales. We used linear regression to summarize patterns in vessel characteristics and relate them to received noise levels. Received noise levels (RNL) were correlated with the number of vessels. RNL also increased when larger vessels were present or when vessels were traveling at relatively high speed. These findings facilitate improved understanding of the contributions of vessel characteristics to the noise levels received by individual cetaceans. Results from this study can be used to refine existing vessel regulations in order to better manage SRKW to recovery.

2:35

2pABa6. Use of an animal-borne active acoustic tag to conduct minimally-invasive behavioral response studies. Selene Fregosi, Holger Klinck (Cooperative Inst. for Marine Resources Studies, Oregon State Univ. and NOAA Pacific Marine Environ. Lab., Hatfield Marine Sci. Ctr., 2030 SE Marine Sci. Dr., Newport, OR 97365, selene.fregosi@noaa.gov), Markus Horning (Marine Mammal Inst., Oregon State Univ., Newport, OR), David K. Mellinger (Cooperative Inst. for Marine Resources Studies, Oregon State Univ. and NOAA Pacific Marine Environ. Lab., Newport, OR), Daniel P. Costa (Dept. of Ecology and Evolutionary Biology, Univ. of California, Santa Cruz, Santa Cruz, CA), David A. Mann (Loggerhead Instruments, Sarasota, FL), Kenneth Sexton (The Sexton Co., Salem, OR), and Luis Huckstadt (Dept. of Ecology and Evolutionary Biology, Univ. of California, Santa Cruz, Santa Cruz, CA)

A pilot study was conducted to evaluate the potential of animal-borne active and passive acoustic tags for conducting minimally-invasive behavioral response studies on pinnipeds. A prototype tag was developed and tested on juvenile northern elephant seals (*Mirounga angustirostris*) using translocation experiments at Año Nuevo State Park, CA, USA, in spring 2012. The principal scientific questions of this pilot study were (1) do low-intensity sounds emitted by an animal-borne tag elicit behavioral responses, and (2) are potential animal responses related to signal content (e.g., threatening vs non-threatening)? Preliminary results indicate that (1) low-intensity sounds emitted by animal-borne tags elicit distinct behavioral responses, (2) these responses appear related to signal content, and (3) the responses may differ based on depth, bathymetry, and location. The results of the study show the promise of this approach as a minimally invasive and cost-effective method to investigate animal responses to underwater sounds, as well as a method to develop mitigation strategies. We are currently in the process of improving the tag design for future field efforts with the goal to increase the sample size, range of acoustic stimuli, and age/sex classes of tagged seals. [Funding from NOAA/NMFS Ocean Acoustics Program.]

2:50

2pABa7. Putting tags in the researcher's toolkit: An examination of the strengths, limitations, and added-value from animal tagging. Robert Gisiner (NAVFAC EXWC EV, US Navy, 1000 23rd Ave., Port Hueneme, CA 93043, bob.gisiner@navy.mil)

The US Navy, through the Office of Naval Research and other offices, has focused on improving animal tags, reducing cost, and increasing availability. From data-rich packages like video and acoustic dataloggers to simple location-only tags, tags provide a variety of new data to studies of marine animals and their ecosystems. Tags realize their full potential when calibrated or validated against other existing alternative sensor systems like visual surveys, photo-identification, genetics, and acoustic monitoring. When tag cost, cost of delivery and recovery or monitoring are weighed against the data uniquely available from tags, an integrated data collection strategy involving animal tagging can be developed to generate the best data at the optimal total cost for a given research or resource management scenario.

3:05–3:20 Panel Discussion

Session 2pABb

Animal Bioacoustics: Noise Impacts on Marine Life

Michael A. Stocker, Chair
Ocean Conservation Research, P.O. Box 559, Lagunitas, CA 94938

Contributed Papers

3:30

2pABb1. Passive acoustic monitoring for marine mammals during Navy explosives training events off the coast of Virginia Beach, Virginia. Cara F. Hotchkin, Mandy Shoemaker, Anurag Kumar (Naval Facilities Eng. Command, Atlantic, 6506 Hampton Blvd., Norfolk, VA 23508, cara.hotchkin@navy.mil), Carl Hager (U.S. Naval Acad., Annapolis, MD), David MacDuffee, Jene Nissen, and Ronald Filipowicz (U.S. Fleet Forces Command, Norfolk, VA)

Navy training events involving the use of explosives pose a potential threat to marine mammals. This study used passive acoustic and visual monitoring data to evaluate marine mammals' behavioral responses to noise from explosive events. Monitoring was conducted during five training events in the Virginia Capes (VACAPES) Range Complex during August/September of 2009–2012. Passive acoustic monitoring methods ranged from a single hydrophone to an array of sonobuoys monitored in real time. Visual monitoring effort over the five events totaled approximately 34 h (day before events: 10.1 h; days of events: 22.3 h; day after events: 1.5 h), yielding a total of 27 marine mammal sightings. Approximately 54 h of acoustic data were collected before, during, and after the five events. Behavioral changes were evaluated based on analysis of vocalizations detected before, during, and after explosions and concurrent data from visual sightings. For time periods with both visual and acoustic monitoring data, detection methods were compared to evaluate effectiveness. Continuing use and evaluation of both visual and passive acoustic methods for monitoring of explosive training events will improve our knowledge of potential impact resulting from explosive events and help improve management and conservation of marine mammals.

3:45

2pABb2. Use of Automated passive acoustic monitoring methods for monitoring for marine mammals in conjunction with US Navy Mid-frequency Active Sonar training events. Stephen W. Martin, Roanne A. Manzano-Roth, and Brian M. Matsuyama (SSC PAC, 53560 Hull St., Code 71510, San Diego, CA 92152, steve.w.martin@navy.mil)

Automated passive acoustic detection, classification, and localization (DCL) methods are employed to deal with large volumes of acoustic data to support estimating the sound pressure levels (SPLs) that marine mammals are exposed to from mid-frequency active sonar (MFAS) during US Naval training events. These methods are applied to a training event involving MFAS conducted February 2012 in Hawaiian waters with thirty one hydrophones of data collected continuously over an 11 day period. The automated methods detect and determine locations of marine mammals, specifically minke and beaked whales, and the times of the MFAS transmissions utilizing custom C++ algorithms. Streamlined manual validation methods are employed which utilize custom MATLAB display routines. Animal locations uncertainties are addressed for the two different species. Once the transmitting ship and animal locations are determined acoustic propagation modeling is utilized to estimate the sound pressure levels (in dB re 1 micro Pascal) that an animal, or group of animals, were exposed to. Surface ducted propagation conditions can result in species such as beaked whales being exposed to over 30 dB higher SPL's when they return to the surface to breathe compared to when at depth foraging.

4:00

2pABb3. Impact of underwater explosions on cetaceans. Simone Baumann-Pickering (Scripps Inst. of Oceanogr., Univ. of California San Diego, 9500 Gilman Dr., La Jolla, CA 92093, sbaumann@ucsd.edu), Amanda J. Debich, Ana Širović (Scripps Inst. of Oceanogr., Univ. of California San Diego, San Diego, CA), James V. Carretta (Southwest Fisheries Sci. Ctr., National Oceanic and Atmospheric Administration, San Diego, CA), Jennifer S. Trickey, Rohen Gresalfi (Scripps Inst. of Oceanogr., Univ. of California San Diego, San Diego, CA), Marie A. Roch (Dept. of Comput. Sci., San Diego State Univ., San Diego, CA), Sean M. Wiggins, and John A. Hildebrand (Scripps Inst. of Oceanogr., Univ. of California San Diego, San Diego, CA)

Use of seal bombs to deter sea lions from being caught in nets and preying on catch is a common practice for a number of fisheries. Purse seine fisheries in Southern California target primarily squid, but also scombrids and baitfish such as sardine and anchovy, while set gillnet fisheries' primary catch are halibut and white seabass. All of these fisheries use seal bombs as deterrents. Continuous passive acoustic recordings at several sites in the Southern California Bight collected since 2007 revealed an extensive use of smaller explosives, most likely seal bombs, during nighttime hours with a seasonal occurrence matching fishery activities. During several months of the year they were used all night, every night. The median occurrence of explosions when detected was 8 per hour; however, during periods of high fishing effort they reached up to 480 explosions per hour. From behavioral response and opportunistic studies we know that beaked whales as well as endangered blue whales react negatively to anthropogenic sound sources. We are testing the hypothesis that these underwater explosions have a suppressive effect on the acoustic behavior and therefore the communication and foraging of cetaceans, possibly leading to impacts on the individual fitness and overall population health.

4:15

2pABb4. Monitoring of marine mammal occurrence and acoustic behaviors in relation to mid-frequency active sonar using autonomous recorders deployed off the undersea warfare training range, Florida. Thomas F. Norris, Julie Oswald, Tina M. Yack, Elizabeth Ferguson (Bio-Waves, Inc., 144 W. D St., Ste. #205, Encinitas, CA 92024, thomas.f.norris@bio-waves.net), Anurag Kumar (Naval Facilities Eng. Command Atlantic, U.S. Navy, Norfolk, VA), Jene Nissen (U.S. Fleet Forces Command, U.S. Navy, Norfolk, VA), and Joel Bell (Naval Facilities Eng. Command Atlantic, U.S. Navy, Norfolk, VA)

Passive acoustic data were collected from nine Marine Autonomous Recording Units (MARUs) deployed 60–150 km in an area that coincides with the U.S. Navy's planned Undersea Warfare Training Range (USWTR) off Jacksonville FL. MARUs were deployed for 26 days during fall 2009, and 37 days in winter 2009–2010. Data were manually reviewed for marine mammal vocalization events, man-made noise, and mid-frequency active sonar events, which were logged using TRITON software. Seasonal and diel patterns were characterized qualitatively. Patterns and probabilities of vocalization events by species, or species groups, were related to sonar events. Vocalizations were detected for minke whales, North Atlantic right

2p TUE. PM

whales, sei whales, humpback whales, sperm whales, the blackfish group, and delphinids. Minke whale pulse-trains occurred almost continuously during the winter deployment but were absent in fall. Right whale events occurred mostly during winter at shallow-water sites, but unexpectedly were also detected at deep-water sites. Sperm whale events occurred exclusively near the continental shelf break and exhibited a strong diel pattern. Minke whale events had a strong negative relationship with sonar events. These results provide an initial assessment of marine mammal occurrence within the Navy's planned USWTR, and provide new information on vocalization events in relation to sonar.

4:30

2pABb5. Vocalization behaviors of minke whales in relation to sonar in the planned Undersea Warfare Training Range off Jacksonville, Florida. Talia Dominello, Thomas Norris, Tina Yack, Elizabeth Ferguson, Cory Hom-Weaver (Bio-Waves Inc., 364 2nd St. #3, Encinitas, CA 92024, talia.dominello@bio-waves.net), Anurag Kumar (Naval Facilities Eng. Command Atlantic, Norfolk, VA), Jene Nissen (U.S. Fleet Forces Command, Norfolk, VA), and Joel Bell (Naval Facilities Eng. Command Atlantic, Norfolk, VA)

Nine Marine Autonomous Recording Units (MARU's) were deployed in a rectangular array at a site coinciding with the United States (U.S.) Navy's planned Undersea Warfare Training Range (USWTR) approximately 60–150 km offshore Jacksonville, FL (13 September to 8 October and 3 December to 8 January, 2009–2010) at shallow, mid-depth, and deep sites (45, 183, 305 m). Data were reviewed in detail using TRITON (Wiggins, 2007). Event logs were created for each day at every site. Custom-written MATLAB scripts were used to calculate the probability of minke whale vocalization events occurring in the presence and in the absence of mid-frequency sonar. Minke whale vocalization events were completely absent in the fall deployment period, but occurred almost continuously during the winter deployment, indicating a strong seasonal pattern of occurrence. Minke whale vocalizations were detected most frequently at deep-water sites, and only at low levels ($<$0.03% of time) at shallow-water sites. Results of the probability analysis indicated a strong negative correlation to sonar. Minke whale vocalization events were greatly reduced, or completely ceased, during most days with nearly continuous sonar events during an approximate 3-day period. To our knowledge, such changes in acoustic behaviors of minke whales in relation to sonar have not been reported before.

4:45

2pABb6. Behavioral responses of California sea lions (*Zalophus californianus*) to controlled exposures of mid-frequency sonar signals. Dorian S. Houser (Dept. of Conservation and Biological Res., National Marine Mammal Foundation, 2240 Shelter Island Dr., San Diego, CA 92106, dorian.houser@nmmfoundation.org), Stephen W. Martin, and James J. Finneran (US Navy Marine Mammal Program, SSC Pacific, San Diego, CA)

Acoustic dose-response functions can be used to explore the relationship between anthropogenic noise exposure and changes in marine mammal behavior. Fifteen sea lions participated in a controlled exposure study to determine the relationship between the received sound pressure level (SPL) of a mid-frequency sonar signal (1-s duration, 3250-3450 Hz) and behavioral deviations from a trained behavior. Sea lions performed 10 control trials followed by 10 exposure trials within an open-water enclosure. Acoustic playbacks occurred once during each exposure trial when the sea lion crossed the middle of the enclosure. Received levels, ranging from 125 to 185 dB re 1 μPa (rms) SPL, were randomly assigned but were consistent across all trials for each individual. Blind scoring of behavioral responses was performed for all trials. A canonical correlation analysis indicated that cessation of the trained behavior, haul-out, a change in respiration rate, and prolonged submergence were reliable response indicators. Sea lions showed both an increased responsiveness and severity of response with increasing received SPL. No habituation to repeated exposures was observed, but age was a significant factor affecting the dose-response relationship. Response patterns and factors affecting behavioral responses were different from those observed in bottlenose dolphins and are indicative of species-specific sensitivities.

5:00

2pABb7. The characteristics of boat noise in marine mammal habitats. Christine Erbe (Ctr. for Marine Sci. & Technol., Curtin Univ., Kent St., Bentley, WA 6102, Australia, c.erbe@curtin.edu.au)

Environmental management of underwater noise tends to focus on sources related to offshore oil and gas exploration and navy operations. However, in many, specifically near-shore habitats, boat noise prevails, yet remains largely unregulated. Underwater noise of personal watercraft (jet skis) was recorded in Queensland, and consisted of broadband energy between 100 Hz and 10 kHz attributed to the vibrating bubble cloud generated by the jet stream, overlain with frequency-modulated tonals corresponding to impeller blade rates and harmonics. Broadband monopole source levels were 149, 137, and 122dB re 1 μPa @ 1m (5th, 50th, and 95th percentiles). Underwater noise of zodiacs (inflatable boats with outboard motors) operated by whale-watching companies was recorded in southern British Columbia. At slow cruising speeds (10 km/h), underwater noise peaked between 50 and 300 Hz; at fast traveling speeds (55 km/h), underwater noise peaked between 100 and 3000 Hz, exhibiting strong propeller blade rate tonals at all speeds. Broadband source levels increased with speed from 126 to 170dB re 1 μPa @ 1m according to SL = 107 + 32 log10(-speed/km/h). Even though noise levels from jetskis are lower than those of propeller-driven boats, it is not necessarily the broadband source level that correlates with the bioacoustic impact on marine fauna.

5:15

2pABb8. Complex masking scenarios in Arctic environments. Jillian Sills (Ocean Sci., Univ. of California at Santa Cruz, 100 Shaffer Rd., Santa Cruz, CA 95050, jmsills@ucsc.edu), Colleen Reichmuth (Inst. of Marine Sci., Univ. of California at Santa Cruz, Santa Cruz, CA), and Brandon L. Southall (Southall Environmental Associates, Aptos, CA)

Critical ratios obtained using octave-band noise and narrowband signals provide a useful first approximation for understanding the effects of noise on hearing. When considering realistic listening scenarios, it may be necessary to examine the effects of spectrally complex, time-varying noise sources on an animal's ability to detect relevant signals. In the case of Arctic seals, the increasing prevalence of seismic exploration makes an examination of masking by impulsive sounds particularly relevant. However, the characteristics of received sounds from airgun operations vary dramatically depending on the seismic source, environmental parameters, and distance. In order to determine the potential for auditory masking by airguns, we developed a paradigm to quantify the influence of spectral and temporal variations in typical seismic noise on signal detectability. This method calls for calculation of detection probabilities for seals listening for the same signal embedded at different time windows within a background of distant airgun noise. We believe this approach will enable an experimental assessment of masking potential by impulsive noise as distance between the receiver and source is increased. Such an assessment will aid in determining the extent to which standard critical ratio data can be reasonably applied in complex masking scenarios. [Work supported by OGP-JIP.]

Session 2pBA

Biomedical Acoustics: Nanobubbles, Nanoparticles, and Nanodroplets for Biomedical Acoustics Applications

Jonathan Mamou, Cochair
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Invited Papers

1:15

2pBA1. Ultrasound-targeted delivery of systemically administered therapeutic nanoparticles. Kelsie Timbie, Caitlin Burke (Biomedical Eng., Univ. of Virginia, Box 800759, Health System, Charlottesville, VA 22908, rprice@virginia.edu), Elizabeth Nance, Graeme Woodworth (Ctr. for Nanomedicine, Johns Hopkins Univ., Baltimore, MD), Grady W. Miller (Biomedical Eng., Univ. of Virginia, Charlottesville, VA), Justin Hanes (Ctr. for Nanomedicine, Johns Hopkins Univ., Baltimore, MD), and Richard J. Price (Biomedical Eng., Univ. of Virginia, Charlottesville, VA)

The ultrasound (US)-targeted delivery of systemically administered drug and gene-bearing nanoparticles has emerged to become a robust area of investigation with clear clinical potential. Such approaches typically entail the concurrent injection of contrast agent microbubbles (MBs) and nanoparticles, followed by the application of US to the region of interest. US-activated MBs disrupt the surrounding microvessel, permitting nanoparticle delivery with precise spatial localization. Our group has previously shown that US-targeted nanoparticle delivery can amplify collateral artery growth, that the binding of nanoparticles to MBs enhances nanoparticle delivery, that non-viral gene nanocarrier transfection is dependent on both MB diameter and US pressure, and that solid tumor growth can be controlled by the US-targeted delivery of 5 FU nanoparticles. More recent studies center on developing MRI-guided focused ultrasound (FUS) for nanoparticle delivery across the blood brain-barrier (BBB), which is the foremost impediment to drug treatment for most central nervous system diseases. Importantly, densely PEGylated nanoparticles that have been specifically designed to penetrate brain tissue (i.e., brain-penetrating nanoparticles, BPNs) are capable of crossing the BBB after it is opened with FUS and MBs, thereby supporting the use of gene- and drug-bearing BPNs in combination with MR-guided FUS in treating disorders and pathologies of the CNS.

1:35

2pBA2. Ultrasound-mediated stimulation of nanoparticles for therapeutic applications. Tyrone M. Porter (Mech. Eng., Boston Univ., 110 Cummington St., Boston, MA 02215, tmp@bu.edu)

Ultrasound has the unique capability to change the pressure and temperature of tissue in a noninvasive and highly localized manner. Consequently, ultrasound is an attractive tool for enhancing the delivery of drugs to targeted cells and tissues. This talk will review a variety of stimuli-responsive drug carriers (i.e., liposomes, nanoparticles, emulsions, etc.) and the role of ultrasound in facilitating transport and drug delivery.

1:55

2pBA3. Low boiling point perfluorocarbon nanodroplets for biomedical imaging and therapeutics. Paul S. Sheeran (Biomedical Eng., Univ. of North Carolina at Chapel Hill and North Carolina State Univ., Chapel Hill, NC), Terry O. Matsunaga (Radiology, Univ. of Arizona, Tucson, AZ), and Paul A. Dayton (Biomedical Eng., Univ. of North Carolina at Chapel Hill and North Carolina State Univ.,Chapel Hill, NC 27599, padayton@bme.unc.edu)

Perfluorocarbon (PFC) droplets have been examined for biomedical applications for more than a decade. Of particular interest are "phase change" PFC droplets, which can convert from liquid to gaseous phase with an acoustic energy input. In liquid form, these droplets demonstrate stability not capable with microbubbles, but in gaseous form the resultant bubble provides the cavitation source that is crucial for ultrasound imaging or therapeutic mediation. Traditionally, phase-change droplets have utilized PFCs with boiling points close to body temperature, so that they could be readily activated with acoustic energy input. However, the desire reduce these droplets to hundred-nanometer sizes in order to reach the extravascular space has made PFC selection more complicated. Specifically, the influence of surface tension on droplets that are significantly smaller than a micron elevates the boiling point past that of the bulk fluid, making hundred-nanometer sized droplets challengingly difficult to vaporize with diagnostic ultrasound energy levels. Our solution has been a unique approach to condense low boiling point gaseous perfluorocarbons, creating a population of meta-stable PFC nanodroplets. The resulting nanodroplets exhibit stability, yet have very low energy activation requirements. We discuss advantages and disadvantages of this new formulation, and demonstrate applications in biomedical imaging and therapeutics.

2:15

2pBA4. Entropy imaging and nanostructures. Michael Hughes (School of Medicine, Washington Univ., 1632 Ridge Bend Dr., St Louis, MO 63108, mshatctrain@gmail.com), John E. McCarthy (Dept. of Mathematics, Washington Univ., St Louis, MO), Jon N. Marsh, and Samuel A. Wickline (School of Medicine, Washington Univ., St Louis, MO)

Virtually all imaging devices today function by collecting either electromagnetic or acoustic waves and using the energy carried by these waves to determine pixel values to build up what is basically an "energy" picture. However, waves also carry information and this can also be used to determine the pixel values in an image. We have employed several measures of information all of which are based on different forms of entropy. Numerous published studies have demonstrated the advantages of entropy, or "information," imaging over conventional methods in materials characterization and medical imaging. Moreover, the technique is robust as these results were obtained using a variety of imaging systems having different frequency ranges and transducers. Similar results have also been obtained using microwaves. We will present the results of several entropy-imaging-based *in vivo* studies. The first of these were based on a non-gaseous liquid-filled nanoparticle contrast agent used to image tumors. We will also present a study of therapy monitoring of muscular dystrophy conducted without contrast. In both studies entropy images were able to detect changes in backscattered ultrasonic signals that were not detectable using conventional techniques.

2:35

2pBA5. Characterization of liquid-filled nanoparticles for detection and drug-delivery in tumors. Nicolas Taulier, Thomas Payen, Sara Jafari (Parametric Imaging Lab, CNRS - Univ. Pierre and Marie Curie, 15 rue de l'ecole de medecine, Paris 75006, France, lori.bridal@upmc.fr), Jonathan Mamou (F.L. Lizzi Ctr. for Biomedical Eng., Riverside Res., New York, NY), Fattal Elias, Nicolas Tsapis (Institut Galien Paris-Sud, UMR 8612, LabEx LERMIT, Univ Paris-Sud and CNRS, Châtenay-Malabry, France), and Lori Bridal (Parametric Imaging Lab, CNRS - Univ. Pierre and Marie Curie, Paris, France)

Liquid-filled nanoparticles provide long half-life circulation favoring accumulation in the interstitial space of tumors through enhanced permeation and retention. However, these relatively incompressible nanoparticles are much less echogenic than those with a gas-core. We have worked to characterize and optimize the acoustic response of PLGA-shelled, perfluorooctyl bromide (PFOB) core nanoparticles (radii 70 to 200 nm ; shell-thickness-to-radius ratio 0.25 to 1). Acoustic response (attenuation coefficient, ultrasonic velocity, and relative backscattered intensity) was explored *in vitro* for ranges of concentration, shell-thickness, acoustic pressures and pulse durations (20 to 40 MHz). Modification of the surface chemistry of the polymeric shell with fluorescent, pegylated, or/and biotinylated phospholipids was not associated with apparent response modification. Successful incorporation of paclitaxel in the shell has been achieved but currently the thick-shells of these nanoparticles impede ultrasound-triggered delivery. Although work remains to better adapt liquid-filled nanoparticles for detection and delivery, trials demonstrating nanoparticle detection *in vivo* in mice indicate their potential for accumulation, and detection in tumors.

2:55

2pBA6. Ultrasound-mediated delivery of bioactive nanobubbles to vascular tissue. Jonathan T. Sutton, Jason L. Raymond (College of Eng. and Appl. Sci.; Biomedical Eng., Univ. of Cincinnati, 231 Albert Sabin Way, 3940 Cardiovascular Ctr., Cincinnati, OH 45267, suttonjt@mail.uc.edu), Michael C. Verleye (College of Eng.; Dept. of Chemical Eng., Notre Dame Univ., Notre Dame, IN), Gail J. Pyne-Geithman (College of Medicine; Dept. of Neurosurgery, Univ. of Cincinnati, Cincinnati, OH), Jack Rubinstein, and Christy K. Holland (College of Medicine; Internal Medicine, Div. of Cardiovascular Diseases, Univ. of Cincinnati, Cincinnati, OH)

Bubble liposomes (BLs) are under development for ultrasound-triggered release of a potent vasodilator within the vasculature. Nano-sized vesicles facilitate this process by enclosing bubbles to enhance ultrasound image contrast, and deliver therapeutic agents. Assessment of drug delivery in living tissue allows for mechanistic pathways to be revealed. In this study, we used a novel *ex vivo* model to assess the vascular effects of ultrasound-mediated delivery of a bioactive gas-nitric oxide (NO)-from nanobubble liposomes. Porcine carotid arteries were excised post-mortem and mounted in physiologic buffer. Vascular tone was assessed in real time by coupling the artery to an isometric force transducer. NO-loaded BLs were infused into the lumen of the artery, which was exposed to 1-MHz pulsed ultrasound, while acoustic cavitation emissions were monitored. Changes in vascular tone were concurrently measured and compared to control and sham NO exposures. Our results demonstrate that ultrasound-triggered NO release from BLs induces potent vasorelaxation within porcine carotid arteries. This approach is a valuable mechanistic tool to assess the bioeffects that NO elicits within the vasculature upon release from BLs exposed to 1-MHz ultrasound.

Contributed Papers

3:15

2pBA7. Acoustic droplet vaporization is initiated by superharmonic focusing. Oleksandr Shpak (Phys. of Fluids, Univ. of Twente, Witbreuksweg 379-404, Enschede 7522ZA, Netherlands, o.shpak@utwente.nl), Martin Verweij (Dept. of Imaging Sci. and Technol., Delft Univ. of Technol., Delft, Netherlands), Rik Vos, Nico de Jong (Biomedical Eng., Erasmus MC, Rotterdam, Netherlands), Detlef Lohse, and Michel Versluis (Phys. of Fluids, Univ. of Twente, Enschede, Netherlands)

Acoustically sensitive emulsion droplets composed of a liquid perfluorocarbon have the potential to be a highly efficient system for local drug delivery, embolotherapy or for tumor imaging. The physical mechanisms underlying the acoustic activation of these phase-change emulsions into a bubbly dispersion, termed acoustic droplet vaporization, have not been well understood. The droplets have a very high activation threshold, its frequency dependence does not comply with homogeneous nucleation theory and focusing spots have been observed. We showed that acoustic droplet vaporization is initiated by a combination of two phenomena: highly nonlinear distortion of the acoustic wave before it hits the droplet, and focusing of the distorted wave by the droplet itself. At high excitation pressures, nonlinear distortion causes significant superharmonics with wavelengths below the diameter of the droplet. Because these superharmonics strongly contribute to the focusing effect, the mechanism also explains pressure thresholding effects. In an accompanying paper, mathematical modeling aspects are presented. A proposed model is validated with experimental data captured with an ultra high-speed camera on the positions of the nucleation spots.

Moreover, the presented mechanism explains the hitherto counterintuitive dependence of the nucleation threshold on the ultrasound frequency.

3:30

2pBA8. Subwavelength droplets in nonlinear ultrasound fields: Simulation of focusing effects. Martin D. Verweij (Acoust. Wavefield Imaging, Delft Univ. of Technol., P.O. Box 5046, Delft 2600GA, Netherlands, m.d.verweij@tudelft.nl), Oleksandr Sphak (Phys. of Fluids Group, Univ. of Twente, Enschede, Netherlands), Hendrik J. Vos, Nico de Jong (Biomedical Eng., Erasmus MC, Rotterdam, Netherlands), Detlef Lohse, and Michel Versluis (Phys. of Fluids Group, Univ. of Twente, Enschede, Netherlands)

Ultrasound can trigger the evaporation of tiny droplets of emulsified, superheated fluids like perfluorocarbon. This acoustic droplet vaporization (ADV) effect is important because of its potential medical applications. For example, drug-loaded nanodroplets can penetrate the vessel wall and subsequently release their therapeutic load through acoustic activation. Until now, the medical application of ADV has been limited by lack of understanding of the acoustic activation mechanism. In an accompanying paper, a mechanism is proposed that can fully explain the experimentally observed phenomena such as a frequency dependent pressure threshold. The mechanism involves focusing of higher harmonics of the nonlinear activation field. In the current presentation it will be explained how the focusing effect is simulated. A particular problem here is that typical droplet sizes are in the order of micrometers, while typical wavelength sizes are in the order of hundreds of micrometers. This difference in scale will render the traditional analytic solution numerically useless. The problem is avoided by using appropriate expansions of the functions involved. Numerical results are presented that show a focusing effect inside the droplet for a range of incident harmonics. These results have enabled the demonstration of the mechanism in the accompanying paper.

3:45

2pBA9. Characterization of nanometric ultrasound contrast agents with a liquid perfluorocarbon core. Ksenia Astafyeva (Lab. of Parametric Imaging, Pierre & Marie Curie Univ., 15 rue de l'ecole de medecine, Paris 75013, France, ksenia.astafyeva@gmail.com), Jean-Marc Conoir, Matthieu Guédra (Inst. of Jean Le Rond d'Alembert, Paris, France), Elias Fattal (Institut Galien, Univ. Paris South, Chatenay-Malabri, France), Christine Pepin, Ange Polidori (Univ. of Avignon, Avignon, France), Nicolas Taulier (Lab. of Parametric Imaging, Pierre & Marie Curie Univ., Paris, France), Jean-Louis Thomas (Inst. of NanoSci., Paris, France), Nicolas Tsapis (Institut Galien, Univ. Paris South, Chatenay-Malabri, France), Tony Valier-Brasier (Inst. of Jean Le Rond d'Alembert, Paris, France), and Wladimir Urbach (Lab. of Parametric Imaging, Pierre & Marie Curie Univ., Paris, France)

In this work, we propose a new type of nanometric ultrasound contrast agents (nUCA) with a liquid core and we model their acoustic propagation through their dilute solutions. These capsules have a shell made of a biocompatible polymer or fluorinated surfactants and a liquid perfluorocarbon core to undergo a high lifetime. The capsules are small enough (from 100 nm to 1 μm) to pass through the tumor endothelium, and they remain stable *in vitro* for several months. Ultrasound attenuation and speed of sound measurements through dilute suspensions of nUCA were carried out from 3 to 90 MHz at various temperatures and concentrations. The acoustic propagation was modeled by combining (i) a dilatational mode taking into account the radial oscillations of the capsules, and (ii) a translational mode of oscillations induced by viscoinertial interaction with the continuous phase. The model makes possible to fit with good accuracy the experiments using values compatible with literature data. Moreover, it reveals information about unknown parameters of the shell: for instance, the viscoelastic shell has to be described as a Maxwell rheological medium. [This work was supported by Emergence-UPMC program.]

4:00–4:15 Break

4:15

2pBA10. Ultrasonic propagation in suspensions of encapsulated compressible nanoparticles. Matthieu Guédra (Institut Jean le Rond d'Alembert, Université Pierre et Marie Curie - Paris 6, 4 Pl. Jussieu, Paris 75252, France, matthieu.guedra@dalembert.upmc.fr), Ksenia Astafyeva (Laboratoire d'Imagerie Paramétrique, Université Pierre et Marie Curie - Paris 6, Paris, France), Jean-Marc Conoir, François Coulouvrat (Institut Jean le Rond d'Alembert, Université Pierre et Marie Curie - Paris 6, Paris, France), Nicolas Taulier (Laboratoire d'Imagerie Paramétrique, Université Pierre et Marie Curie - Paris 6, Paris, France), Jean-Louis Thomas (Institut des Nano-Sci. de Paris, Université Pierre et Marie Curie - Paris 6, Paris, France), Wladimir Urbach (Laboratoire d'Imagerie Paramétrique & Laboratoire de Physique Statistique de l'ENS, Université Pierre et Marie Curie - Paris 6, Paris, France), and Tony Valier-Brasier (Institut Jean le Rond d'Alembert, Université Pierre et Marie Curie - Paris 6, Paris, France)

Dispersion and absorption are examined for dilute suspensions of encapsulated droplets of nanometric size, with typical radii around 100 nm. This new generation of contrast agents is designed for targeted delivery of drugs. Compared to standard contrast agents used for imaging, particles are of smaller size to pass the endothelial barrier, their shell made up of biocompatible material is stiffer to undergo a longer time life and they have a liquid (PFC) instead of a gaseous core. Ultrasound propagation of these suspensions is modeled by combining (i) a dilatational mode of oscillation assuming a compressible shell with a visco-elastic behavior of Kelvin-Voigt or Maxwell type (relaxation), (ii) a translational mode of oscillation induced by visco-inertial interaction with the ambient fluid, and (iii) polydispersion in terms of radius and shell thickness. Influence of the various effects will be examined. Experimental measurements of the dispersion and absorption properties of nanodroplets solutions over the 1–100 MHz frequency range are performed for various temperatures and concentrations. They allow to fit with good accuracy the model properties and estimate some unknown mechanical properties of the shell of the nanodroplets. [Work supported by programme Emergence-UPMC—project NACUNAT and by Inserm/Plan Cancer—project NABUCCO.]

4:30

2pBA11. Dual perfluorocarbon nanodroplets enhance high intensity focused ultrasound heating and extend therapeutic window *in vivo*. Linsey C. Phillips, Paul S. Sheeran, Connor Puett (Joint Dept. of Biomedical Eng., UNC at Chapel Hill, and NC State Univ., CB 7575, Univ. of North Carolina, Chapel Hill, NC 27599, linsey@email.unc.edu), Kelsie F. Timbie, Richard J. Price (Biomedical Eng., Univ. of Virginia, Charlottesville, VA), G. Wilson Miller (Radiology, Univ. of Virginia, Charlottesville, VA), and Paul A. Dayton (Joint Dept. of Biomedical Eng., UNC at Chapel Hill, and NC State Univ., Chapel Hill, NC)

Perfluorocarbon microbubbles are known to enhance high intensity focused ultrasound (HIFU) ablation by cavitation. However, they can result in superficial skin heating, minimizing their clinical translation. Perfluorocarbon nanodroplets activate only at the higher pressures present at the acoustic focus. We hypothesized that a mixed perfluorocarbon nanodroplet formulation would minimize surface heating while still enhancing ablation. Tissue-mimicking phantoms containing microbubbles or nanodroplets were sonicated (1 MHz, 15 W, 60 s) to assess heating and lesion formation *in vitro*. Microbubbles or nanodroplets were injected into rats ($n = 3$) and HIFU (1 MHz, 15 W, 15 s) was focused into each liver while under MRI guidance. Temperature throughout the liver was tracked by MR thermometry. *In vitro*, microbubbles caused excess surface heating during HIFU, whereas nanodroplets did not. *In vivo*, microbubbles typically circulate for less than 15 min. In comparison, the nanodroplets remained viable in circulation for at least 96 min. HIFU lesions of consistent volume were produced during this time and reached the same maximal temperature rise (Δ550 °C). In the absence of nanodroplets or microbubbles, significantly greater power (25 W) and twice as much time (30 s) was required to generate an ablation lesion in the liver. These results demonstrate the ability of nanodroplets to more safely shorten ablation procedures.

4:45

2pBA12. Use of micro- and nano-sized inertial cavitation nuclei to trigger and map drug release from cavitation-sensitive liposomes. Susan M. Graham, Rachel S. Myers, James Choi, Miriam Bazan-Peregrino (Inst. of Biomedical Eng., Dept. of Eng. Sci., Univ. of Oxford, ORCRB, Oxford OX3 7DQ, United Kingdom, susan.graham@eng.ox.ac.uk), Leonard Seymour (Dept. of Oncology, Univ. of Oxford, Oxford, United Kingdom), Robert Carlisle, and Constantin C. Coussios (Inst. of Biomedical Eng., Dept. of Eng. Sci., Univ. of Oxford, Oxford, United Kingdom)

Encapsulation of cytotoxic drugs into liposomes enhances pharmacokinetics and improves passive accumulation in tumors. However, stable liposomes have limited drug release, and thus action, at the target site. This inefficient and unpredictable drug release is compounded by a lack of low-cost, non-invasive methods to map release in real time. We present a new liposomal vehicle that is exclusively triggered by inertial cavitation. Ultrasound exposure of these liposomes in the absence of SonoVue® provided no increase in drug release, whilst with SonoVue® at inertial cavitation pressure levels a substantial (30%) and significant ($p < 0.001$) increase was observed *in vitro*. A 16-fold increase in the level of drug release within tumors was similarly observed in the presence of inertial cavitation following intravenous delivery. Passive acoustic mapping of inertial cavitation sources during delivery was also found to correlate strongly with the presence of release. However, variability in tumor perfusion indicated that uneven distribution of micron-sized SonoVue® may limit this approach. Nano-scale cavitation nuclei, which may more readily co-localize with 140 nm liposomes, were thus developed and showed similar cavitation energies to SonoVue® *in vitro*. These nano-nuclei may ultimately provide a more reliable and uniform way to trigger drug release *in vivo*.

5:00

2pBA13. Time lapse observation of phase change nano droplet after vaporization stimulated by ultrasound. Kenji Takehara, Takashi Azuma (BioEng., The Univ. of Tokyo, 7-3-1 Hongou, Bunkyou-ku, Tokyo 113-8656, Japan, k.take531@gmail.com), Kiyoshi Yoshinaka (National Inst. of Adv. Industrial Sci. and Technol., Tsukuba, Ibaraki, Japan), Satoshi Yamaguchi (Chemistry and BioTechnol., The Univ. of Tokyo, Bunkyou-ku, Tokyo, Japan), Miyuki Maezawa (Olympus Corp., Shinjuku-ku, Tokyo, Japan), Ichirou Sakuma (Precision Eng., The Univ. of Tokyo, Bunkyou-ku, Tokyo, Japan), Teruyuki Nagamune (Chemistry and BioTechnol., The Univ. of Tokyo, Bunkyou-ku, Tokyo, Japan), Shu Takagi, and Yoichirou Matsumoto (Mech. Eng., The Univ. of Tokyo, Bunkyou-ku, Tokyo, Japan)

Small enough to permeate through tumor blood vessel, and can be detect by ultrasound, phase change nano droplet (PCND) have been studied as contrast agents and therapeutic sensitizer for career. To investigate performance for these purpose, we investigated physical behavior of PCND, especially a lifetime of microbubbles generated by ultrasound stimulation. To investigate bubble's behavior after phase change, we observed a time-lapse change of bubbles population with following method. Focused transducer with a frequency of 3.3 MHz was placed in a water bath of 37 degrees. At the focal point, polyacrylamide gel including PCND was placed. Focused hydrophone was placed perpendicularly to the direction of ultrasound propagation. Two kind of ultrasound pulse wave was used; phase change pulse at the beginning and observation pulse at every 500 μs. The amplitude of scattering signal (SS) reflects the sum of scattering cross-section of bubbles. The time lapse observation of PCND after phase change showed two kinds of behavior, quick and slow decay of scattering signal from the bubbles. The unique increase of SS from the phase-changed bubbles in the monitoring phase was observed. We improved the experimental setup to measure bubble's population properly from various directions using 1-D array transducer and will present the result.

5:15

2pBA14. Detection of unique acoustic signatures for phase-change contrast agents used in medical imaging and therapy. Paul S. Sheeran, Karl H. Martin (Joint Dept. of Biomedical Eng., Univ. of North Carolina and North Carolina State Univ., 10 DuxforDC., Durham, NC 27703, pssheeran@gmail.com), Jordan N. Hjelmquist (Dept. of Biomedical Eng., North Carolina State Univ., Raleigh, NC), Terry O. Matsunaga (Dept. of Medical Imaging, Univ. of Arizona, Tucson, AZ), and Paul A. Dayton (Joint Dept. of Biomedical Eng., Univ. of North Carolina and North Carolina State Univ., Chapel Hill, NC)

Phase-change contrast agents (PCCAs) provide a dynamic platform to approach problems in medical ultrasound (US). Upon US-mediated activation, the liquid core vaporizes and expands to produce a gas bubble ideal for US imaging and therapy. In this study, we demonstrate through underlying theory, high-speed microscopy, and US interrogation that PCCAs composed of highly volatile perfluorocarbons (PFCs) exhibit unique acoustic behavior that can be differentiated from tissue and standard microbubble contrast agents. Experimental results show that when activated with short pulses PCCAs will over-expand ($6.3\times$ to $7.7\times$ the droplet diameter, PFC dependent) due to momentum of expansion, and undergo unforced, under-damped radial oscillation while settling to a final bubble diameter ($5.1\times$ to $5.5\times$ the droplet diameter). Oscillation frequency is inversely related to droplet size—near 100 kHz for droplets ≥ 4 μm in diameter, and 2.5 MHz for droplets near 500 nm. Results from *in vitro* vessel phantoms using confocal piston transducers with an "activate high" (8 MHz, 2 cycles), "listen low" (1 MHz) scheme show that droplet-specific signals can be detected in both time and frequency domain, and that the magnitude of the acoustic "signature" increases with PFC volatility. These signatures may aid in development of droplet-specific detection techniques.

5:30

2pBA15. Study of mechanism of sonoporation using lipid bilayer and surface-modified microbubble. Kodai Hirose, Takashi Azuma (Mech. Eng., The Univ. of Tokyo, 7-3-1, Hongo, Bunkyo-ku, Tokyo 113-8656, Japan, khirose@fel.t.u-tokyo.ac.jp), Kiyoshi Yoshinaka (National Inst. of Adv. Industrial Sci. and Technol., Tsukuba, Japan), Akira Sasaki, Shu Takagi, and Yoichiro Matsumoto (Mech. Eng., The Univ. of Tokyo, Tokyo, Japan)

Microbubble enhanced sonoporation is one of gene therapies and expected to be safe, less invasive, and controllability of treatment area, but its induction rate is very low and its mechanism remains to be explained. The objective is to analyze the mechanism and obtain an optimal design. Because microbubble density distribution was unstable during sonication and ultrasound intensity on the cell surface was affected by distribution change through attenuation change during propagation, microbubble distribution should be controlled and localized only near the membrane to realize high reproducibility. An artificial lipid bilayer modified by biotin-avidin to bind microbubbles was used for this purpose. We have three steps: build lipid bilayer, binding with microbubbles, and introduction of sonication and observation system. We built the lipid bilayer with a diameter of 1 mm using Black Lipid Membrane method. With a capacitance measurement with impedance analyzer (NF, ZM2375), the lipid bilayer thickness was confirmed and its duration is more than an hour. Then, we introduced the sonication and observation system to the lipid bilayer, and we are observing a behavior of the lipid bilayer under irradiation of ultrasound. Afterward we will observe that of a lipid bilayer modified of microbubbles.

5:45

2pBA16. Optimization of acoustic parameters and nanodroplet concentration for spatially controlled, reduced energy high intensity focused ultrasound ablation. Andrew C. Puett (Biomedical Eng., UNC, 6317 S Bradley Overlook, Wilmington, NC 28403, connorpuett@gmail.com), Linsey C. Phillips, Paul S. Sheeran, and Paul A. Dayton (Biomedical Eng., UNC, Chapel Hill, NC)

Background: Perfluorocarbon (PFC)-nanodroplets (ND) provide cavitation sites when vaporized to microbubbles by acoustic energy and lower the power required to ablate tissue by high intensity focused ultrasound (HIFU). However, control over ablation can be problematic. This study explored vaporization, ablation, and PFC-ND concentration *in vitro* to optimize the acoustic pressure (intensity) and insonation time required for spatially controlled HIFU enhancement. Methods: HIFU (continuous wave; 1MHz; 5–20 s; 2–4 MPa) was applied to albumin-acrylamide gels containing PFC-agent (1:1 mix of volatile decafluorobutane and more stable dodecafluoropentane at 10^5-10^7 ND/mL). Controlled ablation was defined as the production of cigar-shaped lesions centered at the acoustic focus. Results: Vaporization field change from "cigar" to "tadpole" began at 5×10^5, 2.5×10^6, and 1×10^7 ND/mL using 4, 3, and 2 MPa, respectively. The volumes of the microbubble clouds (8–200 mm^3) and ablation lesions (1–135 mm^3) within them were dependent on acoustic intensity, insonation time, and PFC-ND concentration. Conclusions: Ablation within microbubble clouds of predictable size, shape, and location can be generated in gels containing PFC-ND using intensities $\leq$650 W/cm^2. Also, pressures and insonation times can be selected to achieve an ablation lesion of desired size for a given PFC-ND concentration. Demonstrating control is an important step toward developing a useful clinical tool.

6:00

2pBA17. Design and characterization of biocompatible perfluorocarbon nanodroplets for theragnostic application. Lucie Somaglino, Ksenia Astafyeva (Laboratoire Imagerie Paramétrique, UMR 7623, CNRS-UPMC, 15, rue de l'école de médecine, EscA 2ème ét., Paris 75005, France, lucie.somaglino@yahoo.fr), Stéphane Desgranges, Ange Polidori, Christiane Contino-Pepin (Institut des Biomolécules Max Mousseron, UMR 5247, CNRS-Université d'Avignon et des Pays de Vaucluse, Avignon, France), Wladimir Urbach, and Nicolas Taulier (Laboratoire Imagerie Paramétrique, UMR 7623, CNRS-UPMC, Paris, France)

We have developed stable emulsions ($\geq$ 3 months) made of nanodroplets (nD) of perfluorocarbon (PFC) dispersed in water to serve as theragnostic agent. nD are stabilized by in house fluorinated surfactants, named FTAC, which chemical structure can be modified to tune their properties. We have characterized nD size distributions (mean diameters from 200 to 600 nm), density, adiabatic compressibility, interfacial tension. US properties of the emulsions have been investigated such as attenuation. Lastly, ultrasonic signals backscattered by nanoemulsions were studied and compared with water to extract signal to noise ratio (SNR), by emitting single negative pulses at $\approx$40 MHz. At similar mean/mode diameters, we showed a strong dependence in SNR values (i) with size distribution, altered by the nature of the surfactant or by a centrifugation/filtration process, (ii) with core nature, and (iii) with nD volume fraction. Besides, hydrophobic drugs such as a thalidomide derivative (with anti-angiogenic properties), have been encapsulated by addition of 10% of triacetin in the nD core. So as to study drug release from nD, a dedicated setup of US cavitation was designed. Cavitation was generated in controlled conditions using a 1 MHz focused transducer (US bursts: 12-22 MPa peak) in a thin wall container immersed in water which temperature and degassed level were kept constant. The generation of cavitation in nD solutions resulted in a strong and reproducible SNR decrease.

TUESDAY AFTERNOON, 3 DECEMBER 2013 MASON, 1:30 P.M. TO 3:10 P.M.

Session 2pEA

Engineering Acoustics, Acoustical Oceanography, and Animal Bioacoustics: Bearing Measurement Methods for Small Wideband Sonars

Kenneth M. Walsh, Chair
K + M Engineering Ltd., 51 Bayberry Lane, Middletown, RI 02842

Chair's Introduction—1:30

Invited Papers

1:35

2pEA1. Use of Bessel side lobe modulation of frequency modulated pulse bearing estimation. Kenneth M. Walsh (K + M Engineering Ltd., 51 Bayberry Ln., Middletown, RI 02842, kwalsh4@mindspring.com)

The beam pattern of a circular transducer is a set of frequency dependent Bessel functions. By using a broad band FM pulse and a 1–3 composite transducer, it appears that the modulation in amplitude and phase due to a reception of the high frequency components can be used to estimate the echo's angle off the transducer axis. The 1–3 composite transducer has a simple amplitude and phase structure.

1:55

2pEA2. Modeling of bio-inspired broadband sonar for high-resolution angular imaging. Jason E. Gaudette (Adv. Acoust. Systems Div., NUWC Div. Newport, 1176 Howell St., B1371/3, Newport, RI, jason.e.gaudette@navy.mil) and James A. Simmons (Dept. of Neurosci., Brown Univ., Providence, RI)

Echolocating mammals perceive images of targets with hyper-resolution and navigate seamlessly through obstacles in complex acoustic environments. The biological solution to imaging with sound is vastly different from man-made sonar. The most prominent difference is that instead of imaging with narrow beams, bats ensonify a large spatial region and exploit broadband echo information to acoustically focus with about one degree of angular resolution. Angular localization may therefore be redefined as a spectral pattern matching problem. By imaging with wider beams, this remarkable performance requires only a single broadband transmitter and two receive elements. Our computational modeling work has led to new insight into the salient spatial information encoded by the bat's auditory system. Although theoretically not required, spatial localization performance increases with the aid of highly complex baffle structures such as those found in biological sonar. Replicating these bio-inspired baffle structures and acoustic processing techniques in man-made systems can reduce sonar array aperture requirements by a factor of 100 or more for a variety of both aerial and underwater acoustic sensing applications. Recent modeling results are presented along with progress toward the design of a compact bio-inspired sonar system for high-resolution imaging. [Work supported by ONR and NUWC Newport.]

2:15

2pEA3. Front-looking and side-looking receiving beams for biosonar imaging and flight guidance. James A. Simmons (Dept. of Neurosci., Brown Univ., Providence, RI 02912, james_simmons@brown.edu)

The biosonar broadcasts of big brown bats are very broadly beamed over $\pm 60°$-$120°$, thus ensonifying virtually all of the objects in the surrounding sonar scene. However, masking-release results indicate that off-side clutter is rejected outside of a significantly narrower beam of $\pm 15°$. Recent experiments establish that clutter interference with target imaging is prevented by sensing the lower amplitude of FM2 relative to FM1 in clutter echoes and forming poorly-focused images of the clutter. Removal of clutter from perception does not address the need to guide flight in complex surroundings bounded by clutter such as vegetation. A separate mechanism used for guidance may be located in the midbrain, where exclusively contralaterally sensitive neurons respond to target range and direction, the ingredients for following the flow of surrounding objects as they slide past the flying bat. A review of neurophysiological results from different laboratories suggests that the big brown bat's biosonar system may contain different receiver pathways—a forward-looking system for target imaging and classification, and a side-looking system for following the surrounding clutter. [Work supported by ONR and NSF.]

2:35

2pEA4. Improving direction-sensing by multibeam sonar. Gerard Llort-Pujol (Image and Information Processing Dept., Institut Mines Télécom - Télécom Bretagne, Brest, France), Kenneth G. Foote (Woods Hole Oceanographic Inst., 98 Water St., Woods Hole, MA 02543, kfoote@whoi.edu), and Christophe Sintes (Image and Information Processing Dept., Institut Mines Télécom - Télécom Bretagne, Brest, France)

The complexity of multibeam sonar systems makes their beamforming susceptible to amplitude and phase distortion, e.g., due to environmental changes. A form of calibration is usually performed *in situ* over a flat area to ensure flatness in the resulting sonar image. However, lack of detailed knowledge of individual channel performance prevents application of an amplitude-weighting function such as the Blackman or Hamming type, which could otherwise be used to reduce the impact of sidelobes without damaging signal quality. Two radically different solutions are proposed: application of the Vernier principle [G. Llort-Pujol, Oceans 2006 MTS/IEEE Conf. Proc. (Quebec City, Canada, 2008); C. Sintes *et al*., Oceans 2011 MTS/IEEE Conf. Proc. (Waikoloa, HI, 2011)] to interferometry performed at low grazing angles, and performance of a standard-target calibration [K. G. Foote *et al*., J. Acoust. Soc. Am. **117**, 2013 (2005)] to measure the two-way sensitivity of individual channels directly. The two methods, which are also applicable to sidescan sonar, are elaborated.

Contributed Paper

2:55

2pEA5. Bearing measurements using a compact wideband sensor by forming three dipoles. David A. Brown (BTech Acoustics LLC, ATMC, 151 Martine St., Fall River, MA 02723, dbAcoustics@cox.net)

This paper summaries the development of a compact sensor for determining the bearing angle of an incoming signal using trinary acoustic dipoles. The approach is a kin to the using orthogonal dipoles an omni-reference, which is common in DIFAR sonobuoy. Symmetrically processing three dipoles can offer advantages in sensitivity and extended bandwidth. Analytical and experimental results on the TRI-phase Bearing Estimator, or TRIBE, are presented.

Session 2pED

Education in Acoustics: Take 5's

Jack Dostal, Chair
Phys., Wake Forest Univ., P.O. Box 7507, Winston-Salem, NC 27109

For a Take-Five session no abstract is required. We invite you to bring your favorite acoustics teaching ideas. Choose from the following: short demonstrations, teaching devices, or videos. The intent is to share teaching ideas with your colleagues. If possible, bring a brief, descriptive handout with enough copies for distribution. Spontaneous inspirations are also welcome. You sign up at the door for a five-minute slot before the session starts. If you have more than one demo, sign-up for two consecutive slots.

TUESDAY AFTERNOON, 3 DECEMBER 2013 CONTINENTAL 1, 1:00 P.M. TO 3:50 P.M.

Session 2pMU

Musical Acoustics and Signal Processing in Acoustics: Digital Musical Instruments

Edgar J. Berdahl, Chair
Music, Louisiana State Univ., 102 New Music Bldg., Baton Rouge, LA 70803

Invited Papers

1:00

2pMU1. Interaction design and the active experience of music. David Wessel (Music CNMAT, Univ. of California Berkeley, 1750 Arch St., Berkeley, CA 94709, davidwessel@me.com)

Music search engines, play list generators, streaming audio, and portable players have taken much of the focus of music technology. The emphasis is on delivery and experiencing music is by playback, playback while jogging or while working about the house, and sadly even while studying. In this talk in the hope of providing an antidote I will examine the role of bodily action in the experience of music and the importance of human computer interaction design in the development of computationally based musical instruments. Central are gestural interfaces and their mapping to musical material. Special emphasis will be given to designing for expression, for exploration and discovery, and to a musical practice that involves a coordinated balance of software development and daily bodily engagement with one's instrument.

1:20

2pMU2. LinnStrument and other new expressive musical controllers. Roger Linn (Roger Linn Design, 1147 Keith Ave., Berkeley, CA 94708, rl@rogerlinndesign.com)

Roger Linn will demonstrate his LinnStrument, a controller for musical performance that captures three dimensions of movement for each touch, polyphonically, in order to provide more expressive control of software music synthesis. In addition, he will show videos of other similar new instruments and compare the unique approaches taken by each designer.

1:40

2pMU3. Sound synthesis for a brain stethoscope. Chris Chafe, Juan-Pablo Caceres, and Michael Iorga (Stanford Univ., CCRMA/Music, Stanford, CA 94305, cc@ccrma.stanford.edu)

Exploratory ascultation of brain signals has been prototyped in a project involving neurologists, real-time EEG and techniques for computer-based sound synthesis. In a manner similar to using a stethoscope, the listener can manipulate the location being listened to. Sounds which are heard are sonifications of electrode signals. We present a method for exploring sounds from arrays of sensors as sounds which are useful for distinguishing brain states. The approach maps brain wave signals to modulations characteristic of human voice. Computer-synthesized voices "sing" the dynamics of wakefulness, sleep, seizures, and other states. The goal of the project is to create a recognizable inventory of such vocal "performances" and allow the user to probe source locations in the sensor array in real time.

2:00

2pMU4. Grafting acoustic instruments and signal processing: Creative control and augmented expressivity. Dan Overholt (Media Technol., Aalborg Univ. Copenhagen, A.C. Meyers Vaenge 15, Copenhagen SV. 2450, Denmark, dano@create.aau.dk)

In this study, work is presented on a hybrid acoustic/electric violin. The instrument has embedded processing that provides real-time simulation of acoustic body models using DSP techniques able to gradually transform a given body model into another, including extrapolations beyond the models to explore interesting new timbres. Models can include everything from various violin bodies to guitars, sitars with their sympathetic strings, and even physically impossible acoustic bodies. The development also presents several practical approaches to sensor augmentation and gestural playing techniques that can be applied to bowed-string and other acoustic instruments, in order to provide immediate creative control over the possibilities offered by DSP. The study has focused on augmenting the expressivity of the violin toward finding novel timbral possibilities, rather than a goal of simulating prior acoustic violins with high fidelity. The opportunity to control a virtually malleable body while playing, i.e., a model that changes reverberant resonances in response to player input, results in interesting audio effects. Other common audio effects can also be employed and simultaneously controlled via the musician's movements. For example, gestural tilting of the instrument is tracked via an embedded Inertial Measurement Unit (IMU), which can be assigned to alter parameters such as the wet/dry mix of an octave-doubler or other effect, further augmenting the expressivity of the player.

2:20

2pMU5. Saxophone fingering identification. Tamara Smyth (Music, Univ. of California San Diego, 9500 Gilman Dr., MC 0099, La Jolla, CA 92093-0099, trsmyth@ucsd.edu) and Marjan Rouhipour (Computing Sci., Simon Fraser Univ., Surrey, BC, Canada)

The focus of this work is to identify the tonehole configuration or "fingering" applied by a player during performance, using only the signal recorded at the bell. Because of a player can use alternate fingerings/overblowing to produce a given frequency, detecting the sounding pitch only reduces the possible candidates—it does not produce a unique result. Several recordings of a professional saxophonist playing notes using all fingerings are considered, and several higher level features are explored for distinguishing between a fundamental and an overblown note. In the latter case, it is observed that during the attack portion of the note, the spectral centroid is usually lower, there is greater inharmonicity and increased pitch instability. Combining these heuristics with the detection of subharmonics has yielded excellent results in detecting overblown notes. With the possible fingerings being greatly reduced by this preprocessing, more computationally expensive statistical methods may be employed for a more accurate estimation of the actual fingering applied. To this end, the recorded sound is calibrated to that produced by a reed model coupled to a waveguide that is informed by an acoustic measurement of the player's saxophone configured with each usable fingering.

2:40

2pMU6. The Faust Synthesis Toolkit: A set of linear and nonlinear physical models for the Faust programming language. Romain Michon (Dept. of Music, Ctr. for Comput. Res. in Music and Acoust., 660 Lomita Court, Stanford, CA 94305-8180, rmichon@ccrma.stanford.edu)

The Faust Synthesis ToolKit is a set of virtual musical instruments written in the Faust programming language and based on waveguide algorithms and on modal synthesis. Most of them were inspired by instruments implemented in the Synthesis ToolKit (STK) and the program SynthBuilder. Our attention has partly been focused on the pedagogical aspect of the implemented objects. Indeed, we tried to make the Faust code of each object as optimized and as expressive as possible. Some of the instruments in the Faust-STK use nonlinear allpass filters to create interesting and new behaviors. Also, a few of them were modified in order to use gesture data to control the performance. A demonstration of this kind of use is done in the Pure Data program. Finally, the results of some performance tests of the generated C++ code are presented.

3:00

2pMU7. Drawn to sound: An audio visual musical instrument using custom electronics and magnetometer. John Granzow and Hongchan Choi (CCRMA, Stanford Univ., 660 Lomita Dr., Stanford, CA 94305, johknee5@gmail.com)

Drawings are amplified through a resonant surface and transmitted via microphone to custom software hosted on an embedded linux computer. An HMC-5883L magnetometer is used to modulate the signal according to the position of the pencil (equipped with magnetic sleeve). 3D vectors are derived from the magnetometer using a custom Arduino library. We project this vector into a 2D plane to get magnitude or distance between the sensor and the magnetized pencil as well as the heading angle. This implementation gives us 2D polar coordinates such that the position of the pencil can be used to vary the audio output. The transformation of the raw drawing sound is excited with proximity to the magnetometer due to the sensors exponentially increasing sensitivity to the magnetic field. Resulting drawings often contain both visually motivated marks as well as gestures that are made for sound such as dark scribbled regions where a desired timbre or pitch shift is repeated throughout a performance. This presentation will discuss hardware design, the implementation of our custom software and circuitry, how these components combine for a compelling performance platform as well as areas where we seek improvement.

3:20

2pMU8. The Stingray embedded acoustic instrument. Edgar J. Berdahl (Music, Louisiana State Univ., 102 New Music Bldg., Baton Rouge, LA 70803, eberdahl@ccrma.stanford.edu)

Many traditional acoustic musical instruments are convenient to use: a performer picks one up, provides an energetic excitation, and the resulting sound radiates immediately. No wires, protocols, or software updates are ever required. The same cannot be said for the vast majority of prior digital musical instruments. The present work addresses how to endow a digital musical instrument with the convenience, look, feel, and personality of a traditional acoustic musical instrument via enclosure prototyping techniques, audio amplification, and embedded computation. For example, the Stingray is an embedded acoustic instrument. Although its battery needs to be occasionally charged, it otherwise can give the impression of a traditional acoustic musical instrument. The control inputs for the Stingray include a piano keyboard and force-feedback motorized faders. The faders allow the performer to interact expressively with the sound, while the piano keyboard enables the precise selection of notes. If desired, the Stingray can be programmed using physical models, including models of hypothetical acoustic instruments that would be infeasible to build physically. In this configuration, the Stingray expressively transforms the performer's gestures into radiated sound in an energy-conserving manner.

3:35

2pMU9. Towards a bendable circuit model of the Casio SK-1 keyboard. Kurt J. Werner (CCRMA, Stanford Univ., 223 Ayrshire Farm Ln., Apt. 205, Stanford, CA 94305, kwerner@ccrma.stanford.edu)

The Casio SK-1 keyboard was introduced in 1985 and synthesizes sampled and built-in sounds via pulse-code modulation and additive ("harmonic") synthesis. Initially important as one of the first home keyboards with sampling capabilities, the SK-1 has become one of the most popular instruments for circuit bending, the process of creatively modifying or augmenting sound-producing electronic devices. I create a parameterized component-level software model of the analog circuitry of the Casio SK-1, with applications to archiving and preserving its historic sound, expanding its basic behavior through circuit-bent modifications and extensions, and providing circuit benders with a resource for informed bending. Throughout, special attention is paid to creating models in terms of the circuit's component values. The SK-1's Percussion, Bass, and Chord Filters are modeled and analyzed in continuous-time as transfer functions and discretized via the bilinear transform. The non-linear (including diodes and transistors) Envelope/Pitch Mixing Circuit and Melody Filter are analyzed with linearizing simplifications to elucidate design intent, and modeled as ordinary differential equations to capture their behavior accurately. A review of techniques for the numerical solution of ordinary differential equations follows. A model of the internal speaker's impulse response and estimated static non-linearity rounds out the project.

Session 2pNS

Noise, Physical Acoustics, and Structural Acoustics and Vibration: Launch Vehicle Acoustics

Kent L. Gee, Cochair
Brigham Young Univ., N243 ESC, Provo, UT 84602

Tracianne B. Neilsen, Cochair
Brigham Young Univ., N311 ESC, Provo, UT 84602

R. Jeremy Kenny, Cochair
NASA, M.S. ER42, Bldg. 4203, Marshall Space Flight Ctr., Huntsville, AL 35812

Invited Papers

1:00

2pNS1. Far-field acoustic modeling of rocket noise to determine community impacts. Michael M. James, Alexandria R. Salton, and Micah Downing (Blue Ridge Research and Consulting, LLC, 15 W. Walnut St., Ste. C, ASHEVILLE, NC 28801, michael.james@blueridgeresearch.com)

The emerging commercial space market is generating interest in commercial launch site ("spaceport") development around the United States. FAA regulations require all new spaceports to acquire a launch site operator's license, which is considered a Federal action subject to environmental review. Potential noise impacts are evaluated based on FAA Order 1050.1E, Change 1, Environmental Impacts: Policies and procedures, which include the assessment of DNL and may be supplemented with additional acoustical metrics. These supplemental metrics may range from speech interference to structural damage impacts. Extensive studies and research have examined the appropriateness of these metrics in relation to aircraft operations. However, the differences between these acoustic sources and operational modes stress the need for computer models and impact criteria specific to launch vehicles. Further measurements and research are needed to improve rocket source characterization, long-range sound propagation of high amplitude waveforms through complex atmosphere, and environmental and community impacts. The evolving nature of the regulatory environment surrounding rocket noise warrants a renewed focus on appropriate noise modeling and impact criteria to determine potential conflicts with launch noise.

1:20

2pNS2. Use of a large microphone array to identify noise sources during a rocket engine test firing and a rocket launch. Jayanta Panda (Experimental Aero-Phys. Branch, NASA Ames Res. Ctr., M.S. 260-1, Moffet Field, CA 94035, jayanta.panda-1@nasa.gov), Robert N. Mosher, and Barry J. Porter (Experimental Aero-Phys. Branch, Aerosp. Computing, Inc., Mountain View, CA)

A 70 microphone, 10 ft$\times$10 ft, microphone phased array was built for use in the harsh environment of rocket launches. The array was setup at NASA Wallops launch pad 0A during a static test firing of Orbital Sciences' Antares engines, and again during the first launch of Antares vehicle. It was placed 400 ft away from the pad, and was hoisted on a scissor lift 40 ft above ground. The data sets provided unprecedented insights into rocket noise sources. The duct exit was found to be the primary source during the static test firing; the large amount of water injected beneath the nozzle exit quenched all other sources. The noise maps during launch were found to be time-dependent. As the engines came to full power and became louder, the primary source switched from the duct inlet to the duct exit. Further elevation of the vehicle caused spilling of the hot plume, resulting in a distributed noise map covering most of the pad. As the entire plume emerged from the duct, and the on-deck water system came to full power, the plume itself became the loudest noise source. These noise maps will help to improve the sound suppression system for future launches.

1:40

2pNS3. Acoustic measurements of the Epsilon rocket at liftoff. Tatsuya Ishii, Seiji Tsutsumi, Kyoichi Ui, Hideshi Oinuma (JAXA, 7-44-1 Jindaiji-higashi-machi, Chofu, Tokyo 182-8522, Japan, ishii.tatsuya@jaxa.jp), Yutaka Ishii (Bruel & Kjaer Japan, Tokyo, Japan), and Kei Wada (Science Service, Inc., Tokyo, Japan)

Launch vehicles generate intense acoustic field caused by the enormous thrust power during liftoff, and this acoustic field leads to harmful payload vibration. Japan Aerospace Exploration Agency (JAXA) plans to launch a new solid propellant rocket, Epsilon, in 2013. This three-stage rocket utilizes a reliable first stage motor of the JAXA's H-2 rocket, SRB-A. Since the SRB-A is expected to cause excessive acoustic load, a countermeasure was required to mitigate the acoustic feedback to the vehicle. Computational works proposed a launch pad structure to attenuate the Mach wave radiation from the plume, and the acoustic wave generated by the plume impinging to the flame deflector. In the previous ASA conference, the authors introduced scale model tests using 1:42 scale rocket motors and the launch pad models. The scale model tests clarified the acoustic benefit of the launch pad structure. The tested geometry of the launch pad structure was adopted to the full-scale structure. Acoustic measurements are planned in the first launch of the Epsilon rocket

in order to evaluate the acoustic influence with this newly constructed launch pad structure. The measurement setup and a brief review of the measurements (if possible) are discussed.

2:00

2pNS4. Scale model thruster acoustic measurement results. Magda B. Vargas and Robert J. Kenny (MSFC, NASA, M.S. ER42, Bldg. 4203, Marshall Space Flight Ctr., Huntsville, AL 35812, magda.b.vargas@nasa.gov)

The Space Launch System (SLS) Scale Model Acoustic Test (SMAT) is a 5% scale representation of the SLS vehicle, mobile launcher, tower, and launch pad trench. The SLS launch propulsion system will be comprised of the Rocket Assisted Take-Off (RATO) motors representing the solid boosters and four Gas Hydrogen (GH2) thrusters representing the core engines. The GH2 thrusters were tested in a horizontal configuration in order to characterize their performance. In phase 1, a single thruster was fired to determine the engine performance parameters necessary for scaling a single engine. A cluster configuration, consisting of the four thrusters, was tested in phase 2 to integrate the system and determine their combined performance. Acoustic and overpressure data was collected during both test phases in order to characterize the system's acoustic performance. The results from the single thruster and 4-thuster system are discussed and compared.

2:20

2pNS5. Near-field/far-field study of the end-effects regime produced by large area ratio nozzles. Raymundo M. Rojo, Charles E. Tinney, Woutijn J. Baars (Aerosp., Univ. of Texas at Austin, 701 28th St., Apt. 407, Austin, TX 78712, raymundo.rojo46@gmail.com), and Joseph H. Ruf (NASA Marshall Space Flight Cente, Huntsville, AL)

Vibro-acoustic loads emanating from large area ratio rocket nozzles during start-up can be catastrophic to the launch system and payload. This study quantifies a particular feature referred to as the "end-effects regime", which is considered the largest source of vibro-acoustic loading during start-up [Nave and Coffey, AIAA Paper 1973-1284]. In this experiment, data acquired during the start-up sequence of several full-scale rocket engines are compared to the laboratory-scale measurements of a thrust-optimized parabolic-contour nozzle conducted in a fully anechoic chamber. The laboratory studies encompass both static and dynamic wall pressures measured inside the nozzle, as well as far-field acoustic surveys. The event produced during the "end-effects regime" was successfully reproduced in the sub-scale model, and was characterized in terms of its mean, variance, and skewness, as well as the spectral properties of the signal obtained by way of time-frequency analyses. The intensity and characteristic frequency of the event of interest are discussed through a comparison of the nominal values for the full-scale and sub-scale system and whether they obey with standard scaling laws.

2:40

2pNS6. Evaluation of Japanese current primary launch vehicle liftoff acoustic environment change due to launch pad facility modifications. Hiroki Ashida, Makoto Hirai (Aerosp. Systems, Mitsubishi Heavy Industries, Ltd., 10, Oye-cho, Minato-ku, Nagoya City, Aichi 455-8515, Japan, hiroki1_ashida@mhi.co.jp), Keita Terashima, and Takumi Ujino (Space Transportation Mission Directorate, Japan Aerosp. Exploration Agency, Ibaraki, Japan)

H-IIA, Japanese primary launch vehicle, has been successfully launched 21 flights with a success rate of 95.4%. During 12 years of its operational phase, extensive acoustic measurements on the vehicle and on the ground have been conducted to refine conventional prediction methods and to evaluate the effect of vehicle/pad configuration changes. In this presentation, the evaluation of the effects of major pad configuration modification on H-IIA liftoff acoustic environment is presented. The effect of water injection is also evaluated.

3:00–3:15 Break

3:15

2pNS7. Numerical study on acoustic generation of a supersonic jet impinging to deflectors. Seiji Tsutsumi, Ryoji Takaki (JEDI, JAXA, 3-1-1 Yoshinodai, Chuuou, Sagamihara, Kanagawa 252-5210, Japan, tsutsumi.seiji@jaxa.jp), Koji Okamoto (Dept. of Adv. Energy, Univ. of Tokyo, Kashiwa, Japan), and Susumu Teramoto (Dept. of Aeronautics and Astronautics, Univ. of Tokyo, Bunkyo-ku, Japan)

Acoustic wave generated from a $M = 1.8$ ideally expanded jet impinging on flame deflectors is investigated numerically in order to obtain the knowledge of flame deflector design to mitigate the acoustic loading on launch systems and payloads. Mechanism of acoustic wave generated from a 45-degree-inclined flat plate placed 5D downstream from the nozzle exit is analyzed first, and it is revealed that source of the acoustic wave is located where shock waves is formed due to the jet impingement. In this study, effect of the deflector shape will be discussed for better understanding the acoustic generation mechanism.

3:35

2pNS8. Towards jet acoustic prediction within the Launch Ascent and Vehicle Aerodynamics framework. Jeffrey A. Housman (Appl. Modeling and Simulation Branch, NASA Ames Res. Ctr., M.S. N-258, Moffett Field, CA 94035, jeffrey.a.housman@nasa.gov), Christoph Brehm (Appl. Modeling and Simulation Branch, Sci. and Technol. Corp., Moffett Field, CA), and Cetin Kiris (Appl. Modeling and Simulation Branch, NASA Ames Res. Ctr., Moffett Field, CA)

Understanding the acoustic environment generated during lift-off is critical for successfully designing new space vehicles. In order for modeling and simulation tools to effectively assist in the development of the vehicles, validation must be performed on simplified model problems. In this paper, time-accurate implicit large eddy and detached eddy simulations coupled with a linear acoustic propagation method are applied to a Mach 1.8 perfectly expanded jet impinging on a flat plate at 45 degrees. The Launch Ascent and Vehicle Aerodynamics (LAVA) code used to simulate this problem is a high-fidelity unsteady simulation tool for modeling fluid dynamics, conjugate heat transfer, and acoustics. A detailed description of the linear acoustic propagation tool is provided. The narrow band far-field

sound pressure levels predicted using LAVA are compared to existing experimental data. POD and spectral methods are applied to analyze the noise sources due to coherent flow structures and jet impingement. Grid and time-step sensitivity studies are performed to assess the spatial and temporal requirements for accurate jet acoustic simulation. Sensitivity of the predicted far-field sound pressure levels to position of the acoustic propagation surface is also assessed.

3:55

2pNS9. Intensity-based approach to characterize near-field acoustic environments of space flight vehicles. Michael M. James, Alexandria R. Salton (Blue Ridge Research and Consulting, LLC, 15 W. Walnut St., Ste. C, Asheville, NC 28801, michael.james@blueridgeresearch.com), Kent L. Gee, and Tracianne B. Neilsen (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT)

An intensity-based measurement probe has been developed to measure the magnitude, directivity, and spectral content of near-field rocket source noise. An array of the intensity-based measurement probes was deployed in a static test firing of the GEM-60 at Alliant Techsystems in Promontory, UT, in 2012. The probes were positioned along the shear layer of the rocket motor plume to enable a comparison of the resultant source characterization obtained via sound power type measurement approaches to traditional acoustic measurement methodologies. The measurement results demonstrate that the intensity-based probes advance the measurement and characterization of the near-field acoustic environment of rockets. Moreover, intensity-based acoustic data provides an important role in formulating more realistic sound source models, improving acoustic load estimations, and aiding in the development of the next-generation space flight vehicles via improved measurements of noise near the rocket plume.

4:15

2pNS10. Methods for estimating acoustic intensity in rocket noise fields. Derek C. Thomas, Benjamin Y. Christensen, and Kent L. Gee (Dept. of Phys. and Astronomy, Brigham Young Univ., N223 ESC, Provo, UT 84602, derekcthomas@gmail.com)

The acoustic field produced by launch vehicles is difficult to measure and characterize. Acoustic intensity measurements provide more information per measurement location than pressure measurements and are therefore interesting for the characterization of rocket noise fields. The extreme environment associated with a rocket requires a robust intensity probe while the large size of the source and the high-amplitude, highly nonlinear behavior of the system produce a signal with significant low and high frequency components. Thus, the probe must also provide accurate results over a large frequency band. The bandwidth limitations of the standard method for estimating acoustic intensity, the p-p finite difference method, motivated the development of an alternative method for intensity estimation. The new phase and amplitude gradient estimation (PAGE) method will be presented and compared to the standard p-p method. Specific features of the rocket noise field that can be leveraged to improve the bandwidth of the intensity estimates will also be discussed.

4:35

2pNS11. Acoustic intensity estimates from a solid rocket motor test firing. Benjamin Christensen, Derek C. Thomas, and Kent L. Gee (Phys. and Astronomy, Brigham Young Univ., 539E 300N, Provo, UT 84606, ukeben@gmail.com)

Acoustic measurements of a static GEM-60 solid motor test firing were taken as part of a continuing effort to characterize the aeroacoustic source regions and noise environment around launch vehicles. Multiple 2D intensity probes, consisting of four coplanar microphones, were used in the measurement. Two intensity estimation techniques have been applied to the data: the finite difference p-p method, and the new phase and amplitude gradient estimation (PAGE) method. We will present and compare results from both methods and compare to measurements made at past test firings. It appears that the PAGE method for estimating acoustic intensity provides usable results over a larger frequency bandwidth.

Session 2pPA

Physical Acoustics and Structural Acoustics and Vibration: Acoustics of Pile Driving: Measurements, Models, and Mitigation II

Kevin M. Lee, Cochair
Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758

Karl-Heinz Elmer, Cochair
OffNoise-Solutions GmbH, Leinstr. 36, Neustadt a. Rbge. 31535, Germany

Invited Papers

1:30

2pPA1. Effects of pile driving on fishes. Arthur N. Popper (Univ. of Maryland, Biology/Psych. Bldg, College Park, MD 20742, apopper@umd.edu), Michelle B. Halvorsen (Battelle-Pacific Northwest National Lab., Sequim, WA), Thomas J. Carlson (ProBioSound, Holmes Beach, FL), Michael E. Smith (Univ. of Western Kentucky, Bowling Green, KY), and Brandon M. Casper (Naval Submarine Medical Res. Lab., Groton, CT)

We examined the physiological effects of impact pile driving on fishes using a specially designed tube that allows replication of the far field acoustic conditions of impulsive stimuli. Studies show that the received signal levels needed to result in onset of effects is a combination of single strike sound exposure level (SELss) and cumulative sound exposure level (SELcum), although not in an equal energy relationship. In contrast to current interim regulations, which indicate that the onset of physiological effects occurs at 187 dB SELcum, our experimental results for six species of fishes showed that the onset of physiological effects, none of which produced mortality, was at about 207 dB SELcum. This onset SELcum had to be at least 7–10 dB higher to result in effects that could potentially be mortal. Additional studies showed that fishes can recover from the effects of pile driving and that a fish species without a swim bladder showed no effects, at least up to an SELcum of 216 dB. Investigations on the effects of pile driving on sensory hair cells of the inner ear, an analog for hearing loss, showed that damage only occurred at SELcum that are substantially higher than onset of other effects.

1:50

2pPA2. A Monte Carlo approach to determining marine mammal exposure risk to long term marine piling operations. Paul A. Lepper (School of Electron., Elec. and Systems Eng., Loughborough Univ., Loughborough LE113TU, United Kingdom, p.a.lepper@lboro.ac.uk), Stephen P. Robinson, Pete D. Theobald, and Tanja Pangerc (Acoust. Group, National Physical Lab., Teddington, United Kingdom)

The expansion of offshore renewable developments, primarily offshore wind, has led to widespread use of large scale percussive piling for foundation construction. In UK waters alone, over 900 foundations installed up to 2013, mostly mono-piles with extensive upscaling of developments planned for the next decade. Pile diameters range from a few meters to greater than 6 m and penetration depths of 20–30 m. These piles are typically percussively driven with several thousand hammer strikes over periods of several hours with individual strike hammer energies in excess of 1900 kJ occasionally used and reported per strike underwater Sound Exposure Level source levels of 215 dB re 1 $\mu Pa^2 s\text{-}m^2$. Potential exists for injury to occur from cumulative sound exposure to repetitive but lower level signals at greater range. If simple receptor behaviors are assumed (static, fleeing, etc.) exposure over time to an entire pile construction sequence can be estimated. These models have been extended using a Monte Carlo approach to model long term, entire wind farm construction scenarios with repetitive foundation construction periods of 24–36 h. The statistical distribution of exposure risk is modeled as well as analysis of the sensitivity of behavioral responses to potential impact effects such as habitat exclusion.

2:10

2pPA3. Noise mitigation systems (NMS) for reducing pile driving noise: Experiences with the “big bubble curtain” relating to noise reduction. Michael A. Bellmann and Patrick Remmers (itap GmbH, Marie-Curie-Str. 8, Oldenburg 26160, Germany, bellmann@itap.de)

For the offshore wind farm Borkum West II in the German North Sea the Noise Mitigation System (NMS) “Big Bubble Curtain” was used during pile driving activities. Within this project systematically variations of different influencing factors on noise reductions such as air volume, nozzle hose sizes, distance of nozzle hoses, etc., were investigated. Additionally the “Big Bubble Curtain” is currently in use for different other OWF in the German North Sea. Therefore, the “Big Bubble Curtain—BBC” is at the moment one of the most investigated NMS under offshore condition. Within this presentation, experiences and results of the above listed projects will be shown and discussed.

2:30

2pPA4. Evaluation of hydro sound and vibration measurements during the use of the Hydro-Sound-Damper (HSD) at the wind farm "London Array". Benedikt Bruns (Technische Universität Braunschweig, Beethovenstrasse 51 b, Braunschweig 38106, Germany, b.bruns@tu-bs.de)

Since some years a noise prevention concept for the protection of marine animals exists in Germany. Based on that, the acoustic underwater noise from the pile driving at offshore wind farms is required to be less than 160 dB (SEL) at a distance of 750 m. This value, however, is often exceeded so that the use of a soundproofing system is necessary. The Hydro-Sound-Damper (HSD) is a new, versatile method to reduce the noise during offshore pile driving. To achieve this, elements of different sizes and materials are used, which are fixed to fishing nets. The principle of operation and the effectiveness of these HSD elements were investigated in the laboratory and in situ under offshore conditions at the world's largest offshore wind farm "London Array." During the offshore tests thorough measurements were performed which metered the propagation of the hydro sound and the vibrations of the sea floor at various distances and directions. The evaluation of these data led to very promising results concerning underwater noise reduction. This article describes the theory and implementation of the HSD at "London Array" and focuses on the interpretation of the data from the hydro sound and vibration measurements.

Contributed Papers

2:50

2pPA5. Efficient application of encapsulated bubbles and foam elements to mitigate offshore piling noise. Karl-Heinz Elmer (OffNoise-Solutions GmbH, Leinstr. 36, Neustadt a. Rbge. 31535, Germany, karl-heinz.elmer@t-online.de)

The very high noise levels of offshore piling noise during the installation of offshore wind converters are dangerous to marine life like harbor porpoises and fishes. Encapsulated bubbles and foam elements are successfully used to reduce the very high noise levels. There are different physical effects such as impedance mismatch, resonance and scattering effects and material damping effects that are responsible to the measured noise reduction between about 10 dB (SEL) and 23 dB (SEL). The radiation of underwater noise from the pile surface, the propagation and the reflections of the radiated waves are investigated together with the influence of encapsulated bubbles and foam elements in the near field of the pile. Only a very small part of the impact energy is radiated directly from the pile into the surrounding water. Most of the hammer energy is driven into the ground. A small part from this is radiated indirectly from the ground into the water. The efficiency of encapsulated bubbles and foam elements in underwater noise mitigation is investigated by theoretical studies, laboratory measurements, and by offshore measurements with different elements and combinations.

3:05

2pPA6. Dependence of resonance frequencies and attenuation for large encapsulated bubbles on bubble wall thickness and bubble fill-material. Gregory Enenstein, Preston S. Wilson, and Kevin M. Lee (Mech. Eng. Dept. and Appl. Res. Lab., The Univ. of Texas at Austin, 4700 W Guadalupe St #A-437, Austin, TX 78751, gregenenstein@gmail.com)

Arrays of large encapsulated bubbles are currently under development for the purpose abating low-frequency anthropogenic underwater noise from various sources including marine pile driving and oil and gas exploration and production. An existing predictive model [Church, J. Acoust. Soc. Am. **97**, 1510–1521 (1995)], which was originally intended to describe propagation through suspensions of microbubbles used as ultrasound contrast agents, was previously found to be in good agreement with resonance frequency and attenuation measurements using large encapsulated with radii on the order of 10 cm [Lee *et al*., J. Acoust. Soc. Am **132**, 2039 (2012); Lee and Wilson, Proceedings of Meetings on Acoustics **19**, 075048 (2013)]. For the current study, both laboratory and lake experiments were performed on large encapsulated bubbles to investigate the dependence of the bubbles' resonance frequencies and attenuation on the bubble wall thickness. Additionally, laboratory measurements were made to investigate the effects on encapsulated bubble resonance frequencies and damping using bubble fill-materials other than air, and a lake experiment was then performed to relate these effects on the damping to the attenuation provided by arrays of such bubbles. [Work supported by AdBm Technologies.]

3:20–3:35 Break

3:35

2pPA7. Acoustic measurements and modeling of air-filled, underwater resonator cavities. Laura Tseng, Kevin M. Lee (Appl. Res. Laboratories: The Univ. of Texas at Austin, 10,000 Burnet Rd., Austin, TX 78758, Ltseng@utexas.edu), Preston S. Wilson (Mech. Eng., The Univ. of Texas at Austin, Austin, TX), and Mark S. Wochner (AdBm Technologies, Austin, TX)

This paper investigates the near-resonance acoustical properties of submerged air-filled resonators intended for use in an underwater noise abatement system. These resonators are a potential alternative to encapsulated bubbles. The resonators are similar to Helmholtz resonators in shape and design, but without a neck, consisting of underwater inverted air-filled cavities with rigid walls. A finite element model was developed to investigate the acoustic behavior of the resonators near their resonance frequencies, and based on the results of the model, physical realizations of the resonators were designed and fabricated for testing. Experiments were performed with the resonators in a closed water-filled tank operated in the long wavelength limit [J. Acoust. Soc. Am. **132**, 2039 (2012)], where their resonance frequencies and Q-factors were measured using the technique described by Leighton *et al*., [J. Acoust. Soc. Am. **112**, 1366–1376 (2002)]. Comparison between the results from the measurements and modeling will be discussed. [Work supported by Shell Global Solutions.]

3:50

2pPA8. Radiated sound from a scale-model pile submerged in a two--layer medium. Kevin M. Lee, Todd A. Hay, Taylor W. Weaver, Preston S. Wilson, and Mark F. Hamilton (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, klee@arlut.utexas.edu)

Underwater noise due to both marine pile driving and offshore wind farm operation is not only radiated directly from the pile into the water, but also from the seabed surrounding the pile. While there is much interest in mitigating the noise from these activities, a better understanding of the source mechanisms and propagation is needed to determine optimal strategies for noise abatement. A recent analytical model of the acoustic field radiated by submerged piles includes radiation from the pile directly into the water and into a stratified viscoelastic sediment as well as propagation into a shallow water waveguide from both the direct and sediment radiation paths [Hay *et al*., Proceedings of Meetings on Acoustics **19,** 070038 (2013)]. As a step towards validating this model, scale-model experiments were conducted in the high kilohertz frequency range with a model pile consisting of a mechanically excited metallic tube inserted into a laboratory tank filled with two stratified layers to simulate the water/sediment interface. Measurements of the acoustic field in the experiment are compared with the model predictions, and the relevance of these results to implementing noise abatement strategies will be discussed. [Work supported by ARL:UT IR&D.]

4:05

2pPA9. On the resonance frequency of an ideal arbitrarily shaped bubble. Kyle S. Spratt, Kevin M. Lee (Appl. Res. Labs., The Univ. of Texas at Austin, 4307 Caswell Ave., Apt. E, Austin, TX 78751, sprattkyle@gmail.com), Preston S. Wilson (Dept. of Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), and Mark S. Wochner (AdBm Technologies, Austin, TX)

Large encapsulated bubbles have recently been described for use in abating low-frequency anthropogenic underwater noise [J. Acoust. Soc. Am. **130**, 3325–3332 (2011)], and the use of encapsulation allows for the possibility of bubbles that are nonspherical in their equilibrium state. For the purpose of more accurately determining such bubbles' resonance frequencies, a lumped-element model of the linear oscillation of an ideal, arbitrarily shaped gas bubble in an incompressible liquid is presented. The corresponding boundary-value problem required to predict the resonance frequency of the bubble is seen to be equivalent to a classic problem in electrostatics [J. Acoust. Soc. Am. **25**, 536–537 (1953)]. Predictions made for the resonance frequency of prolate and oblate spheroidal bubbles using this model are tested against a finite-element model of the full acoustic scattering problem. Particular attention is then paid to the case of an ideal toroidal bubble of arbitrary thickness, and predictions made for the resonance frequency of such a bubble using the lumped-element approach are compared to a finite-element model of the full acoustic scattering problem as well as to existing approximate models for the dynamics of thin toroidal bubbles. [Work supported by AdBm Technologies, LLC and the ARL:UT McKinney Fellowship in Acoustics.]

4:20

2pPA10. The effect of sediment stiffness on the piling pulse—Results from a wave-equation analysis model. Michael A. Wood and Victor F. Humphrey (Fluid Dynam.. and Acoust. Group, Inst. of Sound and Vib. Res., Faculty of Eng. and the Environment, Univ. of Southampton, Southampton, Hampshire SO17 1BJ, United Kingdom, maw1v07@soton.ac.uk)

The rapid expansion of wind farms in UK waters has led to increased concern of the anthropogenic noise emitted into the sea due to piling. The piling process generates high levels of noise capable of propagating over long distances in the water column; it is important that this noise can be predicted, and the likely environmental impact determined. The model presented in this work comprises a method known as Wave Equation Analysis of Piles. This technique models the stress-wave as it propagates along the pile using a finite-difference approach. Additionally, the plastic nature of the sediment is modeled at both the pile wall and pile toe. Previous work has shown that by considering the radial expansion of the pile, the results of this model may be coupled to an acoustic model. The effect of the sediment on the pile motion has previously received little attention. The results show that although the first downward-going pulse is independent of the sediment parameters, reflected pulses are affected. The calculations show that there exists an impedance condition based on the soil parameters: for low soil stiffnesses the pulse is inverted at the toe end, whereas for higher stiffnesses no inversion is seen.

4:35

2pPA11. Physical models and improvement of bubble curtain for the suppression of underwater noise from a pile drive. Alexander Sutin and Hady Salloum (Stevens Inst. of Technol., 711 Hudson St., Hoboken, NJ 07030, asutin@stevens.edu)

Man-made sounds in ocean and inland water environments become biologically significant when they affect the ability of animals and fish to survive and reproduce. Extremely strong sounds produced by pile driving can highly exceed the environmental safety limits and application of bubble curtain is one of the most effective methods for sound suppression. We present several physical models explaining sound suppression by bubble curtain and discuss the methods of improving bubble curtain efficiency. The physical models of sound suppression by a bubble curtain include: (a) Estimation of bubble curtain impedance leading to decreasing of the pile drive acoustic coupling with surrounding bubbly water and (b) theoretical model for the estimation of sound attenuation by resonance bubbles. The developed models were analyzed for the optimization of the pile drive sound suppression. Several methods for generating a bubble curtain with small bubbles and bubbles with varied sizes are considered. We also suggest a way for improving the efficiency of bubble curtains by increasing the lifetime of the bubbles using bubble coating. Coated micro bubbles are widely used as ultrasound contrast agents in cardiology. One of the simplest ways for micro bubble coating is the passing of bubbles through oil.

TUESDAY AFTERNOON, 3 DECEMBER 2013 CONTINENTAL 2/3, 1:00 P.M. TO 5:00 P.M.

Session 2pPP

Psychological and Physiological Acoustics: Building a Stairway to Hearing

Sridhar Kalluri, Chair
Starkey Hearing Res. Ctr., Starkey Hearing Technologies, 2150 Shattuck Ave., Ste. 408, Berkeley, CA 94704

Contributed Papers

1:00

2pPP1. Testing and extending the Woodworth model. William M. Hartmann (Phys. & Astronomy, Michigan State Univ., East Lansing, MI 48824, hartman2@msu.edu) and Neil L. Aaronson (Natural Sci.& Mathematics, Richard Stockton College of NJ, High Bridge, NJ)

The Woodworth model and formula for interaural time difference is frequently used as a standard in physiological and psychoacoustical studies of binaural hearing for humans and other animals. It is a frequency-independent, ray-tracing spherical head model that is expected to agree with an exact diffraction model in the high-frequency limit. The predictions by the Woodworth model for antipodal ears and for incident plane waves are compared with the predictions of the exact model as a function of frequency to quantify the discrepancy when the frequency is not high. In a second calculation, the Woodworth model is extended to arbitrary ear angles, both for plane-wave incidence and for finite point-source distance. This extended Woodworth model leads to different formulas in six different regions defined by ear angle and source distance. It is noted that the characteristic cusp in Woodworth's well-known function comes from ignoring the longer of the two paths around the head in circumstances when the longer path is actually important. This error can be readily corrected.

1:15

2pPP2. Minimum audible angle at the subjective front during listener's active and passive head rotation. Yôiti Suzuki (Res. Inst. of Elec. Commun., Tohoku Univ., Katahira 2-1-1, Aoba-ku, Sendai 981-0942, Japan, yoh@riec.tohoku.ac.jp), Akio Honda (Tohoku Fukushi Univ., Sendai, Japan), Kagesho Ohba, Shuichi Sakamoto (Res. Inst. of Elec. Commun., Tohoku Univ., Sendai, Japan), and Yukio Iwaya (Tohoku Gakuin Univ., Sendai, Japan)

Listener's head movement, particularly horizontal rotation, effectively improves sound localization acuity (Wallach, 1939; Thurlow, 1967; Kawaura, 1989). However, few findings have been obtained concerning sound localization during head rotation. In the present study, we directly investigated the minimum audible angle (MAA) at the front during horizontal rotation. A sound stimulus (30-ms noise burst) was presented from a loudspeaker of a circular array ($r = 1.1$ m), with a loudspeaker separation of 2.5 degrees. The listener, sitting at the center of the circle, was asked to answer whether the sound stimulus was presented from the left or right of the subjective front (2AFC). We designed three listening conditions, static, active rotation and passive rotation. In the static condition, listeners were asked to keep their heads still. For the active rotation condition, listeners were asked to rotate their heads. Meanwhile, for the passive rotation condition, listeners sitting on a revolving chair were rotated by an experimenter. In the latter two conditions, the test stimulus was triggered during head movement. Results showed the MAA to deteriorate significantly in the two rotation conditions. This implies that the improvements in sound localization due to head motion could be explained by the multiple-look model (Viemeister, 1991).

1:30

2pPP3. Difference of the perceived auditory space between walking and passive self-motion. Shuichi Sakamoto (Res. Inst. of Elec. Commun., Tohoku Univ., 2-1-1 Katahira, Aoba-ku, Sendai, Miyagi 980-8577, Japan, saka@ais.riec.tohoku.ac.jp), Hideaki Terashima (Graduate School of Information Sci., Tohoku Univ., Sendai, Japan), Wataru Teramoto (Dept. of Comput. Sci. and Systems Eng., Muroran Inst. of Technol., Muroran, Japan), Yôiti Suzuki (Res. Inst. of Elec. Commun., Tohoku Univ., Sendai, Japan), and Jiro Gyoba (Graduate School of Arts and Letters, Tohoku Univ., Sendai, Japan)

We are investigating how auditory space was represented during linear self-motion (Teramoto *et al*., 2013). Several studies have suggested that whether the listener's motion is active or passive affected sound localization (Hirahara *et al*., 2013). In the present study, therefore, we set up three conditions: active motion condition, passive motion condition, and no motion condition. In active motion condition, observers were walking straight ahead. In passive motion condition, observers were transported forward by a robotic wheelchair. During the self-motion, a short noise burst was presented from one of the loudspeakers which were aligned parallel to the traveling direction when the listener's coronal plane reached the location of one of the speakers (null point). The listeners indicated the direction in which the sound was perceived relative to their coronal plane (i.e., a two-alternative forced-choice task). The results of experiment showed that the sound position aligned with the subjective coronal plane was displaced compared with the null point. However, there was no significant difference between auditory space in active and passive motion conditions. This result suggests only action of the kinetic system during self-motion and planning and execution of voluntary movement would not affect perceived auditory space.

1:45

2pPP4. Sound source localization from tactile aids for unilateral cochlear implant users. Xuan Zhong, Shuai Wang, Michael Dorman, and William Yost (Dept. of Speech and Hearing Sci., Arizona State Univ., 975 S Myrtle Ave., Lattie F Coor Hall 2211, Tempe, AZ 85287, xuan.zhong@asu.edu)

The present research asks whether two tactile aids with directional microphones, by providing additional inter-channel level information and etc., could help unilateral cochlear implant (CI) localize sound sources. For normal hearing subjects, sound source localization based on tactile vibration cues alone can be as accurate as auditory localization in the frontal horizontal plane (Gescheider, 1970). CI users may as well benefit from additional tactile aids just as normal hearing people do. The current study uses two bone-anchored hearing aids (BAHA) as sources of tactile vibration. The two BAHAs, bonded together by a special gadget to maintain a particular distance and angle, both have directional microphones, and are programed so that one point to the front-left side and the other to the front-right side. Unilateral CI users voluntarily participated in the experimental study. Wide band noise stimuli were presented at 65 dB SPL. The subjects hold one BAHA in each hand and do localization tasks with (1) CI only and (2) with CI and tactile sensation combined. Preliminary data shows CI users can get some benefit from the additional information provided by tactile aids in 360 degree localization (45 degree spacing) on the horizontal plane.

2:00

2pPP5. Characterization of the available feedback gain margin at two device microphone locations, in the fossa triangularis and Behind the Ear, for the light-based contact hearing device. Suzanne C. Levy, Daniel J. Freed (EarLens Corp., 200 Chesapeak Dr., Redwood City, CA 94063, slevy@earlenscorp.com), and Sunil Puria (EarLens Corp., Stanford, CA)

Assistive devices compensate for hearing impairment by amplifying sounds with gain, which is limited by acoustic feedback. The light-based Contact Hearing Device (CHD) provides amplification to 10 kHz by mechanically vibrating the umbo with a wireless Tympanic-Contact Actuator (TCA). Driving the eardrum mechanically generates a pressure wave, which travels laterally down the ear canal and produces feedback. Placing the microphone in the fossa triangularis (FT) may preserve more natural acoustic cues than the BTE location, although it may reduce available feedback gain margin (FGM). Thirteen subjects with bilateral mild-to-severe hearing impairment were fit with CHDs (26 ears). The TCA was driven with light-pulses and the feedback pressure was measured at the FT and above the pinna at the BTE microphone locations. The mean FGM varied from 32 to 48 dB and 38 to 60 dB for the FT and BTE microphone locations, respectively. FGM was lowest in the 3–5 kHz range and highest at about 7 kHz (below 1 kHz FGM is not measurable). The STD of FGM varied from 5.3 to 13 dB due individual anatomies. A microphone at the BTE has 6 to 12 dB additional FGM over the FT location, and allows a broader inclusion range to fit patients with amplification to 10 kHz.

2:15

2pPP6. A long-overdue review of empirical uncertainties in the "fatigue" found through simultaneous dichotic loudness balance. Lance Nizami (Independent Res. Scholar, Wilkie Way, Palo Alto, CA 94306, nizamii2@att.net)

SDLB uses equalization of the loudnesses of stimuli at the two ears (one stimulus intermittent, one constant) to measure the alleged "fatigue" over time of the loudness of the constant stimulus (Hood, 1950). Hood found 50 dB of "fatigue", and SDLB remained influential for over a quarter-century. However, the interpretation of SDLB was long questioned; recently, a novel model uniting the physiology and the behavior emerged (Nizami 2012, Int. Soc. Psychophys., Ottawa, Canada), and others independently re-measured "fatigue". Classically, the stimulus waveforms at the two ears were similar, sometimes identical, permitting two equalization techniques when the stimuli coincided in time: centering of the sound between the ears (lateralization), or matching the loudness contributions from each ear (loudness-matching). Further, "fatigue" was habitually expressed as across-listeners averages. But careful scrutiny reveals that (1) over the "fatiguing" duration, lateralization may give way to loudness-matching, (2) dedicated loudness-matching may nonetheless yield only half as much "fatigue" as lateralization, and (3) the standard deviation of "fatigue" can be half its mean value, such that some listeners would not have "fatigued." In sum, the magnitude of "fatigue" is remarkably uncertain, and is likely to remain so until auditory physiology is compellingly integrated into explanations of SDLB.

2:30

2pPP7. Influence of measurement method and context of presentation on the loudness difference between increasing and decreasing intensity sounds. Emmanuel Ponsot, Patrick Susini (IRCAM, 1 Pl. Igor Stravinsky, Paris, France, patrick.susini@ircam.fr), and Sabine Meunier (LMA, CNRS, UPR 7051, Aix-Marseille Univ, Centrale Marseille, Marseille, France)

Four experiments were conducted to assess the loudness of both increasing and decreasing intensity sounds using different methods and context of presentation as between-subjects factors. In Exp 1 and Exp 2, loudness was assessed directly by using magnitude estimation procedures, with increasing and decreasing sounds presented respectively either in the same block or in separate blocks. In the other two experiments, loudness was measured by the mean of pairwise comparisons. While increasing and decreasing sounds were compared with each other in Exp 3, they were compared respectively with constant-intensity sounds in Exp 4. Two-intervals, 2AFC interleaved-adaptive procedures were used to prevent from potential biases. As a result, very similar trends were observed in the four experiments. In particular, the loudness difference between increasing and decreasing sounds always felt within the same range: decreasing intensity sounds need to be about 3 dB louder that increasing sounds to be perceived with equal loudness. This study thus indicates that this loudness asymmetry actually corresponds to a true perceptual effect and is not due to any experimental bias, since a clear consistency across the results was found using different measurement methods and context of presentation.

2:45–3:00 Break

3:00

2pPP8. Effect of sound duration on loudness estimates of increasing and decreasing intensity sounds. Emmanuel Ponsot, Anne-Laure Verneil, and Patrick Susini (IRCAM, 1 Pl. Igor Stravinsky, Paris 75004, France, ponsot@ircam.fr)

The influence of sound duration on global loudness of non-stationary stimuli was investigated. Loudness of 2 and 6-s increasing and decreasing intensity sounds with different ranges of intensity-variation was assessed using a magnitude estimation procedure. Results once again uphold the existence of a loudness difference between the two patterns: while they only differ in their temporal profile, increasing sounds were perceived louder than decreasing sounds. In addition, global loudness estimates were increased with duration for the two types of sounds, and a small but significant interaction occurred between type and duration. A contrast analysis revealed that while global loudness of increasing and decreasing sounds raised with duration in a similar way in the case of low and moderate intensities (below 75 dB SPL), global loudness was significantly more affected by duration with increasing than with decreasing intensity profiles for high-intensity stimuli. This result suggests the existence of an underlying memory process combined with a "peak-end rule" as being responsible for the loudness asymmetry typically observed between the two types of intensity-pattern being judged [Susini *et al*. (2010). "End level bias on direct loudness ratings of increasing sounds" J. Acoust. Soc. Am. EL **128**(4), 163–168].

3:15

2pPP9. A novel bispectral approach to study cochlear non-linearities in transient evoked otoacoustic emissions. Gabriella Tognola, Alessia Paglialonga, Emma Chiaramello, and Stefano Moriconi (Inst. of Electronics Computer, and Telecommun. Eng. ISIB, CNR It. Natl. Res. Council, Piazza L. da Vinci, 32, Milan 20133, Italy, gabriella.tognola@polimi.it)

A new approach, based on bispectral analysis, is proposed to study non-linearity in cochlear active mechanisms, as evaluated in transient evoked otoacoustic emissions (TEOAEs). Based on an ad hoc formulation of the bispectral analysis, specifically developed in this study to fit the particular features of TEOAEs, this innovative method detects non-linearity by extracting quadratic frequency couplings (QFCs) in TEOAE recordings. The method directly estimates non-linear TEOAE components, which may not be detected by conventional spectral analysis and can thus provide important information useful for better modeling cochlear active mechanisms and to detect sub clinical cochlear dysfunctions. The method was characterized with synthesized TEOAEs as a function of the main TEOAE parameters and then used to analyze TEOAEs recorded in normal hearing adults and full-term neonates to: (i) obtain normative data; (ii) to evaluate test retest repeatability of non-linear interaction mechanisms; and (iii) to investigate the influence of stimulus intensity on non-linear interaction mechanisms. Results revealed that most of the energy of non-linear components is located in the 1–4 kHz range and that the test retest repeatability of non-linear interaction mechanisms is high. The factor that most affects non-linearities is stimulus intensity.

3:30

2pPP10. Finite element model of feed-forward/feed-backward amplification in the mouse cochlea. Joris Soons (Biomedical Phys., Univ. of Antwerp, 496 Lomita Mall, Stanford, California 94305, jsoons@stanford.edu), Sunil Puria, and Charles R. Steele (Mech. Eng., Stanford Univ., Stanford, CA)

Thousands of hair cells in the organ of Corti, situated along the basilar membrane (BM), detect displacements due to sound input. For low input sounds, these displacements are amplified by active outer hair cells (OHCs). A proposed theory is the feed-forward/feed-backward mechanism for the OHC amplification where an expanding hair cell gives a forward push through the Deiters Cells and a backward pull on the BM through the Phalangeal process. Previously this was implemented mathematically using WKB theory (Yoon *et al*. 2011, Biophys. J.). In the present work, we explicitly modeled this as a Y-shaped arrangement of the OHC-DeitersCell-PhallangealProcess to form a building block using beam elements in a finite element formulation. These Y-shaped blocks were chained together to construct a single-row organ-of-Corti model from the base to apex, coupled to the BM and scalae fluid, of a mouse cochlea. The OHC force is proportional to the shear on the BM. For a 10 kHz stapes input tone, the passive BM reaches a peak gain of about 26 dB. For the active case the BM gain increases to 58 dB and shifts apically by about 0.6 mm. These results are consistent with physiological measurements in several other living animals.

3:45

2pPP11. Should the acceptable noise level be considered to be an acceptable noise range? Jonas Brännström, Lucas Holm, Tobias Kastberg (LogopeDC, phoniatrics, and Audiol., Lund Univ., Clinical Sci. Lund, Lund SE-22185, Sweden, jonas.brannstrom@med.lu.se), and Steen J. Olsen (Dept. of Otorhinolaryngology, Head and Neck Surgery, Univ. Hospital, Rigshospitalet, Copenhagen, Denmark)

The acceptable noise level (ANL) test is used to quantify the amount of competing background noise (BNL) that a listener is willing to accept when listening to speech at the most comfortable level (MCL). ANL is calculated by subtracting the BNL from the MCL. Most studies show large intersubject ANL variability and a few also demonstrate large intrasubject variability. Very few predictor variables for ANL have been identified and it has been proposed that the ANL depends on an inherent characteristic of the listener. However, some of the variability seems to depend on poor precision of the ANL test. After removing the effect of poor precision, some variability still remains. One possible explanation for these findings may be that the ANL is not a single level but a range of levels. Using recent data, this presentation examines the notion of an acceptable noise range.

4:00

2pPP12. Effects of speakers' language background on speech perception in adults. Mark A. Dame, Harisadhan Patra, Petula C. Vaz (Audiol. & Speech Pathol., Bloomsburg Univ. of PA, 226 CEH, 400 E 2nd St., Bloomsburg, PA 17815, hpatra@bloomu.edu), and Biswajit Ray (Electronics Eng. Technol., Bloomsburg Univ. of PA, Bloomsburg, PA)

With an increasingly changing sociolinguistic environment in the U.S., clinicians are challenged to accurately evaluate individuals' speech perception under realistic everyday listening conditions. This study investigated how contextual linguistic information, speakers' language background, speech-rate, and background noise individually and interactively affect listeners' ability to recognize sentences. Ten normal-hearing American English native speakers, aged 20–22 years were recruited as listeners. Eight normal-hearing native speakers of four different languages (American English, Chinese-Mandarin, German, and Spanish) were recruited as speakers. These speakers read 120 sentences, 60 high-predictability and 60 low-predictability sentences from the revised Speech in Noise test (Bradlow and Alexander, 2007), which served as test stimuli. The listeners reported the target word of each sentence presented either in quiet or in multitalker babble. Results revealed that noise, high speech-rate, and lack of contextual cues could have significant adverse affects on listeners' test scores. Speakers' language background also had significant effects on listeners' performance. Specifically, listeners had the most difficulty in perceiving the Chinese-Mandarin speakers, followed by the Spanish, and German speakers; more so in multitalker babble than in quiet. Further studies are warranted since results may have implications for audiology diagnosis and rehabilitation to address effective everyday communication.

4:15

2pPP13. Training Mandarin-speaking amusics to recognize pitch direction: Pathway to treat musical disorders in congenital amusia? Fang Liu (Dept. of Linguist and Modern Lang., The Chinese Univ. of Hong Kong, Rm. G36, Leung Kau Kui Bldg., Shatin, N.T., Hong Kong, China, fangliufangliu@gmail.com), Cunmei Jiang (Music College, Shanghai Normal Univ., Shanghai, China), Tom Francart (ExpORL, Dept. of NeuroSci., KU Leuven, Leuven, Belgium), Alice H. Chan (Div. of Linguist and Multilingual Studies, School of Humanities and Social Sci., Nanyang Technolog. Univ., Singapore, Singapore), and Patrick C. Wong (Dept. of Linguist and Modern Lang., The Chinese Univ. of Hong Kong, Hong Kong, China)

Congenital amusia is a lifelong disorder of musical perception and production that has been hypothesized as due to impaired pitch direction recognition. This study investigated whether amusics could be trained to identify pitch direction in speech and music, and if so, whether enhanced pitch direction recognition would benefit musical processing in amusia. Eighteen Mandarin-speaking amusics and 18 matched controls were evaluated using the Montreal Battery of Evaluation of Amusia (Peretz *et al.*, 2003) and tested on two psychophysical pitch threshold tasks for identification of pitch direction in the Mandarin syllable /ma/ and its piano tone analog. Subsequently, nine of the eighteen amusics undertook a 10-session training program on pitch direction identification in /ma/ and piano tone. Compared with those untrained, trained amusics demonstrated significantly improved thresholds for pitch direction identification in speech syllables [pretest: 2.92 (4.63) semitones; posttest: 0.16 (0.06)], and marginally significant improvement in pitch direction identification thresholds for piano tones [pretest: 5.20 (4.56); posttest: 0.89 (1.46)]. These findings suggest that pitch sensitivity of individuals with congenital amusia could be improved through auditory training, providing evidence for neural plasticity in the amusic brain, which may lead the way to other rehabilitative programs for treating this musical disorder.

4:30

2pPP14. Neuromagnetic beta-band oscillation for rhythmic processing induced by subjectively accented structure. Takako Fujioka (CCRMA (Ctr. for Comput. Res. in Music and Acoust.), Dept. of Music, Stanford Univ., 660 Lomita Dr., Stanford, CA 94305-8180, takako@ccrma.stanford.edu), Laurel J. Trainor (Dept. of Psych., Neurosci. and Behavior, McMaster Univ., Hamilton, ON, Canada), and Bernhard Ross (Rotman Res. Inst., Baycrest, Toronto, ON, Canada)

Musical rhythm facilitates synchronized body movements and schema-based, predictive timing perception. Our previous magnetoencephalography (MEG) study demonstrated that beta-band (~20 Hz) activity in bilateral auditory cortices shows synchronized modulation that predicts the time point of the next beat (Fujioka *et al.* 2009, 2012). Furthermore, after finger tapping to a different musical meter such as a march or waltz (every 2nd or 3rd beat), the broadband evoked response from auditory cortex differentiates the metric conditions (Fujioka *et al.*, 2010). Here we examined how beta-band activity indexed subjective metrical perception during listening to unaccented beats (1) after listening to acoustically accented beats, and (2) after finger tapping with either the left or right index finger. The auditory cortices showed beat-synchronized modulation in line with the previous studies in both march and waltz conditions. However, distinction between down-beat and up-beat positions was stronger in march than in waltz condition, with symmetrical activities in the left and right auditory cortices. This contrast between the two metric conditions was stronger in the side contralateral to the tapping finger in the tapping condition. This suggests that contribution of auditory cortices to metric processing depends on both its timing structure and rhythmic movement in the contralateral hemisphere.

4:45

2pPP15. On the learnability of auditory concepts. Ronaldo Vigo, Mikayla Barcus, Yu Zhang, and Charles Doan (Ohio Univ., 200 Porter Hall, Athens, OH 45701, vigo@ohio.edu)

The field of categorization and concept learning research has been dominated by findings involving visual stimuli. Among these findings is the learning difficulty ordering of the family of category structures associated with visual categorical stimuli consisting of four objects defined over three dimensions. This ordering has been observed numerous times in several rigorous studies (Kruschke, 1992; Love *et al.*, 2004; Nosofsky *et al.*, 1994; Shepard *et al.*, 1961; Vigo, 2011a, 2013a. 2013b) and has been influential in shaping current theories of conceptual behavior. In recent work, we have freed the field from this visual bias by examining the learnability of auditory categorical stimuli that are instances of the aforementioned structures. We found that, in general, for auditory categorical stimuli the learning difficulty ordering of these structures is somewhat different from that of their visual counterparts. However, we also found that this difference may be explained and accurately predicted by a simple encoding mechanism proposed in generalized invariance structure theory (GIST; Vigo, 2013b). We view this result as evidence in support of the proposition that a single basic conceptual system underlies the acquisition of both auditory and visual concepts.

Session 2pSA

Structural Acoustics and Vibration: Computational Structural Acoustics

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Chair's Introduction—1:00

Invited Papers

1:05

2pSA1. Material identification in frequency-domain coupled acoustic-structure interaction using an error in constitutive equation functional. Wilkins Aquino, James Warner, and Manuel Diaz (Civil and Environ. Eng., Duke Univ., Hudson Hall, Durham, NC 27708, wa20@duke.edu)

This work focuses on the inverse identification of linear elastic material parameters in the context of frequency-domain, coupled acoustic-structure interaction. The approach postulates the inverse problem as an optimization problem where the solution is obtained by minimizing a modified error in constitutive equation (MECE) functional. The MECE functional measures the discrepancy in the constitutive equations that connect kinematically admissible strains and dynamically admissible stresses while incorporating the measurement data as an additional quadratic penalty term. The method is formulated generally for the recovery of material properties in a coupled acoustic-structure system using solid displacement and/or fluid pressure measurement data. Numerical results demonstrate that the proposed methodology can identify the spatial distribution of elastic moduli from partial and noisy measurements taken in either the fluid or solid subdomains.

1:25

2pSA2. A parallel Helmholtz solver for acoustics and structural acoustics. Clark R. Dohrmann and Timothy F. Walsh (Computational Solid Mech. & Structural Dynam., Sandia National Labs., P.O. Box 5800, M.S. 0380, Albuquerque, NM 87185-0380, crdohrm@sandia.gov)

In this talk, we present a parallel Helmholtz solver for the solution of acoustic and structural acoustic problems in the frequency domain. The solver is based on overlapping Schwarz domain decomposition concepts for parallelism, and artificial structural damping is introduced in the preconditioner to deal with resonant or near resonant conditions at the subdomain level. An efficient method to solve problems over a range of frequencies is also described. Numerical examples are presented for acoustic and structural acoustic problems, as well as a study of solver performance for higher order finite element discretizations.

1:45

2pSA3. Computational methods for the interior structural acoustics of small spaces. Karl Grosh, Yizeng Li (Dept. of Mech. Eng., Univ. of Michigan, 2350 Hayward St., Ann Arbor, MI 48109-2125, grosh@umich.edu), and Robert Littrell (Baker-Calling, Inc., Santa Monica, CA)

In some biomechanical systems and micro-electro-mechanical systems (MEMS), the interaction of a viscous compressible fluid confined in a space bounded in part by a flexible structure is of central importance. Two specific examples are MEMS microphones (condenser or piezoelectric) and the cochlea. In both the manmade and biological acoustical sensor, the interior space is typically smaller than an acoustic wavelength, and a successful design involves trade-offs between sensitivity, bandwidth, and noise (including thermal, mechanical, electrical, or channel generated noise); the latter two criteria depend critically on the viscous and thermal forces in the system. A direct numeric approach to modeling viscous and thermal effects is often prohibitively expensive, as boundary layers must be resolved in the mesh. In this talk, we will present approximate methods that enable the inclusion of viscothermal effects in a computational framework. In particular, a variational approach amenable to inclusion in a finite element based code will be presented. Results for this method, which retains the accuracy of a Navier-Stokes formulation with the computational cost of a standard scalar acoustic formulation, will be given along with the limitations of the method and a discussion of alternative numerical approaches.

Contributed Papers

2:05

2pSA4. Convolution formulations for non-negative intensity in a plane. Earl G. Williams (Acoust. Div., Naval Res. Lab., Code 7106, 4555 Overlook Ave., Washington, DC, DC 20375, earl.williams@nrl.navy.mil)

New spatial convolution formulas for a variant of the active normal intensity in planar coordinates have been derived that use measured pressure or normal velocity near-field holograms to construct a positive-only (outward) intensity distribution in the plane, quantifying the areas of the vibrating structure that produce radiation to the far-field. This is an extension of the outgoing-only (unipolar) intensity technique recently developed for arbitrary geometries by Steffen Marburg. The method is applied independently to pressure and velocity data measured in a plane close to the surface of a point-driven, unbaffled rectangular plate in the laboratory. It is demonstrated that the sound producing regions of the structure are clearly revealed using the derived formulas and that the spatial resolution is limited to a half-wavelength. A second set of formulas called the hybrid-intensity formulae are also derived which yield a bipolar intensity using a different spatial convolution operator, again using either the measured pressure or velocity. Using the experiment results it is shown that the velocity formula yields the classical active intensity and the pressure formula an interesting hybrid intensity that may be useful for source localization. Computations are fast and carried out in real space without Fourier transforms into wavenumber space. [Work supported by the Office of Naval Research.]

2:20

2pSA5. Minimization of the sound radiated by a curved underwater panel. Micah R. Shepherd and Stephen A. Hambric (Appl. Res. Lab, Penn State Univ., PO Box 30, M.S. 3220B, State College, PA 16801, mrs30@psu.edu)

The sound radiated by a curved underwater panel excited by a point drive is minimized using an evolutionary strategy. The panel thickness is varied in incremental strips along the length of the panel to obtain the optimal configuration. The panel weight and sound radiation are minimized simultaneously using a single weighted objective function. The weighting coefficient is then varied to obtain a Pareto front describing the competing nature of the two objectives. The optimal designs are compared for each weighting coefficient illustrating the tradeoff between minimizing weight and radiated sound power. Normal modes and radiation efficiency curved are also compared. When the optimizer favors minimizing weight, the element thickness is uniformly minimized. Conversely, when the optimizer favors minimizing sound power, the optimal design accepts energy less easily and becomes an inefficient radiator.

2:35

2pSA6. Vibroacoustic modeling of ventilation window system with internal partial partitions. Xiang Yu and Li Cheng (Dept. of Mech. Eng., The Hong Kong Polytechnic Univ., Hung Hom, Kowloon 00000, Hong Kong, li.cheng@polyu.edu.hk)

Domestic buildings exposed to traffic and environmental noise has always been a major concern. Among various noise control measures, ventilation windows offer appealing features by simultaneously allowing good sound attenuation and air ventilation. However, the modeling of such systems is challenging in that the simulation tools should cope with the complexity of the system; to reach reasonably high frequency range; and to offer the flexibility needed for system optimization. In the present study, the sound transmission through a double-glazed ventilation window is investigated using the patch transfer function method along with a handy treatment of the air aperture and micro-perforated elements. Numerical results show the sound insulation performance can be improved with internal partial partitions, which can be either solid or micro-perforated.

2:50–3:05 Break

3:05

2pSA7. Sound field alteration through cavity shape design using a Wavelet-Galerkin Method. Su Zhang (Dept. of Mech. Eng., The Hong Kong Polytechnic Univ., Hong Kong, Hong Kong) and Li Cheng (Dept. of Mech. Eng., The Hong Kong Polytechnic Univ., Hung Hom, Kowloon 00000, Hong Kong, li.cheng@polyu.edu.hk)

The general problem of internal sound field prediction along with the manipulation of the cavity boundary shape to alter the sound pressure distribution is dealt with in this paper. Owing to the compactly supported orthogonal property of the wavelet and its extraordinary fitting capability, Daubechies wavelet scaling function is used as a global basis function to expand the unknown sound pressure under the general Galerkin framework. The proposed formulation is shown to offer high accuracy on the numerical calculation of the sound pressure field with the use of a remarkably small number of messing points. A genetic based shape optimization algorithm is proposed and demonstrated. As an example, an enclosure with an inner rigid acoustic screen is investigated. By optimizing the shape of the screen, the sound pressure level within a chosen area is successfully reduced. Results show the remarkable potentials of the proposed approach as a topology optimal tool for the general inner sound field problems.

3:20

2pSA8. A comparison of perfectly matched layers and infinite elements for exterior Helmholtz problems. Gregory Bunting (Computational Solid Mech. and Structural Dynam., Sandia National Labs., 709 Palomas Dr. NE, Albuquerque, NM 87108, gbunting@purdue.edu), Arun Prakash (School of Civil Eng., Purdue Univ., West Lafayette, IN), and Timothy Walsh (Computational Solid Mech. and Structural Dynam., Sandia National Labs., West Lafayette, Indiana)

Perfectly matched layers and infinite elements are commonly used for finite element simulations of acoustic waves on unbounded domains. Both involve a volumetric discretization around the periphery of an acoustic mesh, which itself surrounds a structure or domain of interest. Infinite elements have been a popular choice for these problems since the 1970s. Perfectly matched layers are a more recent technology that is gaining popularity due to ease of implementation and effectiveness as an absorbing boundary condition. In this study, we present massively parallel implementations of these two techniques, and compare their performance on a set of representative structural-acoustic problems on exterior domains. We compare the ability of these methods to absorb acoustic waves on ellipsoidal domains with waves of relatively high oblique angles of incidence. We also examine the conditioning of the linear systems generated by the two techniques by examining the number of Krylov-iterations needed for convergence to a fixed solver tolerance. [Sandia National Laboratories is a multi-program laboratory managed and operated by Sandia Corporation, a wholly owned subsidiary of Lockheed martin Corporation, for the U.S. Department of Energy's National Nuclear Security Administration under contract DE-AC04-94AL850000.]

3:35

2pSA9. A plane wave method for modeling acoustic variables in cavities. Matthew Kamrath (Graduate Program in Acoust., Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, kamrath64@gmail.com) and Gary Koopmann (KCF Technologies, Inc., State College, PA)

The pressure field within a cavity that contains a homogeneous fluid is modeled in the frequency domain as a superposition of N plane waves with arbitrary orientations and unknown complex amplitudes. Specifying the pressure, the normal velocity, or the specific impedance at N locations produces a system of N linear equations with N unknowns. Then, solving for the complex amplitudes yields an approximation of the pressure and velocity fields inside the cavity. Preliminary comparisons to analytic results are presented.

3:50

2pSA10. Tuned elastic shells with matched acoustic impedance and sound speed in water. Alexey Titovich and Feruza Amirkulova (Mech. and Aerosp. Eng., Rutgers Univ., 98 Brett Rd., Piscataway, NJ 08854, alexey17@eden.rutgers.edu)

Tuning of an elastic shell to match the acoustic properties of the surrounding fluid (water) is done with an internal mechanism consisting of a concentered central mass supported by an axisymmetric distribution of elastic stiffeners. The effective impedance and index are both matched at low frequencies (ka < 1) by adjusting the mass and stiffness of the internal oscillator for a given shell thickness. The subsonic flexural modes of the shell, which are excited by the point attachments of the stiffeners, are ultimately suppressed in the low frequency range with a sufficiently large number of stiffeners. As a result, the scattering cross-section of the tuned shell-stiffener-mass system is negligible compared to the empty shell at frequencies below the resonance of the internal oscillator. An optimal shell thickness exists which maximizes the frequency range of water-like behavior. Furthermore, tuning the effective properties of each such shell in an array is used to achieve acoustic transparency and lensing. Inclusion of damping decreases the magnitude of the internal oscillator resonance without significantly affecting the scattered field. Several planar simulations are presented demonstrating the applications.

4:05

2pSA11. Analytical scattering method of flexural waves considering the evanescent field in a thin plate. Sungjin Cho and Junhong Park (Mech. Eng., Hanyang Univ., Haengdang-dong 17, Seongdong-gu, Seoul 133-791, South Korea, sjcho0407@hanyang.ac.kr)

This paper presents a theoretical solution on analyzing the scattering behaviors of the flexural waves by a circular obstacle medium having different material properties. In deriving the exact solution, the evanescent field together with the scattered field is considered. The modified Bessel function of the second kind is used for the evanescent field. As the evanescent wave is exponentially decaying in all directions near the edge of a circular obstacle medium when the mass density and critical angle are discontinuous, the scattering pattern without evanescent field is difficult to analyze the wave scattering phenomenon near the boundary layer. The resulting equations are solved using symbolic calculation of software MATLAB. The dependence of the scattering behaviors on the mechanical properties of the inner medium is investigated. The influence of the evanescent field on the scattering is also investigated in various configurations. The result shows that the scattering pattern has a strong dependence on the evanescent field in near field of a circular obstacle.

4:20

2pSA12. Influence of cross-sectional discontinuity on the damping characteristics of viscoelastically supported rectangular plates. Jeongwon Park, Sangkeun Ahn (Dept. of Mech. Eng., Hanyang Univ., 222 Wangsimni-ro, Seongdong-gu, Seoul 133-791, South Korea, jwparks@hanyang.ac.kr), Ji Woo Yoo (Res. and Development Div., Hyundai-Kia Motors, Hwaseng, Gyeonggi, South Korea), and Junhong Park (Dept. of Mech. Eng., Hanyang Univ., Seoul, South Korea)

Cross-sectional shape of a structure is one of the important design parameter associated with the vibration responses and resulting sound radiation. In this study, the influence of cross-sectional discontinuity on the vibration characteristics of plate structures was investigated. The variation of cross-sectional geometry in the rectangular plate was modeled as the change of bending stiffness for flexural wave propagation analysis. The complex translational and rotational stiffness at plate edges were used for modeling of the damping at the boundaries. The ratio between the incident and reflected waves from the boundaries was predicted for the flexural waves of different wavelengths to analyze the effect of support stiffness on the vibration damping. Using the wave propagation model, the condition of the viscoelastic boundary properties and the discontinuous flexural stiffness of plate for minimum reflection ratio, i.e., maximum dissipation of the vibration was calculated. Modal damping characteristics of the plate with and without the discontinuity in the cross-section were measured and compared to the predicted reflection ratios. The measured damping ratios on the different boundary conditions showed similar pattern with the predicted vibration energy dissipation at the viscoelastic supports.

4:35

2pSA13. Modal analysis of a vibrating string via electromagnetic field excitation. Anton A. Filyayev (Audio Arts & Acoust., Columbia College Chicago, 33 E. Congress Ave., Chicago, IL 60604, anton.filyayev@loop.colum.edu)

An electromagnetic field (EM) generating device ("EBow") was used to excite a fixed string held in tension between two boundaries on a custom-built apparatus in the style of a monochord. A high-speed camera was used to visualize and record the vibration of the string across the length of the string for every EBow position. The data sets were analyzed to create amplitude vs time graphs. Every excitation position was found to result into a different mode of vibration that emphasized fundamental, second, third, and fourth eigenmodes. Each mode resulted in a distinct displacement pattern, which was found to follow curve fits with R<+>2<+> values ranging from 0.94 to 0.99. The recorded amplitudes also agreed with expected nodal and anti-nodal positions for every observed mode signifying the data were robust. A discrete Fourier series expansion was performed on the polynomial fits and yielded simple second-order expressions (including a damping term) which are in agreement with the expected excitation models for a fixed string.

Session 2pSCa

Speech Communication: Speech Technology

Yi Xu, Chair
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Contributed Papers

1:30

2pSCa1. Measurement of formants in synthetic vowels. Christine H. Shadle, Hosung Nam (Haskins Labs., 300 George St., New Haven, CT 06511, shadle@haskins.yale.edu), and D. H. Whalen (Speech and Hearing Sci., CUNY Graduate Ctr., New York, NY)

The measurement of formant frequencies of vowels is among the most common measurements in speech studies, but is known to be biased by the particular fundamental frequency (F0) exciting the formants, and to be inaccurate for formants close together or for speakers using a high F0. To allow a comparison across multiple measurement techniques, vowels were synthesized using the Klatt synthesizer with known formant values. The synthetic vowels were constructed with five different F1 values and nine different F0 values; formant bandwidths, and higher formant frequencies, were constant. The F0s varied in such a way that the most intense harmonic in F1 or F2 either matched the center frequency or deviated in the range of 3–87 Hz. Manual measurements by four subjects were compared to automatic measures using the LPC Burg algorithm, LP closed-phase covariance, and spectra smoothed cepstrally or by averaging repeated DFT's. Formants were also measured from pruned reassigned spectrograms. Error patterns differ among the methods, but most tracked the frequency of the most intense harmonic; the smallest errors occur with closed-phase covariance and reassigned spectrogram. Implications for such measures on vowels in isolated words of real speech are discussed. [Work supported by NIH-NIDCD grant DC-002717.]

1:45

2pSCa2. A further comparison of fundamental frequency tracking algorithms. Hongbing Hu (Intel Corp., Binghamton, New York), Peter Guzewich, and Stephen Zahorian (Elec. and Comput. Eng., State Univ. of New York at Binghamton, PO Box 6000, Binghamton, NY 13902, zahorian@binghamton.edu)

"Yet another Algorithm for Pitch Tracking -YAAPT" was published in a 2008 JASA paper (Zahorian and Hu), with additional experimental results presented at the fall 2012 ASA meeting in Kansas City. The results presented in both the journal paper and at the fall 2012 meeting indicated that YAAPT generally has lower error rates than other widely used pitch trackers (YIN, PRAAT, and RAPT). However, even YAAPT-created pitch tracks had significant "large" errors (pitch doubling and pitch-halving) for both clean and noisy speech. Recently additional post-processing heuristics have been incorporated to reduce the incidence of these type errors—thus reducing the need for hand correcting pitch tracks for situations where extremely accurate tracks are desired. For the case of an all-voiced track, interpolation through unvoiced intervals has been improved. The updated version of YAAPT is presented along with experimental results. The experiments are conducted with multiple databases, including British English, American English, and Mandarin Chinese. For most conditions evaluated, YAAPT gives better performance than the other fundamental frequency trackers.

2:00

2pSCa3. Automated assessment of English fundamental frequency contours for non-native speakers from China and India. Keelan Evanini and Xinhao Wang (Educational Testing Service, Rosedale Rd., M.S. R-11, Princeton, NJ 08541, kevanini@ets.org)

This study investigates the F0 contours produced by non-native speakers of English from two different countries. Speakers from China ($N = 202$; L1 Mandarin) and India ($N = 230$; multiple L1s) read a paragraph out loud in the context of an assessment of non-native English speaking proficiency; in addition, a control set of native speakers ($N = 85$) read the same paragraph. The words and phonemes in all of the responses were provided with time stamps using forced alignment with the prompt text, and mean normalized F0 values were extracted for each word from each speaker. A model F0 template for the paragraph was generated by taking the mean word-level F0 value across all native speakers, and the correlation between the word-level F0 contour and this template was calculated for each non-native speaker (as in Schwanenflugel *et al.* 2004). This correlation was used as a feature for assessing F0 contours and was correlated with expert human scores of English proficiency. The results demonstrated that the feature performed somewhat better for the speakers from India ($r = 0.390$) than the speakers from China ($r = 0.326$). Furthermore, the correlations improved when function words were removed from the analysis and only content words were considered ($r = 0.396$ and $r = 0.346$).

2:15

2pSCa4. Detecting voice disguise from speech variability: Analysis of three glottal and vocal tract measures. Talal B. Amin (Elec. and Electron. Eng., Nanyang Technolog. Univ., Singapore, Singapore), James S. German (Humanities and Social Sci., Nanyang Technolog. Univ., HSS-03-46, Singapore, Singapore, jsgerman@ntu.edu.sg), and Pina Marziliano (Elec. and Electron. Eng., Nanyang Technolog. Univ., Singapore, Singapore)

The deliberate attempt by speakers to conceal their identity (voice disguise) presents a challenge for forensics and for automated speaker identification systems. Using a database of natural and disguised voices of three professional voice impersonators, we build on earlier findings (Amin *et al*., 2012) by exploring how certain glottal and vocal tract measures, including fundamental frequency (f0), glottal timing (Open Quotient), and vowel formants, are exploited to create novel voice identities. Specifically, we explored whether the amount and type of variation exhibited by impersonators can be used to develop a metric for distinguishing natural from disguised voices. As expected, variation in f0 and Open Quotient was speaker-dependent, and corresponded closely to social attributes (i.e., gender/age) of the voice identities involved. In a novel finding, the effects of voice identity on vowel formants were highly dependent on vowel category, and could not be readily characterized as global modifications to the vowel space (Bradlow *et al*.,1996). We therefore developed a no-reference objective metric for voice disguise that treats formant variability on a vowel-by-vowel basis. This metric consistently assigned high rankings to natural voices (3.3/27 on average). This correlated closely with the subjective disguisedness ratings of 18 naïve listeners, even outperforming them slightly.

2:30

2pSCa5. Glottal articulations in tense vs lax phonation contrasts. Jianjing Kuang (Linguist, Univ. of Pennsylvania, UCLA Campbell Hall 3125, Los Angeles, California 90095, kuangjj@gmail.com) and Patricia Keating (UCLA, Los Angeles, CA)

This study explores the glottal articulations of one type of phonation contrast—the tense vs lax phonation contrasts of three Yi (Loloish) languages—which is interesting because neither phonation type is very different from modal voice, and both are independent of the languages' tonal contrasts. Electroglottographic (EGG) recordings were made in the field, and traditional EGG measures showed many small but significant differences between the phonations. Tense phonation involves more overall contact and briefer but slower changes in contact. Functional Data Analysis was then applied to entire EGG pulse shapes, and the resulting first principal component was found to be mostly strongly related to the phonation contrasts, and correlated with almost all the traditional EGG measures. Unlike the traditional measures, however, this component also captures differences in abruptness of contact. Furthermore, previously-collected perceptual responses from native speakers of one of the languages correlated better with this component than with any other EGG measure or any acoustic measure. The articulatory differences between these tense and lax phonations, involving glottal aperture and how glottal closure is made, are not extreme, but apparently they are consistent enough, and perceptually robust enough, to support this linguistic contrast. [Work supported by NSF.]

2:45–3:00 General Discussion

3:00–3:30 Break

3:30

2pSCa6. The effect of non-linear dimension reduction on Gabor filter bank feature space. Hitesh A. Gupta, Anirudh Raju, and Abeer Alwan (Elec. Eng., Univ. of California Los Angeles, 550 Veteran Ave., Apt. 102, Los Angeles, CA 90024, hiteshag@ucla.edu)

In this paper, we modify the Gabor feature extraction process, while applying the Gabor filters on the power-normalized spectrum and concatenating with power normalized cepstrum coefficients (PNCC), for noise robust large vocabulary continuous speech recognition. In Chang *et al*., ICASSP (2013), a similar Gabor filter bank (GBFB) feature set with multilayer perceptron (MLP) processing (to reduce the feature dimension) has been used with mel frequency cepstrum coefficients showing improvements on Aurora-2 and renoised Wall Street Journal corpora. On a subset of the Aurora-4 database (only male), our method has shown promising results (when using PCA) being 7.9% better than 39-dimensional PNCC features. But, the GBFB features are a rich representation of the speech spectrogram (as an overcomplete basis), and an appropriate dimension reduction/manifold learning technique is the key to generalizing these features for the large vocabulary task. Hence, we propose the use of Laplacian Eigenmaps to obtain a reduced manifold of 13 dimension (from a 564-dimensional GBFB feature set) for the training dataset with a MLP being used to learn the mapping so that the same can be applied to out-of-sample points, i.e., the test dataset. The reduced GBFB features are then concatenated with the 26-dimension PNCC plus acceleration coefficients. This technique should lead to better accuracies as speech lies on a non-linear manifold rather than a linear feature space. [This project was supported in part by DARPA.]

3:45

2pSCa7. Language material for English audiovisual speech recognition system development. Andrzej Czyzewski (Multimedia Systems Dept., Gdansk Univ. of Technol., Narutowicza 11/12, Gdansk 80-233, Poland, ac@pg.gda.pl), Tomasz Ciszewski, Dorota Majewicz (Faculty of Philology, Univ. of Gdansk, Gdansk, Poland), and Bozena Kostek (Multimedia Systems Dept., Gdansk Univ. of Technol., Gdansk, Poland)

The bi-modal speech recognition system requires a 2-sample language input for training and for testing algorithms which precisely depicts natural English speech. For the purposes of the audio-visual recordings, a training data base of 264 sentences (1730 words without repetitions; 5685 sounds) has been created. The language sample reflects vowel and consonant frequencies in natural speech. The recording material reflects both the lexical word frequencies and casual speech sound frequencies in the BNC corpus of approx. 100m words. The semantically and syntactically congruent sentences mirror the 100m-word corpus frequencies. The absolute deviation from source sound frequencies is.09% and individual vowel deviation is reduced to a level between 0.0006% (min.) and 0.009% (max.). The absolute consonant deviation is 0.006% and oscillates between 0.00002% (min.) and 0.012% (max.). Similar convergence is achieved in the language sample for testing algorithms (29 sentences; 599 sounds). The post-recording analysis involves the examination of particular articulatory settings which aid visual recognition as well as co-articulatory processes which may affect the acoustic characteristics of individual sounds. Results of bi-modal speech elements recognition employing the language material are included in the paper.

4:00

2pSCa8. Characteristics of automatic and human speech recognition processes. Mark VanDam (Speech & Hearing Sci., Washington State Univ., PO Box 1495, Spokane, WA 99202, mark.vandam@wsu.edu) and Noah H. Silbert (Commun. Sci. & Disord., Univ. of Cincinnati, Cincinnati, OH)

In a previous report [VanDam and Silbert (2013) *POMA* **19**, 060006], we investigated performance of a commercially available automatic speech recognition (ASR) system [LENA Research Foundation, Boulder, CO] on acoustic recordings from family speech in naturalistic environments. We found that the ASR more accurately labeled children over adults and fathers over mothers, and human judge labels included substantial individual variation. The present work extends previous work by investigating the possible sources for both machine- and human labeling decisions. Classification tree models were fit to several acoustic variables for machine- and human labels of *CHILD*, *MOTHER*, and *FATHER*. Results suggest that (a) fundamental frequency (f_0) and duration measures influenced label assignment for both machine and human classifications, (b) the error of the fitted models is lower for the machine labeling procedure than for human judges, (c) machine- and human decision processes use the acoustic criteria (i.e., f_0 and duration) differently, and (d) f_0 is more important than duration for all labelers. Results may have implications for improving implementation and interpretation of ASR techniques, especially as they are useful for understanding child language applications and very large, naturalistic datasets that demand unsupervised ASR techniques.

4:15

2pSCa9. Crying for help: The Frye hearing and forensic acoustic analyses in State of Florida vs George Zimmerman. Al Yonovitz (The Univ. of Montana, Dept. of Communicative Sci. and Disord., Missoula, MT 59812, al.yonovitz@umontana.edu), Herbert Joe (Yonovitz and Joe, LLP, Irvine, CA), and Joshua Yonovitz (Yonovitz and Joe, LLP, Missoula, Montana)

Neighborhood watch volunteer George Zimmerman observed suspicious activity and called police. In minutes, he and Trayvon Martin were in an altercation when Mr. Zimmerman shot and killed Trayvon Martin and claimed self defense. The State's audio experts evaluated the 9-1-1 audio recordings to determine who spoke which background phrases, if any. A Frye hearing is where the court determines the reliability and admissibility of an expert's opinion by determining whether an expert's methodologies are generally accepted within the relevant scientific community. The judge in this case ruled that the State's two audio experts would not be permitted to testify during the trial. While the judge found the "aural perception and spectral analysis... are sufficiently established to have gained general acceptance within the scientific community," she took exception to how they were applied in this case. This case provided a venue and forum for opinions on voice analysis, methodologies, standards, and the quality required of evidentiary audio for determination of speaker identification and elimination. Positions taken by witnesses, a critique of the Frye Hearing and the scientific basis for witness and legal conclusions will be discussed.

4:30

2pSCa10. Intensity slopes as robust measure for distinguishing glottalic vs pulmonic stop initiation. Sven Grawunder (Dept. of Linguist, Max Planck Inst. for Evolutionary Anthropology, Deutscher Platz 6, Leipzig 04103, Germany, grawunder@eva.mpg.de)

A novel cross-linguistically robust measure is introduced for the linguistically relevant distinction of pulmonic vs glottalic (ejective) stops.

We propose to parametrize the abruptness of the following vowel onset by using intensity slope (RMS-trajectory) at the voicing onset. This measure was previously discussed only for voicing distinction of pulmonic stops (Harrington, 2012). The dependencies on vowel quality of the following vowel, voice onset time (VOT), phonetic prominence and speaking rate are investigated on a small-scale sample of two speakers from Avar (Nakh-Dagestanian), Ingush (Nakh-Dagestanian), and Georgian (Kartvelian). The results demonstrate robust significant differences of intensity slopes between pulmonic and glottalic stops, the latter showing the steepest slopes/the most abrupt onsets. In some cases of prevoiced stops, the intensity slopes allow even a tripartite distinction of (pre-)voiced, voiceless (aspirated), and ejective stops. However, in order to countervail the influence of vowel specific onset characteristics (e.g., degree of lip rounding) or the influence of VOT, the sample must be controlled for vowel quality and place of articulation. And, since higher speaking rates and less prominent syllables involve higher rates of laryngealized (creaky) vowel onsets, a breakdown of the abruptness distinction is observed under these conditions.

4:45–5:00 General Discussion

TUESDAY AFTERNOON, 3 DECEMBER 2013 — PLAZA A, 1:00 P.M. TO 5:00 P.M.

Session 2pSCb

Speech Communication: Speech Perception II (Poster Session)

Meghan Sumner, Chair
Dept. of Linguist., Stanford Univ., Margaret Jacks Hall, Bldg. 460, 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.

2pSCb1. Perception of Scottish Gaelic alternating (leniting) consonants. Natasha L. Warner (Dept. of Linguist, Univ. of Arizona, Box 210028, Tucson, AZ 85721-0028, nwarner@u.arizona.edu), Ian Clayton (Dept. of English, Boise State Univ., Boise City, ID), Andrew Carnie, Muriel Fisher, Dan Brenner, Michael Hammond, Diana Archangeli, and Adam Ussishkin (Linguist, Univ. of Arizona, Tucson, AZ)

Scottish Gaelic, an endangered Celtic language, demonstrates alternations in word-initial consonants, as in “pòg” [p^hok] ’kisses’ vs “phòg” [fok] ’kissed.’ This process, called lenition, leads to apparent neutralizations of Gaelic segments, for example of the [f] of “ phòg” with [f] of “foghlam” [foɫəm] “education,” which is not caused by lenition. A perception experiment can show whether listeners hear any residual difference between lenited segments (e.g., [f]<-[p^h]) and the phonetically similar segments ([f]<-[f]). This project used a gating study to investigate when in the word listeners determine which type of sound they are hearing. Preliminary results from 17 native Gaelic listeners indicate that listeners cannot distinguish lenited from phonetically matched consonants (e.g., the two types of [f]) from cues in the consonant itself, but can distinguish both from the unlenited phonologically matched consonants (e.g., [p^h]) very accurately. Listeners become able to distinguish lenited from phonetically matched segments (the two types of [f]) during either the following vowel or the segment after that, depending on what coarticulatory cues with the latter parts of the word are available. Thus, listeners need enough acoustic information to provide lexical disambiguation in order to determine the morphological source of lenited sounds.

2pSCb2. An online investigation of compensation for coarticulation in Mandarin learners of English. David Li (Linguist, Univ. of Southern California, 2133 Sun Ridge Dr., Chino Hills, CA 91709, li.david.c@gmail.com) and Elsi Kaiser (Linguist., Univ. of Southern California, Los Angeles, CA)

Phonological variation can cause lexical ambiguity: The word “run” may sound like “rum” in sentences like “A quick run picks you up” (Gaskell and Marslen-Wilson, 2001) because /n/ is assimilated to /m/ due to the influence of /p/. Previous studies have shown that listeners compensate for coarticulation, perceiving acoustically identical sounds differently in different contexts (e.g., Mann, 1980). We investigated whether native Mandarin speakers with English as a second language (L2) compensate for assimilation in the same manner/to the same extent as native English speakers (L1). Given that there are no Mandarin words with the nasal coda [m], do Mandarin learners of English compensate for phonological variation not present in their L1? We conducted a visual-world eye-tracking study—using English stimuli—to investigate compensation-for-assimilation in L1 English and L1 Mandarin speakers. Identical acoustic sequences were embedded in carrier sentences with fast vs slow speech rates to test interpretation of potentially ambiguous words (cf. rum/run) by L1 and L2 speakers. Eye-movements reveal real-time differences in how L1 and L2 speakers interpret coarticulated speech, with L1 speakers showing signs of rapid compensatory processes. We also discuss how listeners’ interpretation is influenced by other acoustic cues in coarticulated speech.

2pSCb3. Classification of affricate burst place in consonant-vowel contexts in English. Jung-Won Lee, Hong-Goo Kang (Elec. and Electron. Eng., Yonsei Univ., 134 Shinchon-dong, Seodaemun-gu, Seoul 120-749, Korea, Republic of, jaesuk2002@dsp.yonsei.ac.kr), and Jeung-Yoon Choi (Res. Lab. of Electron., Massachusetts Inst. of Technol., Cambridge, MA)

This study investigates characteristics of affricate burst place of articulation compared with the bursts for the three places of articulation for stops (labial, alveolar, and velar) in English. The data comprise consonant-vowel tokens in the TIMIT corpus. To assess which stop place of articulation may be used to simultaneously model affricate bursts, Jensen-Shannon divergence measures are found for probability distributions of acoustic-phonetic features. In addition, we conduct classification experiments using combinations of acoustic-phonetic features and Mel-frequency cepstral coefficients (MFCCs), to see how well affricate burst place is classified using models for the three stop places. The experimental results show that although affricate place is similar to the alveolar place of articulation for stops, a separate post-alveolar place for affricate burst provides a better model. The results suggest that a separate affricate place model will be useful in a feature-based speech recognition system that explicitly detects place of articulation for consonants.

2pSCb4. Effect of consonantal place of articulation on the perception of phonetic voicing in plosives. Viktor Kharlamov and Anna Loukianova (Linguist, Univ. of Arizona, Box 210028, Tucson, AZ 85721, kharlamov@email.arizona.edu)

For plosive consonants, production of phonetic voicing often varies across consonantal places of articulation. Anterior stops (e.g., /b/, /d/) tend to show more glottal pulsing compared to the plosives produced towards the back of the oral cavity (e.g., /g/), which is traditionally attributed to the smaller volume of air passing through the glottis during the closure stage of posterior stops. The current work demonstrates that a similar effect of consonantal place of articulation also exists in perception. Results of a series of identification experiments show that English listeners ($n = 60$) are more sensitive to glottal pulsing in word-final dorsals than labials or coronals. The effect is especially robust when, within each experimental block, listeners are exposed to variability in consonantal place of articulation (i.e., when labial, coronal, and dorsal plosives are blocked together) but not to variability in phonetic vowel duration (i.e., when a given block contains only short or long vowels). This suggests that glottal pulsing is more perceptually salient in posterior plosives, for which voicing is less expected in production, and that the sensitivity to voicing in dorsals is stronger when listeners can do an online comparison of voicing cues across different places of articulation.

2pSCb5. Discrimination between fricatives and affricates pronounced by Japanese native speakers at various speaking rates. Kimiko Yamakawa and Shigeaki Amano (Human Informatics, Aichi Shukutoku Univ., 9 Katahira, Nagakute, Aichi 4801197, Japan, jin@asu.aasa.ac.jp)

Fricatives [s] and affricates [ts] uttered at a normal speaking rate are successfully discriminated with two variables: a rise part duration and a steady + decay part duration [Yamakawa *et al.*, Acoust. Sci. Tech. **33**(3), 154–159 (2012)]. This study examined whether [s] and [ts] uttered at various speaking rates are also well discriminated with these variables. Discriminant analyses with the two variables were performed on [s] and [ts] in word-initial position in a carrier sentence pronounced by eight native Japanese speakers at fast, normal, and slow speaking rates. Discriminant error rates were low for the fast (9.8%, $n = 512$), normal (5.7%, $n = 512$), and slow (13.1%, $n = 512$) speaking rates. However, the error rate was high (22.0%, $n = 1536$) when the three speaking rates were analyzed together. It decreased to 15.7% when the two variables were normalized with an averaged mora duration of the carrier sentence. These results suggest that [s] and [ts] can be discriminated at various speaking rates with a normalized rise part duration and normalized steady + decay part duration. [This work was supported by JSPS KAKENHI Grant Numbers 22720173, 24652087, and 25284080, and by Aichi Shukutoku University Special Research Grant 2011-2012 and Corporate Research Grant 2013-2014.]

2pSCb6. Relationship of training of pitch-pattern reconstruction ability to speech perception in normal-hearing young adults. Kristen M. Cortese (Commun. Disord. & Sci., Rush Univ., 1001 West Cypress Dr., Arlington Heights, IL 60005, kristen_cortese@rush.edu), Stanley Sheft, Valeriy Shafiro, and Derek J. Stiles (Commun. Disord. & Sci., Rush Univ., Chicago, IL)

Previous research has indicated that processing of pitch changes is related to speech perception. Current work investigated the relationship of pitch-pattern training to speech perception in young normal-hearing adults. The training protocol was based on an interactive pattern-reconstruction task in which listeners assembled four or five tones (frequency: 400–1750 Hz, duration: 75–600 ms) to match a random target sequence. Control and experimental groups, of 13 subjects each, were tested. The study included a pretest, three training sessions (experimental group only), and a posttest. Training consisted of three 40-min sessions of interactive pitch-pattern reconstruction. In the pre- and posttest, listeners also reconstructed patterns of sinewave speech, and completed measures of speech perception which included sinewave speech and speech-in-noise tasks. Despite training with only tonal patterns, results showed significant improvement in the ability to reconstruct patterns for both tonal and sinewave-speech stimuli. Significant improvements were obtained for intelligibility of sinewave speech between pre- and posttest, which were greater for the experimental than control group. A greater relationship between results from the intelligibility and pattern-reconstruction conditions post training was also found. Overall, results suggest that the training protocol may benefit speech perception, especially in conditions of degraded speech. [Work supported by NIDCD.]

2pSCb7. Comprehending speech at artificially induced rates. Lucia da Silva (Linguist, Univ. of Br. Columbia, 6335 Thunderbird Crescent Box 137, Vancouver, BC V6T2G9, Canada, lucia@alumni.ubc.ca), Adriano V. Barbosa (Dept. of Electronics, Federal Univ. of Minas Gerais, Belo Horizonte, Brazil), and Eric Vatikiotis-Bateson (Linguist., Univ. of Br. Columbia, Vancouver, BC, Canada)

This study confirms and extends Fairbanks, Guttman and Miron's study (1957), where comprehension of time-compressed (2:1), recorded speech was improved when played back twice. Specifically, we examine the perception of pitch corrected and non-pitch corrected compressed speech when presented twice in succession and twice with delays of minutes between stimulus presentations. Additionally, we investigate whether visual speech information aids or hinders comprehension by comparing fast rate effects when presented acoustically and audiovisually. A motivation for this study is the increasing dependency on web-based multimedia recordings for knowledge transfer.

2pSCb8. Perceptual interactions among components of a spectral-domain voice source model. Marc Garellek (Linguist. Dept., UCSD, 9500 Gilman Dr. #0108, La Jolla, CA 92093-0108, mgarellek@ucsd.edu), Robin A. Samlan, Jody Kreiman, and Bruce Gerratt (Dept. of Head and Neck Surgery, UCLA, Los Angeles, CA)

A psychoacoustic model of the source spectrum has been proposed in which source contributions to overall voice quality can be quantified by four spectral slope components: H1-H2 (the amplitude difference between the first and second harmonics), H2-H4, H4-2000 Hz (i.e., the harmonic nearest to 2000 Hz), and 2000–5000 Hz. The natural variability of these components has been described, along with the just noticeable differences (JNDs) for each component. The goals of this study are to identify how perceptual sensitivity to each component slope varies as a function of the adjacent slope(s). The JNDs were obtained for stimuli based on synthetic copies of one female voice. The stimuli were manipulated so that spectral slope varied in 0.5 dB increments (1 dB for 2000–5000 Hz) for a particular component at "high" and "low" values of the adjacent slopes. Thirty-three listeners completed an adaptive up-down paradigm. Preliminary results suggest that sensitivity to a particular spectral component varies as a function of some adjacent components more than the other. Furthermore, sensitivity varies based on whether the adjacent component has a high or low slope. These interactions among spectral components will be interpreted with respect to the variability in spectral configuration observed in 144 voices.

2pSCb9. Lexical segmentation of speech from energy above 5 kHz. A. Davi Vitela (Univ. of Arizona, 1131 E. 2nd St., Tucson, AZ 85721, adv1@email.arizona.edu), Brian B. Monson (DUKE-NUS, Singapore, Singapore), and Andrew J. Lotto (Univ. of Arizona, Tucson, AZ)

Research in the field of speech perception has traditionally focused on the acoustic information or cues present in the frequency region below 5 kHz, thus ignoring high frequency energy (HFE). A recent series of studies, however, demonstrated that listeners could determine the mode of production (speech or singing) and further, could identify what was being spoken or sung from the HFE alone and with a speech-shaped masking noise in the lower frequencies. This begs the question as to what types of information listeners are extracting to guide their perception. The current study examined the ability of listeners to transcribe short semantically-unpredictable but syntactically-well-formed spoken phrases that were high-pass filtered at 5.6 kHz. Of particular interest was the ability of some listeners to correctly determine the placement of word boundaries even without the availability of low-frequency information. These findings add to a growing literature on the linguistically relevant information present in higher frequency regions. Results will be framed within current theories of acoustic signatures to lexical segmentation. [Work supported by NIH-NIDCD.]

2pSCb10. Prosodic disambiguation of Korean relative clause attachments. Younah Chung (Linguist., UCSD, UCSD Linguist Dept., 9500 Gilman Dr., La Jolla, CA 92093-0108, yachung@ucsd.edu) and Chongdok Kim (Inst. of Lang. and Information Studies, Yonsei Univ., Seoul, South Korea)

Korean relative clauses do not have to be adjacent to the substantives they modify; genitive noun phrases can interpose between the relative clause and the substantive. Thus, the syntactic analysis of a given relative clause is inherently ambiguous, such that the relative clause can modify either the head noun of a genitive noun phrase or the complex noun phrase as a whole. However, in spoken language, one rarely finds this type of ambiguity even though the relative clause could modify either of the two nouns. This suggests that speakers use prosody to convey intended meaning, with different prosodic cues associated with different syntactic boundaries, thus, providing information necessary for disambiguation. We examined the clarification of surface structure ambiguity in Korean relative clause attachments via prosody. The survey confirmed that native speakers think relative clauses are surface ambiguous. The acoustical analysis revealed that prosodic disambiguation consists of pause insertion, pitch raising, and final lengthening at the syntactic boundary. Regarding the relative importance of different prosodic measures used by speakers, we discovered that pitch raising was a more powerful measure than final lengthening in prosodic disambiguation, and that pausing played the least important role in the parsing of an ambiguous relative clause.

2pSCb11. The effect of phonetic orthography on the perception of Mandarin syllables. Yu-Jung Lin, Chung-Lin Yang (Linguist., Indiana Univ., 800 N Union St. Apt. 405, Bloomington, IN 47408-2230, lin41@indiana.edu), and Chien-Jer Charles Lin (East Asian Lang. and Cultures, Indiana Univ., Bloomington, IN)

It has been found that vowels have perceptual advantage over tones (e.g., Ye and Connie, 1999) and that phonetic orthographic information can be activated in speech perception (e.g., Frauenfelder, Segui, and Dijkstra, 1990). In the present study, we used a tone-vowel detection task where Zhuyin (Taiwanese) and Pinyin users (Mainland Chinese) were asked to monitor Mandarin syllables containing [i4] to test how tonal and vowel information are processed in syllable monitoring, and to examine if Zhuyin and Pinyin users' responses are affected by their phonetic orthographic systems. The results demonstrated that in both groups, vowel-mismatched syllables (e.g., bu4) were rejected faster than tone-mismatched syllables (e.g., bi2). Moreover, the RT of rejecting vowel-mismatched syllables and double-mismatched syllables (e.g., bu2) were similar, indicating that as long as vowel was mismatched, tonal information caused little difference in RT when rejecting the mismatched syllables. This finding was consistent with Ye and Connine (1999). Furthermore, both groups resort to their phonetic orthography in simple syllables (e.g., bi4, zhi4). However, in complex syllables (e.g., biao4), different patterns were observed between the two groups, possibly due to the factors such as how tones are marked in their phonetic orthography or how they learned the orthography.

2pSCb12. Perceiving politeness from speech acoustics alone: A cross-linguistic study on Korean and English. Bodo Winter (Cognit. and Information Sci., Univ. of California, Merced, 3681 San Jose Ave., Apt. 4, Merced, CA 95348, bodo@bodowinter.com), Lucien Brown, Kaori Idemaru (East Asian Lang. & Literatures, Univ. of Oregon, Eugene, OR), and Sven Grawunder (Linguist., Max Planck Inst. for Evolutionary Anthropology, Leipzig, Germany)

Politeness is a crucial aspect of everyday speech communication; however, there are to date only few acoustic studies on this topic. Winter and Grawunder (2012) showed that for Korean speakers, politeness is reflected in pitch, intensity, voice quality and speaking rate. Here, we extend this production study by investigating whether Korean and English listeners can perceive the intended politeness of short Korean utterances based on speech acoustics alone. In two experiments with a total of 47 English and 30 Korean listeners, we found that both groups can detect the intended politeness purely based on the phonetic qualities of speech. In one experiment, accuracy was low (Korean: 58%, English: 53%) because speakers heard multiple voices in a randomized fashion, not allowing them to familiarize with any particular voice. In a design that was blocked by speaker voice, accuracy was higher (Korean: 70%, English: 58%), showing that vocal politeness can be used as a cue when the voice is known. This shows that politeness is not only expressed by honorific lexical forms commonly employed in Korean, but also by speech acoustics. It is remarkable that English speakers performed above chance at all, pointing to cross-linguistic regularities in how politeness is expressed vocally.

2pSCb13. The role of vowels in determining a male speaker's sexual orientation. Erik C. Tracy (Psych., Univ. of North Carolina Pembroke, PO Box 1510, Pembroke, NC 28372, erik.tracy@uncp.edu)

Prior research (Tracy and Satariano, 2011) found that listeners primarily use the vowel in a monosyllabic word to determine the speaker's sexual orientation. Given this result, if listeners were presented utterances that contained multiple vowels, then the accuracy of their sexual orientation ratings should increase. In Experiment 1, listeners were presented with trisyllabic words ("authentic"). They heard the first syllable (/ə/), the first two syllables (/əθɛn/), and the entire word (/əθɛntɪk/). Sexual orientation ratings were the most accurate for the entire word and the least accurate for the first syllable. These results were replicated in experiment 2, which used only bisyllabic words. An items analysis from both experiments revealed that a syllable that contained more phones resulted in higher accuracy ratings compared with a syllable that contained fewer phones. For example, listeners' sexual orientation ratings were more accurate when they heard the first syllable of "lavender" compared to the first syllable of "authentic." These findings suggest that when the number of vowels is kept constant, listeners' responses are more accurate if the utterance contains additional phones.

2pSCb14. Acoustic features mediating height estimation from human speech. John Morton (Psych., Washington Univ., 2309 Laurenwood Dr., Chesterfield, MO 63017, jrmhvc333@yahoo.com), Mitchell Sommers (Psych., Washington Univ., St. Louis, MO), Steven Lulich (Linguist, Indiana Univ., Bloomington, IN), Abeer Alwan, and Harish Arsikere (Eng., UCLA, Los Angeles, CA)

The current experiment was aimed at providing the first direct evidence regarding the role of subglottal resonances in height discrimination. Past research has investigated whether or not listeners are able to discriminate which of two talkers is taller (Rendall *et al.*, 2007), but has not established what parameters of the acoustic speech signal listeners use to distinguish speaker height. In the current study, we examined the role of subglottal resonances in height discrimination. Subglottal resonances generally refer to resonances of the lower airways starting in the lungs and terminating at the glottis. Subglottal resonances would be a good candidate for use in height estimation because they remain relatively stable across vowels and are relatively unaffected by other factors influencing the supra-laryngeal vocal tract; instead they are influenced nearly exclusively by the lower airway acoustic

properties. Listeners participated in two tasks, the first task was a two-alternative forced choice (2AFC) height discrimination test, in which listeners heard sentences produced by two talkers of the same gender and were asked to determine which of the two was the tallest. The second task involved listening to five talkers (all same gender) sentences, and then ranking those individuals from tallest to shortest. Findings indicate that listeners are able to discriminate and rank speakers heights better than chance, and the role of subglottal resonances will be reported.

2pSCb15. Lexically biased perceptual adaptation. Michael McAuliffe (Linguist., Univ. of Br. Columbia, Totem Field Studios, 2613 West Mall, Vancouver, BC V6T 1Z4, Canada, mcauliff@interchange.ubc.ca)

Listeners are able to rapidly shift their acoustic-phonetic categories in the face of speaker-specific variation. However, an open question is how top-down expectations can influence this perceptual adaptation. Under Adaptive Resonance Theory [Grossberg (2003)], if an acoustic token provides initial activation of a linguistic category, top-down expectations about that category can boost the resonance and lead to activation of the category even with an ambiguous token. Ambiguous tokens in positions with stronger expectations are hypothesized to lead to less perceptual adaptation than in positions with lower expectations. To test this hypothesis, two groups of participants will be exposed to 10 ambiguous s-ʃ sounds during a lexical decision exposure task. The groups differ in where in the word these sounds will occur, either in the onset or coda of CVC words. Lexical bias, as a form of top-down expectations, is stronger for ambiguous coda sounds [Pitt and Szostak (2012)]. I predict that in a s-ʃ categorization task following exposure, the coda group will differ less than the onset group from a control group that only completes the categorization task, indicative of less perceptual adaptation.

2pSCb16. Degraded word recognition in isolation vs a carrier phrase. Kathy M. Carbonell and Andrew J. Lotto (Speech Lang. & Hearing Sci., Univ. of Arizona, 1131 E 2nd St., Tucson, AZ 85721, kathyc@email.arizona.edu)

Recognizing a spoken word presented in isolation is a markedly different task from recognizing a word in a carrier phrase. The presence of a carrier phrase provides additional challenges such as lexical segmentation but also provides additional information relevant to word recognition such as speaking rate and talker-specific spectral characteristics. The current set of studies is part of an attempt to determine how target word recognition differs in isolation versus in a carrier phrase. In an initial experiment, a set of 220 spoken CVC words were noise-vocoded (6 channel) and presented to listeners either in isolation or following a noise-vocoded carrier phrase—"The next word on the list is…" The target words were transcribed in each condition and scored for initial consonant accuracy and overall word accuracy. Despite the lack of semantic or syntactic information provided by the carrier phrase accuracy for both word and consonant recognition were much higher in the carrier phrase context. A second experiment, used different talkers for the carrier phrase and target word. A smaller but significant benefit over isolation was present for these mixed talker stimuli suggesting that the benefits of the carrier phrase include, but are not limited to, talker-specific information.

2pSCb17. Differences in the recognition of careful and casual speech. Meghan Sumner, Jeremy Calder, Annette D'Onofrio (Dept. of Linguist, Stanford Univ., Margaret Jacks Hall, Bldg. 460, Stanford, CA 94305, sumner@stanford.edu), Kevin B. McGowan (Dept. of Linguist, Stanford Univ., Stanford, Michigan), and Teresa Pratt (Dept. of Linguist, Stanford Univ., Stanford, CA)

Previous work in spoken word recognition and speech perception has shown two seemingly conflicting patterns. While some studies have shown a processing benefit for more frequent word variants (i.e., in a casual speech mode), others have found a benefit for more canonical word forms (i.e., in a careful speech mode). This study aims to reconcile these findings, proposing that different types of processing apply to each speech mode –top-down processing for casual speech, and bottom-up for careful speech. Listeners in an auditory priming task heard natural (non-spliced) sentences spoken in either a careful or casual speech mode. The final word of the auditory prime was either semantically predictable from the preceding sentence context or unpredictable. After the audio prime, listeners responded in a lexical decision task to a visual probe: either the final word heard in the prime, an unrelated word, or a nonword. Preliminary results suggest that, regardless of speech style, reaction times are faster for related targets in the semantically predictable conditions than for unrelated targets. Crucially, responses to the target word in the careful condition are delayed compared to casual speech for semantically unpredictable sentences. The implications for the apparent paradox in previous results will be discussed.

2pSCb18. Use of acoustic cues in the perception of complex syllable structure. Goun Lee (Linguist., Univ. of Kansas, 1541 Lilac Ln., Blake Hall, Rm. 427, Lawrence, KS 66045-3129, cconni@ku.edu)

This study examines second/foreign language learners' (L2ers') use of acoustic cues in the perception of syllable structure (onset clusters) that does not exist in the native language (L1, here, Korean). It investigates whether the target-like perception of syllable structure is facilitated by VOT ([stɪn] vs [sətɪn]) and partial devoicing of the liquid ([plim] vs [pəlim]) as compared to when either cue is absent ([blint] vs [bəlint]). This study also examined whether production errors can be predicted by perception errors. Twelve Korean-speaking English L2ers and 12 native English speakers completed AXB and production tasks in which they listened to and produced nonce words in the above three conditions. The AXB results showed significant effects of L1 and condition, but no interaction between the two. Participants were most accurate on the condition with VOT (English: 98%, Korean: 85%), followed by partial devoicing of the liquid (English: 96%, Korean: 79%) and without either cue (English: 94%, Korean: 74%). L2ers' perception and production patterned alike, but no correlation was found between the two. These findings suggest that acoustic cues like VOT and partial devoicing of the liquid facilitate L2ers' perception of new syllable structure, but do not play a direct role in its production.

2pSCb19. The perception of English [h] and [ɹ] by Brazilian Portuguese speakers. Denise M. Osborne (Univ. of Arizona, 814 E. 9th St. Apt 7, Tucson, AZ 85719, dmdcame@hotmail.com)

Brazilian Portuguese (BP) learners of English have difficulties in differentiating between initial English [h] and [ɹ] (Zimmer, Silveira, and Alvers, 2009). This study investigates how speakers who have BP as their L1 and English as their L2 perceive the phonetic distance of English [h] and [ɹ], and how they and monolingual BP speakers map these phonemes onto BP sound categories. 32 native BP learners of English participated in three consecutive experiments: AXB Discrimination, Identification, and Assimilation Tests. In addition, 18 monolingual BP speakers participated in the same Assimilation test. Significant effects in ANOVAs on the three experiments support the results. Beginners and intermediates were able to hear the distinction acoustically. Only intermediates, however, used the distinction to identify English words. Both monolingual BP speakers and intermediates assimilate English [h] primary to BP double <rr> (reflecting the dialectal and allophonic variation of the BP rhotic sounds). Beginners showed a failure to assimilate L2 sounds to their L1 BP categories. Comparison across these experiments shows that, at some stages of learning, BP speakers experience difficulty in evaluating these L2 sounds relative to their L1 inventory and in using the L2 distinction at the lexical level.

2pSCb20. Effects of hearing loss and linguistic context for phoneme and word recognition in noise. Adam Svec, Benjamin Munson, and Peggy B. Nelson (Dept. of Speech-Language-Hearing Sci., Univ. of Minnesota, 164 Pillsbury Dr. SE, Minneapolis, MN 55455, svecx002@umn.edu)

This project measured whole-word and phoneme recognition in normally hearing (NH) and hearing-impaired (HI) participants listening in steady-state noise. Stimuli were modeled after those used in Olsen *et al.* *[EarHear 1997]* including 10-word lists of consonant-vowel-consonant (CVC) monosyllables containing 10 vowels and 20 consonants. Words were presented in isolation and in contextually correct sentences. Speech-shaped noise was presented at a range of signal-to-noise ratios, from -9 to $+3$ dB SNR for NH listeners, and from -6 to $+6$ dB SNR for HI listeners. Results suggested that HI listeners needed an increased proportion of components (p_p) to recognize a whole word (p_w) relative to their NH counterparts (e.g., a higher j value) when stimuli were presented at unfavorable SNRs. For -3 and 0 dB SNR conditions, j factors for NH listeners (0 dB SNR: $j = 2.17$) were consistently lower than those for HI listeners (0 dB SNR: $j = 2.52$). However, for the 3 dB SNR conditions, j factors were higher for NH listeners ($j = 3.06$) than those for HI listeners ($j = 2.58$). This suggests an unexpected interaction between hearing acuity, background noise, and the recognition of phonemes versus whole words. [Research supported by DC008306 to Peggy Nelson.]

2pSCb21. Sociolinguistic perceptions of Californian /æ/ backing. Annette D'Onofrio (Linguist., Stanford Univ., 450 Serra Mall, Bldg. 460, Stanford, CA 94305, annetted@stanford.edu)

This paper investigates perceptions of a feature of the "California Vowel Shift" (Eckert, 2004): the backing of the TRAP vowel. The present study probes whether or not Californian and non-Californian listeners exhibit knowledge of TRAP backing's dialectal patterning in perception. An American English speaker's TRAP vowel was manipulated to create a 9-step continuum from /a/ to /æ/ in a sentence frame. 144 native U.S. listeners heard one of these steps, accompanied by orthographic information about the word spoken (e.g., the word was explicitly given as "blocked" or "blacked"). Listeners provided social categorizations of the speaker. Listeners were more likely to categorize the speaker as Californian when presented with TRAP orthography (e.g., "blacked") paired with a token on the continuum that was closer to /a/. 60 listeners participated in a phoneme categorization task using the same manipulated stimuli. When listeners were told that a speaker was from California, they were more likely to place the boundary between "blacked" and "blocked" closer to the /a/ end of the continuum than if told the speaker was from elsewhere. Together, these experiments indicate that listeners exhibit awareness of TRAP-backing's dialectal association with California, the top-down knowledge of which can affect lexical categorization.

2pSCb22. Detecting speaker change in background audio streams. Hilary Toh, Pei Xuan Lee (Victoria Junior College, 23 Li Hwan View, Singapore, Singapore 556913, Singapore, toh.si.yin.hilary.2011@vjc.sg), Boon Pang Lim, and Nancy F. Chen (Human Lang. Technol., Inst. for Infocomm Res., Singapore, Singapore)

The cocktail party effect refers to the ability of humans to selectively focus on one speech stream among multiple human-generated background conversations and noise sources. This ability is influenced by various factors including directionality, speaking rates, different speech accents and speaking styles. It is displayed most clearly as a binaural effect, requiring listeners to be able to identify and differentiate between multiple sound sources simultaneously. Past studies [Cherry, J. Acoust. Soc. Am. **25**(5), 975–979 (1953)], have shown that listeners can detect a change of speaker voice from male to female in unattended audio streams, while focusing their attention on repeating speech in an attended audio stream. Our work further investigates this by focusing on speaker change across the same gender: a more difficult task given a smaller variance in pitch across speakers. Experiments were conducted on 24 listeners and findings suggest that they are able to recognize speaker changes of the same gender. Two factor ANOVA shows an interaction between listener and speaker gender ($F(1, 22) = 7.688$, $p = 0.01$), suggesting listeners perform better when detecting speaker changes within their own gender.

2pSCb23. A word to the eyes: Visual cues benefit lexical segmentation in noise. Kate Helms Tillery, Sarah J. Cook, Rene L. Utianski, Julie M. Liss, and Michael F. Dorman (Dept. of Speech and Hearing Sci., Arizona State Univ., Coor Hall 2211, P.O. Box 870102, Tempe, AZ 85287-0102, kate.helms-tillery@asu.edu)

In English, most word-initial syllables are stressed. Listeners use these syllable strength cues to identify word boundaries in degraded acoustic conditions. This is evident in the types of lexical boundary errors they make when tasked with parsing a continuous stream of degraded speech: word boundaries are more often inserted before strong syllables than before weak syllables. Listeners also use visual cues from the talker's face to glean both phonemic and prosodic speech information in degraded listening conditions. While the benefits of lip-reading have received much attention, the role of visual cues in lexical segmentation remains largely unexplored. The present study examined the effect of auditory-visual cues on lexical boundary decisions. Normal-hearing listeners identified target phrases degraded by multi-talker babble. Responses in auditory-only and auditory-visual conditions were analyzed for percent words correct and lexical boundary error type. Results indicate large inter-individual variability, but overall an increase in word identification accuracy and a decrease in lexical boundary errors in the auditory-visual condition. Further, some listeners made a greater proportion of lexical boundary insertions before strong syllables, suggesting that the addition of visual cues increased their use of syllable strength to identify word boundaries. Implications for clinical populations will be discussed.

2pSCb24. Talker familiarity effects on speech-in-speech perception. Angela Cooper and Ann R. Bradlow (Linguist., Northwestern Univ., 2016 Sheridan Rd., Evanston, IL 60208, akcooper@u.northwestern.edu)

Talker familiarity can facilitate the extraction of linguistic content from speech signals embedded in broadband noise; however, relatively little research has investigated the impact of talker familiarity with competing speech in the background. This study explores the effects of familiarity with the target or competing talker in speech-in-speech perception. Listeners were first familiarized with and trained to identify three female voices. They then completed a sentence recognition task in the presence of 1-talker babble. Familiarity with either the target or background talker was manipulated in separate conditions. Results revealed significantly better sentence recognition for familiar relative to unfamiliar target talkers in the presence of an unfamiliar background talker; however, sentence recognition with an unfamiliar target talker did not differ depending on background talker familiarity. Thus, while listeners were able to capitalize on familiarity with a talker's voice to aid target speech recognition, familiarity with the competing talker was neither facilitative nor inhibitory. This suggests that the influence of talker familiarity is limited to the actively attended stream. This stands in contrast to other aspects of the unattended stream which have been shown to exert an influence on speech-in-speech recognition including language and semantic content of the background speech.

2pSCb25. Investigating the perception of relative speaker size using synthetic talkers. Santiago Barreda (Physiol., Univ. of Arizona, Dept. of Linguist., Edmonton, AB T6G 2E7, Canada, sbarreda@ualberta.ca) and Terrance M. Nearey (Linguist., Univ. of AB, Edmonton, AB, Canada)

Several previous experiments have investigated the perception of speaker size by presenting listeners with acoustic stimuli that differ in average f0 and/or apparent vocal tract length (i.e., higher formant frequencies overall), and asking listeners to make judgments of relative or absolute speaker size. Typically, these experiments use stimuli with a fixed phonetic content so that the acoustic characteristics of the stimuli may be compared directly. In this experiment, listeners were presented with pairs of vowels produced by synthetic speakers with different apparent vocal tract lengths and the same f0, and were asked to make judgments of relative size. However, listeners were presented with either the same, or different vowels produced by the two speakers. In some cases, differences associated with varying vocal tract lengths were in conflict with differences arising from the formant patterns associated with the differing vowel categories (e.g., lower F1 and F2 for /u/ vs /e/). Results suggest that judgments of relative size are affected by both the vowel categories presented and the apparent vocal tract lengths of the synthetic speakers. That is, more than a simple (category-corrected) vocal tract length estimate is involved in making size judgments for an unknown speaker.

2pSCb26. The effects of listener biases on speech intelligibility. Jamie Russell (Linguist., Univ. of Br. Columbia, 9320 Dixon Ave., Richmond, BC V6Y1E7, Canada, jamie.russell@alumni.ubc.ca) and Molly Babel (Linguist., Univ. of Br. Columbia, Vancouver, BC, Canada)

Race, gender, and age have traditionally been considered the three big social categories perceivers attend to. Recent research has highlighted how

perceivers, listeners in this case, use the acoustic-phonetic information from spoken language to evaluate and categorize individuals. For example, studies have shown that different dialects elicit varying judgments of intelligence, friendliness and attractiveness. Such biases have implications regarding our assessment of individuals, be it in the clinic or the courtroom. While we know that our perception of speech can trigger biases, it is important to explore the opposite—whether our biases affect our perception of speech. This study aims to contribute to the knowledge base of the latter by examining the effects of listener biases on speech intelligibility. Listeners will be presented with sentences, independently controlled for clarity, produced by model talkers of different dialects and backgrounds. This auditory stimuli will be accompanied by an image or description of the talker as a method of eliciting or constructing a preconceived linguistic bias. After each sentence, participants will be asked to type out what they heard. Analyses will examine the proportion of correct words, confidence ratings, response time, and other measures exploring participants' ratings of vocal qualities.

2pSCb27. Speaker normalization in noisy environments using subglottal resonances. Harish Arsikere and Abeer Alwan (Elec. Eng. Dept., Univ. of California, Los Angeles, 56-125B Eng. IV Bldg., 420 Westwood Plaza, Los Angeles, CA 90095-1594, hari.arsikere@gmail.com)

This work investigates the use of subglottal resonances (SGRs) for speaker normalization in noisy environments. Based on our previous work, a noise-robust algorithm is developed for estimating the first three SGRs from speech signals; it achieves robustness by factoring the short-term (or local) signal-to-noise ratio into the estimation process. The SGR estimates provided by this algorithm are refined by applying maximum-likelihood (ML) corrections, and are used in a non-linear frequency-warping technique that we recently developed. This SGR-based normalization (SN) scheme is evaluated on the AURORA-4 database in clean and noisy conditions. Using power-normalized cepstral coefficients (PNCCs) as front-end features, SN reduces the average word error rate by 8.7% relative to ML-based vocal-tract length normalization (VTLN). A fast version of SN (without ML corrections of SGR estimates) is also found to outperform VTLN (by 5.9% relative); it is computationally less complex than VTLN and hence a potential alternative for real-time applications.

Session 2pSP

Signal Processing in Acoustics, Underwater Acoustics, Structural Acoustics and Vibration, and Animal Bioacoustics: Defense Applications of Acoustical Signal Processing

R. Lee Culver, Cochair
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Brian G. Ferguson, Cochair
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Chair's Introduction—1:25

Invited Paper

1:30

2pSP1. Defense applications of acoustic signal time delay estimation. Brian G. Ferguson and Kam W. Lo (Maritime Div., DSTO, PO Box 44, Pyrmont, NSW 2009, Australia, Brian.Ferguson@dsto.defence.gov.au)

Measurements of the differences in the times of arrival (or time delays) of a signal at spatially-separated sensors can be used to provide localization information about a source. For continuous broadband signals, the time delay for a given pair of sensors is estimated using the generalized cross-correlation method. The time delay estimate corresponds to the time lag at which the cross-correlation function attains its maximum value. The variance of the time delay estimates depends on the signal-to-noise ratio, signal bandwidth, and integration time. Errors in the source localization parameters depend on the variance of the time delay estimates as well as on the source-sensor geometry, stationarity of the sound propagation medium, uncertainty in the actual sensor positions, and the presence of multipath arrivals. Numerous practical examples of time delay estimation and the instantaneous localization of sources of military interest are presented for acoustic sensors deployed on land and under water. Also, localization performance is observed to degrade when the sound propagation medium becomes nonstationary, the direct path and multipath arrivals are unresolvable, or the differential Doppler effect is significant. Finally, the results of source motion parameter estimation using sequences of time delay estimates from pairs of sensors are presented.

Contributed Papers

1:50

2pSP2. Passive acoustic localization of small aircraft. Alexander Sedunov, Alexander Sutin, Hady Salloum (Stevens Inst. of Technol., 711 Hudson St., Hoboken, NJ 07030, asedunov@stevens.edu), and Nikolay Sedunov (Stevens Inst. of Technol., Jersey City, NJ)

Stevens Institute of Technology has built the Acoustic Aircraft Detection (AAD) system for the detection, tracking and classification of Low Flying Aircraft (LFA). LFA may be of concern as they have been used for illicit operations. The AAD consists of several nodes deployed in a wide area, where each node acquires signals from a cluster of five microphones. The calculation of the cross-correlation function of acoustic signals received by various microphone pairs is applied for finding the direction of the signal arrival. Fusion of time difference of arrival estimates from several pairs of acoustic sensors has resulted in the improvement of angle estimation accuracy. Triangulation of the direction of arrival estimates from two or more nodes was applied for determining the target position and altitude. Kalman filtering was used for smoothing the target tracks and decreasing the localization uncertainties. Software for predicting the performance was developed and was used to inform sensor placement in the field tests. The field tests were conducted with various kinds of LFA—single-engine, helicopters, and ultralights. Comparison of the acoustic tracking results with the GPS ground truth showed that the errors are similar to the theoretical predictions. [This work was funded by DHS S&T.]

2:05

2pSP3. Automated acoustic detection and classification of small aircraft. Yegor Sinelnikov, Alexander Sutin, Alexander Sedunov, and Hady Salloum (Stevens Inst. of Technol., 711 Hudson St., Hoboken, NJ 07030, asedunov@stevens.edu)

Automated acoustic detection and classification of low flying aircraft can be used for the prevention of illicit operations. In this presentation, we review past techniques and a new acoustic classification and detection technique. The sound detection algorithm is based on locking in narrow frequency components in the spectra of the recorded aircraft sound. The classification algorithm uses tonal components in the spectral and cepstral domains using discrete time window peak picking. A set of chosen classifiers include: fundamental (pitch) frequency extracted by the cepstral analysis, frequency of tonal components with the maximal amplitude, power spectral estimate in various frequency bands, and number of peaks in predefined frequency and frequency windows. Various small single engine aircraft, ultralights, and helicopters are acoustically detected and classified as they approach the sensor. The algorithm enables simultaneous target detection, reduces noise sensitivity, and minimizes classifier feature space, while maintaining good classification separation. The acoustic classification algorithm was incorporated into the Acoustic Aircraft Detection system developed by Stevens. This combination allowed automated Doppler correction of the harmonic lines, the optimal part of the aircraft signal selection, and establishing classification merits based on aircraft track. [This work was funded by DHS S&T.]

2:20

2pSP4. Coordinated optimization of stationary and moving sensors with scenario-specific coverage and cost goals. Sergey Vecherin, D. Keith Wilson (Signature Phys. Branch, U. S. Army ERDC-CRREL, 72 Lyme Rd., Hanover, NH 03755, Sergey.N.Vecherin@usace.army.mil), and Chris Pettit (Aerosp. Eng. Dept., U. S. Naval Acad., Annapolis, MD)

The problem of simultaneously optimizing coverage among stationary sensors placed on the ground and moving sensors on aircraft is considered. The formulation extends a previously developed algorithm for optimizing placement of ground sensors subject to signal propagation effects and terrain-dependent coverage preferences. To this end, candidate aircraft routes should be characterized in terms of cost and coverage. Cost can reflect a variety of disincentives, for example, the probability that an aircraft can be heard on the ground. Coverage reflects the area of regions with probability of detection higher than a specified threshold. The extended algorithm can be applied to a variety of planning scenarios, such as determining an optimal combination of routes for multi-aircraft operations, optimization of aircraft routes to supplement ground-sensor coverage, optimization of ground sensors to cover blind spots of aircraft coverage, and simultaneous optimization of ground sensors and aircraft coverage. An illustrative problem of routing an unmanned aircraft system (UAS) to provide surveillance around a roadway while minimizing aircraft audibility at specified locations on the ground is considered in detail.

2:35

2pSP5. Performance comparison of distributed and centralized systems based on information. Tsih C. Yang (College of Marine Sci., Nat. Sun Yet-sen Univ., 70 Lien Hai Rd., Kaohsiung 804, Taiwan, tsihyang@gmail.com)

Distributed systems using many sensors widely distributed over a large area present an alternative way for target detection and environmental sensing. The diversity of opportunities for detection by widely distributed sensors seems attractive but the signals received on distributed sensors are usually not coherent due to the wide separations between the receivers. Hence, conventional approaches based on signal coherence/gain no longer apply. In this paper, information theory will be used to compare their performance. Treating the target radiated signal as a communication signal, transmitting continuous Gaussian-distributed alphabets, the Shannon channel capacity yields the maximum information that the receivers can learn about the (target) transmitted signal. The channel capacity can then be used as a metric to compare the performance of various sensor systems in the ideal case. Matched tracking processing is introduced to motivate a capacity-based detector and detection capacity. Based on that, it is found that the distributed systems can achieve in principle an area of coverage two to three times larger than that of a centralized system under the right conditions, and the area of coverage by the entire system can be significantly larger than the sum of detection areas of individual nodes.

2:50–3:05 Break

3:05

2pSP6. Acoustic signatures of small boats measured in controlled lake conditions. Alexander Sutin, Michael DeLorme, Hady Salloum, Alexander Sedunov, Nikolay Sedunov, Robert Weiss, Mikhail Tsionskiy, and Howard Goheen (Stevens Inst. of Technol., 711 Hudson St., Hoboken, NJ 07030, asutin@stevens.edu)

Measurements were conducted under controlled conditions at Lake Hopatcong, NJ, to compare acoustic signatures of various small boats. The tested boats included: an outboard-driven Panga, a multiple-outboard driven go-fast boat, an electric drive vessel, a personal watercraft (PWC), and two small outboard-driven rigid hull inflatable boats (RIBs). Stevens Passive Acoustic Detection System (SPADES) was used for acoustic measurements and the specially developed "Portable Vessel Data System" (PVDS) was used to record vessel position, speed, shaft RPM, and vessel orientation. Vessel acoustic Source Level measurements were conducted by comparing the recorded vessel acoustic signals with a signal generated by a calibrated emitter from the same point. Dependencies of acoustic signatures on boat speed and loading were also investigated. For several vessels, the Source Level decreased when its speed increased. Analysis of tonal components in the vessels acoustic signatures with Detection of Envelope Modulation on Noise (DEMON) allowed to determine the number of engines per boat and even the gear ratio for transformation of shaft rotation to the propeller rotation. [This work was supported by DHS S&T.]

3:20

2pSP7. Active–passive acoustic system for underwater port protection. Hady Salloum (Stevens Inst. of Technol., Hoboken, NJ), Andrew Meeecham (Sonardyne, Inc., Newport, RI), and Alexander Sutin (Stevens Inst. of Technol., 711 Hudson St., Hoboken, NJ 07030, asutin@stevens.edu)

Active and passive acoustic systems may be used for port protection against threats from divers, free swimmers, Unmanned Underwater Vehicles, etc. Currently, underwater port protection is provided by Diver Detection Sonars (DDS), and Sonardyne has developed the market-leading active system, Sentinel. Stevens' scientific research in passive acoustics led to the development of algorithms for threat detection that were realized in the Stevens Passive Acoustic Detection System (SPADES). Sonardyne and Stevens have formed a partnership to investigate a combined active/passive system that will leverage the functionality of existing sensors to design, develop, and produce a system with superior capabilities and performance as opposed to a single system. The combined system provides detection not only for underwater targets, but also for surface vessels. Possible configurations include: (1) a scenario where acoustic pulses emitted by the Sonardyne sonar and reflected by the target can be received and processed by SPADES; (2) an "acoustic fence," whereby the detection of targets occurs as they cross a line between the transmitter and receiver. This configuration takes advantage of the significant increase in target scattering strength in the forward direction versus that from the backscatter; and (3) a fusion between detections and/or tracks from the passive and active sensors to provide an increased probability of detection and reduced probability of false alarm.

3:35

2pSP8. Practical applications of track segment association algorithms to an active sonar network for underwater port surveillance. Patrick Edson and Peter J. Stein (Scientific Solutions, Inc., 99 Perimeter Rd., Nashua, NH 03063, pedson@scisol.com)

Acoustic detection and tracking of small (swimmer-size) targets while minimizing the incidents of false alerts can be challenging in a shallow, multipath, high-clutter harbor environment. One common problem involves track intermittency where periods or locations of high clutter or multipath interference inhibit detection and cause tracks to fail and restart. This paper describes the results of applying track segment association and related algorithms from airborne and ground radar applications [*e.g.,* Zhang and Bar-Shalom, IEEE Trans. Aerosp. Electron. Syst. **47**(3), 1899–1914 (2011)] to an active sonar network for underwater port surveillance [Edelson *et al.*, J. Acoust. Soc. Am. **129**(4), 2598 (2011), Stein *et al.*, J. Acoust. Soc. Am. **121**(5), 3084 (2007)] using real-world swimmer and synthetic target data.

3:50

2pSP9. Adaptive processing for mitigation of biologically induced active-sonar clutter and reverberation. Jason E. Summers (Appl. Res. in Acoust. LLC, 1222 4th St. SW, Washington, DC 20024-2302, jason.e.summers@ariacoustics.com)

Scattering from biological entities can be a significant source of reverberation and clutter for active sonar systems operating in deep or shallow water. Resonant scattering from the swim bladders of fish is a dominant source of this biologically induced reverberation and clutter for midfrequency active sonar (1–10 kHz), which can obscure targets, result in false targets, and otherwise overload operators of tactical systems. Though significant questions remain regarding the fisheries oceanography needed to predict clutter and reverberation, the basic physics of scattering from individual fish and groups of fish is well understood. This motivates a data-driven, clutter adaptive approach to mitigation of biologically induced reverberation and clutter that utilizes physical models without requiring unavailable databases of fish-species abundance and local spatiotemporal distributions. In this work, physics-based models relating physical parameters of fish and their spatiotemporal behaviors to spectral properties and spatiotemporal

2p TUE. PM

distributions of scattering form the basis of *in situ* parameter estimation. In this presentation, adaptive processing developed for the early stages of the active-sonar processing chain using these *in situ* estimates is described, including waveform and matched-filter design and within and across-beam normalization. [Work supported by a NAVSEA Phase I SBIR award.]

4:05

2pSP10. Accuracy and resolution of bearing measurements for crossed-dipole receiver. Chunsheng Liu and Mark V. Trevorrow (Defence R&D Canada Atlantic, 9 Grove St., Dartmouth, NS B2Y 3Z7, Canada, chunsheng.liu@drdc-rddc.gc.ca)

A well-known system for estimating the bearing to an underwater acoustic source consists of two crossed dipole sensors and an omni-directional sensor. At DRDC Atlantic, several bearing estimation methods for this three-channel system in the presence of ocean noise have been examined with the goal of minimizing the bearing error. Example algorithms are arctangent and maximum likelihood beamformer. These algorithms have tradeoffs between estimation accuracy and computational efficiency, because any practical bearing-estimation algorithm must remain efficient enough for implementation in a real-time sonar processor. The accuracy of these algorithms can be demonstrated through comparison with the Cramér-Rao lower bound. Previously the calculation of bearing accuracy was based on the assumption of isotropic noise and a single signal source. However, the case of anisotropic noise and multiple-sources is more realistic, especially for bistatic or multistatic active sonar. Adaptive methods such as the Minimum Variance Distortionless Response (MVDR) beamformer will be explored to address this problem. The paper will discuss the potential improvement of bearing estimate accuracy and bearing resolution through numerical models.

4:20

2pSP11. Fast nearfield to farfield conversion for circular synthetic aperture sonar. Daniel Plotnick and Phillip L. Marston (Washington State Univ., 1510 NW Turner Dr., Apt. 4, Pullman, WA 99163, dsplotnick@gmail.com)

Monostatic circular synthetic aperture sonar (CSAS) images are formed by processing azimuthal angle dependent backscattering from a target at a fixed distance from a collocated source/receiver. In the laboratory data is taken by fixing the source location and spinning the target via a rotating mount. Typical CSAS imaging algorithms [Marston *et al*., Proc. IEEE Oceans (2011); Ferguson *et al*., J. Acoust. Soc. Am. **117**, 2915 (2005)] assume the scattering data are taken in the farfield. Experimental constraints may make farfield measurements impractical and thus require a target to be scanned in the nearfield. If left uncorrected, this results in distortions of the target image and possible distortions of the angular dependence of features. A fast approximate Hankel function based algorithm is presented that converts nearfield data to farfield data. Images and spectrograms of an extended target are compared for both cases. Spatial sampling requirements for data collection are also considered. [Work supported by ONR.]

TUESDAY AFTERNOON, 3 DECEMBER 2013 — CONTINENTAL 6, 2:00 P.M. TO 4:25 P.M.

Session 2pUW

Underwater Acoustics: Sound Propagation Through and Scattering by Internal Waves, Spice, and Finestructure in Shallow Water II: Past, Present, and Future

Steven I. Finette, Cochair
Acoust. Div., Naval Res. Lab., 4555 Overlook Ave. SW, Washington, DC 20375-5320

James Lynch, Cochair
Woods Hole Oceanogr., M.S. # 11, Bigelow 203, Woods Hole, MA 02543

Invited Papers

2:00

2pUW1. Predicting internal waves of acoustical significance. Timothy F. Duda (Appl. Ocean Phys. and Eng. Dept., Woods Hole Oceanographic Inst., Woods Hole, MA 02543, tduda@whoi.edu)

The broad dynamic range of internal gravity wave periods and scale lengths, as well as the variable group geometries of these waves, means that they are responsible for a variety of acoustic propagation effects. As a result, their degree of acoustical significance and the specific effects that they cause depend on time and place. A variety of internal-wave (IW) acoustic effects will be reviewed, with analysis of the regimes where each effect may have first-order propagation consequences. Many of the effects are specific to details such as the IW amplitudes, the angle between acoustic and IW propagation paths (and thus IW direction), IW proximity to sound source and/or receiver, and whether the IW field can be considered to be a random medium or, alternatively, a set of identifiable scattering features. The current state of our knowledge motivates a question: To what degree are specific acoustically relevant aspects of the IW field, and thus the acoustic effects, predictable? The answer to this question is being investigated through efforts to build ocean models bridging large-scale ocean dynamics, smaller-scale IW, and submesoscale dynamics, plus two-dimensional and three-dimensional acoustic propagation behavior.

2:20

2pUW2. Anisotropy in horizontal plane for the sound propagation in the ocean in the presence of internal waves. Boris Katsnelson (Marine GeoSci., Univ. of Haifa, 1, Universitetskaya sq, Voronezh 394006, Russian Federation, katz@phys.vsu.ru)

Review of researches dedicated to the sound propagation in the presence of internal waves is presented (starting with Evgeny Malyuzhinets, 1959 and Owen Lee, 1961). Main attention is paid on dependence of properties of the sound field (for example, sound amplitude fluctuations) on direction of propagation in horizontal plane. It was established—three main mechanisms: mode coupling, adiabatic variations, and horizontal refraction can be manifested in dependence on direction of sound propagation relative wave front of internal waves. Theoretical results as well as results of numerical modeling are presented illustrating different properties of acoustical signals for mentioned mechanisms including dependence on mode number and frequency. Experimental observations are also presented confirming real manifestations of mentioned three mechanisms.

Contributed Papers

2:40

2pUW3. Modeling the nonlinear internal wave field for coastal acoustics. Arthur E. Newhall, Ying-Tsong Lin, James F. Lynch, Timothy F. Duda (Appl. Ocean Phys. and Eng., Woods Hole Oceanogr. Inst., 98 Water St., 210 Bigelow Lab., M.S. #11, Woods Hole, MA 02543, anewhall@whoi.edu), and Karl R. Helfrich (Physical Oceanogr. Dept., Woods Hole Oceanographic Inst., Woods, MA)

It is now well known that coastal nonlinear internal waves, a common ocean feature, have large effects on the propagation of sound in the coastal ocean. Modeling such waves for input to acoustic models using regional ocean numerical models is difficult due to the fine spatial and temporal resolution needed. A multi-university program called Integrated Ocean Dynamics and Acoustics (IODA) is addressing this problem using two approaches—one a two-way, direct nested ocean model approach, and the other a blending of mesoscale ocean models and fine scale internal wave models. In this talk, we will discuss the latter approach. Issues of the internal wave source positions, their nonlinear propagation, earth rotation effects, field interpolation, output verification and integration of the ocean model with acoustics codes will be discussed. The results going from an ocean model to fully 3-D acoustic fields will be presented. [Work sponsored by ONR MURI program.]

2:55

2pUW4. Effect of internal waves on the waveguide invariant distribution. Daniel Rouseff (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St,, Seattle, WA 98105, rouseff@apl.washington.edu)

Multipath propagation in the ocean waveguide results in constructive and destructive interference between the paths. The resulting interference pattern, mapped versus range and frequency, often exhibits striations, alternating bands of high and low intensity. In simple waveguides, the trajectory of these striations can often be described by a single scalar parameter, the so-called waveguide invariant. In more complicated waveguides, the concept can be generalized to create the waveguide invariant distribution. In the present work, the effects of shallow water internal waves on the waveguide invariant distribution are examined. It is shown how internal waves can blur otherwise sharp striations and flatten the waveguide invariant distribution. [Work supported by ONR.]

3:10

2pUW5. Modal interference and frequency striations induced by moving nonlinear internal waves. Tsih C. Yang (College of Marine Sci., Nat. Sun Yet-sen Univ., 70 Lien Hai Rd., Kaohsiung 804, Taiwan, tsihyang@gmail.com) and Jin-Yuan Liu (Nat. Taitung Univ., Taitung, Taiwan)

Spectrogram of a broadband signal propagated over distances often displays striated bands of constant acoustic intensity levels plotted against the range and frequency. This phenomenon is due to modal interference and the slope of the striation is related to the source range and frequency by the waveguide invariant parameter "beta." As an application, one can estimate the range to a moving source based on the striation of the spectrogram (as a function of frequency and time), knowing the source velocity and the waveguide invariant parameter. Conversely, no frequency striation is expected in the spectrogram of the received signal when both the source and receiver are fixed (the frequency content remains unchanged with respect to time). However, when nonlinear internal waves, appearing as a train of moving solitons, are present in the propagation path, the spectrogram of the received signal will display frequency striations (with respect to time) even for a fixed source and receiver. This paper applies the mode coupling equation to calculate the modal interference, intensity fluctuation and the striation slope. Broadband spectrograms are simulated to show the frequency striation as the solitons move from the receiver to the source. Results from at-sea data will be presented for comparison.

3:25

2pUW6. Synthetic aperture sonar imaging of breaking internal wave structures. Anthony P. Lyons (Appl. Res. Lab, Penn State Univ., University Park, State College, PA 16803, apl2@psu.edu), Roy E. Hansen, Torstein O. Sæbø (Norwegian Defence Res. Establishment (FFI), Kjeller, Norway), and Hayden J. Callow (Kongsberg Maritime, Horten, Norway)

In October 2012, the Centre for Maritime Research and Experimentation (CMRE) conducted the ARISE'12 trials from the NATO research vessel Alliance, off of Elba island, Italy. During this trial, data were collected by the Norwegian Research Defence Establishment (FFI) using a HUGIN AUV with interferometric synthetic aperture sonar (SAS). Large visible structures in the SAS images and in the SAS bathymetries were caused by features in the water column, called boluses, which formed after breaking internal wave events. Variation in backscattered intensity from the seabed was caused by focusing of sound by the bolus and errors in bathymetry by the changes in phase as the acoustic energy moved through the lower sound speed bolus. A ray model and a 3D parabolic equation (PE) model were used to simulate the effects of the bolus on the acoustic field incident on the bottom and results compared favorably with SAS data. Models for the change in bolus size and speed as it propagated upslope also gave reasonable comparison to estimates obtained from SAS intensity images. [The work performed by APL was supported by the US Office of Naval Research.]

3:40

2pUW7. Modeling sound propagation through internal waves using a spectral element method. Sumedh M. Joshi (Ctr. for Appl. Mathematics, Cornell Univ., 657 Rhodes Hall, Ithaca, NY 14850, sumedh.m.joshi@gmail.com), Megan S. Ballard (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), and Peter J. Diamessis (Dept. of Civil and Environ. Eng., Cornell Univ., Ithaca, NY)

Considered here is the problem of low-frequency sound propagation over shallow, shoaling bathymetry in the presence of perturbations to the background sound velocity profile due to internal waves (IW). The question we attempt to answer is: to what degree can heuristic models of IWs coupled to numerical sound propagation models capture the variability in sound propagation observed in the environment? A high-order finite element model is employed to compute the acoustic field as it propagates through these IWs. The generality of the finite element method allows for spatial and temporal sound speed variations, and its convergence properties yield arbitrarily small error as the grid is rened. Simulations in the presence

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and absence of IWs will demonstrate the degree to which IWs influence sound propagation. Different models of IWs will demonstrate the sensitivity of the sound propagation to the choice of heuristic used for the IWs. Results will be shown for shoaling waveguides of O(100 m) depth and O(10 km) range, and frequencies of O(50 Hz). [Work supported by the NDSEG Fellowship.]

3:55

2pUW8. From Swarm 1995 to Shallow Water Experiment 2006: What have we learned in shallow waveguide acoustics? Mohsen Badiey (College of Earth, Ocean, and Environment, Univ. of Delaware, 261 S. College Ave., Robinson Hall, Newark, DE 19716, badiey@udel.edu)

Last 20 years have witnessed some very interesting scientific discoveries in shallow water waveguide physics. While before1995 SWARM experiment most studies in shallow water waveguides considered two-dimensional (2D) slices of the ocean waveguide to describe the behavior of sound wave propagation, many studies in the recent years have focused on the three dimensionality of the acoustic wave propagation in the presence of water column variability. This paper summarizes the highlights of what has been done with reference to the published literature and some ongoing work. [Work supported by ONR322OA.]

4:10

2pUW9. Estimates of source range using horizontal multipath in continental shelf environments. Megan S. Ballard (Appl. Res. Labs. at the Univ. of Texas at Austin, P.O. Box 8029, Austin, TX 78758, meganb@arlut.utexas.edu)

In a continental shelf environment, energy from an acoustic source propagating obliquely upslope repeatedly reflects from the sea surface and sloping seafloor with a consequent change in direction with each bottom reflection. Measurements of this type of effect can be observed in beamformed data from a horizontal line array. The arrivals from a single source are seen on two beams: a direct path with bearing angle corresponding to the source location and a refracted path with a bearing angle inshore of the source. In this work, the horizontal multipath effects are exploited to estimate the location of an acoustic source. Using the hybrid modeling approach of vertical modes and horizontal rays, rays are traced in the horizontal plane with refraction determined by the modal phase speed. Invoking reciprocity, the rays orientate from the center of the array and have launch angles equal to the estimated bearing angles of the direct and refracted paths. The location of the source in the horizontal plane is estimated from the point where the rays intersect. The technique is applied to data recorded on a horizontal line array located about 12 km east of the southern coast of Florida. [Work supported by ONR.]

TUESDAY AFTERNOON, 3 DECEMBER 2013 UNION SQUARE 14, 1:45 P.M. TO 3:15 P.M.

Meeting of Accredited Standards Committee (ASC) S1 Acoustics

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

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

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

People interested in attending the meeting of the TAGs for ISO/TC 43 Acoustics, ISO/TC 43/SC 3, Underwater acoustics, and IEC/TC 29 Electroacoustics, take note - those meetings will be held in conjunction with the Standards Plenary meeting at 9:00 a.m. on Tuesday, 3 December 2013.

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

Meeting of Accredited Standards Committee (ASC) S12 Noise

W.J. Murphy, Chair, ASC S12
NIOSH, 4676 Columbia Parkway, Mail Stop C27, Cincinnati, OH 45226

S.J. Lind, Vice Chair, ASC S12
The Trane Co., 3600 Pammel Creek Road, Bldg. 12-1, La Crosse WI 54601-7599

Accredited Standards Committee S12 on Noise. Working group chairs will report on the status of noise standards currently under development. Consideration will be given to new standards that might be needed over the next few years. Open discussion of committee reports is encouraged.

People interested in attending the meeting of the TAG for ISO/TC 43/SC 1 Noise, take note - that meeting will be held in conjunction with the Standards Plenary meeting at 9:00 a.m. on Tuesday, 3 December 2013.

Scope of S12: Standards, specifications and terminology in the field of acoustical noise pertaining to methods of measurement, evaluation and control, including biological safety, tolerance and comfort, and physical acoustics as related to environmental and occupational noise.

TUESDAY EVENING, 3 DECEMBER 2013 8:00 P.M. TO 10:00 P.M.

OPEN MEETINGS OF TECHNICAL COMMITTEES

The Technical Committees of the Acoustical Society of America will hold open meetings on Tuesday and Thursday evenings beginning at 8:00 p.m. and on Wednesday evening 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 Tuesday are as follows:

Acoustical Oceanography	Plaza B
Animal Bioacoustics	Union Square 23/24
Architectural Acoustics	Golden Gate 4/5
Engineering Acoustics	Mason
Noise	Continental 9
Physical Acoustics	Continental 7/8
Psychological and Physiological Acoustics	Continental 2/3