Session 3aAAa

Architectural Acoustics, ASA Committee on Standards, and Noise: Public Policy Implementation of School Acoustics

Jason E. Summers, Chair
Applied Research in Acoustics LLC, 1222 4th Street SW, Washington, DC 20024-2302

This moderated panel discussion brings together members of the Society involved in the development and adoption of the classroom-acoustics standard (ANSI S12.60-2002) and its adoption by local and state governments with representatives from the education-policy community and those familiar with the facilities and infrastructure challenges of local school districts. Following a brief introduction from the chair, Society member David Lubman will speak briefly on his experiences working with the Society and federal agencies in the technical development of the standard. Society member Bennett M. Brooks will speak to his experience working with other interested citizens as an advocate for the adoption of the standard in Connecticut. They will be joined by representatives from local schools and foundations that support education who will address facilities and infrastructure challenges and the relationship of this issue to the broader concerns of education policy. After remarks and moderated discussion by the panelists, the session will be open to questions from the audience.

WEDNESDAY MORNING, 20 MAY 2015 BALLROOM 3, 10:25 A.M. TO 12:00 NOON

Session 3aAAb

Architectural Acoustics and Noise: Session in Honor of Dick Campbell

K. Anthony Hoover, Cochair
McKay Conant Hoover, 5655 Lindero Canyon Road, Suite 325, Westlake Village, CA 91362

Eric L. Reuter, Cochair
Reuter Associates, LLC, 10 Vaughan Mall, Suite 201A, Portsmouth, NH 03801

Chair's Introduction—10:25

Invited Papers

10:30

3aAAb1. Remembering the life and contributions of Dick Campbell, a man of many talents. Eric L. Reuter (Reuter Assoc., LLC, 10 Vaughan Mall, Ste. 201A, Portsmouth, NH 03801, ereuter@reuterassociates.com)

Aside from his many technical contributions to audio and acoustics, Dick Campbell was a teacher and mentor to many, and had a diverse set of interests outside of his scientific pursuits. A horseman, pilot, sailor, marina owner, musician, tinkerer, community leader, and devoted father, he never had any trouble staying busy, even when faced with some harsh challenges. The author was a student of Dick's at Worcester Polytechnic Institute, and greatly valued his enthusiastic mentoring. This biographical presentation will include several personal anecdotes, as well as some stories and surprises collected from Dick's family and his many friends and colleagues.

10:50

3aAAb2. A series of special sessions on architectural acoustics and audio. K. Anthony Hoover (McKay Conant Hoover, 5655 Lindero Canyon Rd., Ste. 325, Westlake Village, CA 91362, thoover@mchinc.com) and Alexander U. Case (Sound Recording Technol., Univ. of Massachusetts Lowell, Lowell, MA)

Along with his love of concert halls, opera houses, and pipe organs that are often considered the top of the architectural acoustics food chain, Dick Campbell always shared his enthusiasm for smaller rooms, loudspeakers, computers, signal processing, and all forms of media. Daring to incorporate MIDI-driven, synthesized sounds produced through loudspeakers into the pit orchestra, Dick knew how to rabble rouse—and inspire. One legacy of his politely provocative thinking is a series of 11 special sessions—so far—started with him at the 1999 Columbus meeting, and continued to this day by the authors. The thread of related special sessions had its most recent iteration at the Fall 2014 meeting in Indianapolis, "Architectural acoustics and audio—Rooms, systems, and techniques for adapting, enhancing, and fictionalizing acoustic traits", with 22 papers covering a wide range of ideas and concepts. A tour of these sessions' past topics and authors reveals a rich body of work that might otherwise have been overlooked in ASA meetings without the seed planted by Dick Campbell.

11:10

3aAAb3. My time with Dr. "Dick" Campbell. William R. Michalson (Dept. of Elec. and Comput. Eng., Worcester Polytechnic Inst., 100 Inst. Rd., Worcester, MA 01609, wrm@wpi.edu)

I have been asked by a former student if I would have any interest in speaking about my experiences with Dick Campbell as a colleague and friend during our time working together. My answer? Of course, I would be honored to speak about the unofficial "Mayor of Woods Hole." We met shortly after I joined WPI. Dick's interest in acoustics and my interest in music electronics began a working relationship that evolved into a years-long friendship. Our collaborations benefited (and annoyed) the students we co-advised. Dick was bedridden when we last worked together, but that was Dick—have laptop, will Skype. He never seemed to let the "little things" get in the way of anything that grabbed his interest. I will talk about some of the projects we worked on together, including acoustic modeling, identification of mosquito species by their acoustic signature, and others. Additionally, I'll talk about the times I spent in his office at WPI, the "acoustics lab" in the Higgins garage, and his laboratory in Woods Hole—the World headquarters for "Bang-Campbell." Thank you for your attention, and thank you Dick. You've been an extraordinarily influential person in my life and the lives of many others.

11:30–12:00 Open Forum Contributions Invited and Welcomed

WEDNESDAY MORNING, 20 MAY 2015 — KINGS 2, 8:00 A.M. TO 11:45 A.M.

Session 3aBA

Biomedical Acoustics: Acoustic Radiation Force in Biomedical Applications I

Mostafa Fatemi, Cochair
Physiology & Biomedical Engineering, Mayo Clinic College of Medicine, 200 First St. SW, Rochester, MN 55905

Matthew W. Urban, Cochair
Department of Physiology and Biomedical Engineering, Mayo Clinic College of Medicine, 200 First St. SW, Rochester, MN 55905

Chair's Introduction—8:00

Invited Papers

8:05

3aBA1. Acoustic radiation force in biomedical applications: origin, past, present, and future. Armen Sarvazyan (Artann Labs., 1459 Lower Ferry Rd., Trenton, NJ 08618, armen@artannlabs.com)

In this talk, an overview of history and physical basis of biomedical applications of acoustic radiation force with a further look into new developments in this field will be presented. In 1902, Lord Rayleigh published his classical work on the theory of sound, introducing the concept of acoustic radiation pressure. Experimental demonstration of radiation force acting on particles in the standing wave field was made by August Kundt (1874). Much later, Wood and Loomis built radiation force balance (1927). Detailed analysis of ultrasound radiation force related to biomedical applications was made in numerous reviews and original articles of Wes Nyborg in the

1960s and 1970s. In spite of over century-old history, most of the significant biomedical applications of acoustic radiation force became known and were extensively studied only during last couple of decades. We will present and discuss recent progress in numerous applications such as the elasticity imaging, assessing viscoelastic properties of biological tissue, monitoring lesions during therapy, the manipulation of cells in suspension, acoustical tweezers, targeted drug and gene delivery. In addition to well established applications, the acoustic radiation force has considerable potential in numerous new areas, which will be discussed in the talk. [NIH R21AR065024.]

8:25

3aBA2. Direction of acoustic radiation force on spherical scatterers in soft tissue. Benjamin C. Treweek, Yurii A. Ilinskii, Evgenia A. Zabolotskaya, and Mark F. Hamilton (Appl. Res. Labs., Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, btreweek@utexas.edu)

A theory for acoustic radiation force on a viscoelastic sphere of arbitrary size in soft tissue has been reported previously for a nonaxisymmetric incident field described via spherical harmonic expansion [Ilinskii *et al.*, POMA **19**, 045004 (2013)]. At the Fall 2014 ASA meeting, the model was used to compute the radiation force on scatterers with different sizes and properties at various positions relative to the focus of an axisymmetric incident beam. For a particle located away from the focus, the model predicts a change in the direction of the axial or the transverse component of the radiation force depending on properties of both the particle and the host medium. The focus of the present contribution is this change in direction. Scatterers with various sizes and mechanical properties are considered, and small particles are found to be more prone to this phenomenon. Additionally, the reversal in direction is found to be sensitive to variations in the shear modulus of the host medium. Comparisons are made with liquid as the shear modulus of the host medium spans the range of values encountered in soft tissue. [Work supported by the ARL:UT McKinney Fellowship in Acoustics.]

8:45

3aBA3. Radiation torque, streaming, and viscous absorption due to orthogonal waves in slightly viscous fluids. Likun Zhang (Ctr. for Nonlinear Dynam., Univ. of Texas at Austin, 2515 Speedway, Stop C1610, Austin, TX 78712-1199, lzhang@chaos.utexas.edu) and Philip L. Marston (Dept. of Phys. and Astronomy, Washington State Univ., Pullman, WA)

Torque generated by orthogonal waves in slightly viscous fluids is analyzed to explore its connection with the near-boundary viscous absorption and streaming [Zhang and Marston, J. Acoust. Soc. Am. **131**, 2917–2921 (2014)]. The analysis is based on a viscous correction and beam superposition of vortex fields. The analysis reports an approximation for the torque on a compressible small solid sphere and the viscous power absorption [Eqs. (15) and (21) in Zhang and Marston]. The torque expression in the dense sphere limit recovers prior expressions using near-boundary flow and streaming analyses: [Busse and Wang, J. Acoust. Soc. Am. **69**, 1634–1638 (1981)] and [Lee and Wang, J. Acoust. Soc. Am. **85**, 1081–1088 (1989)]. Our analysis extends prior results to give the torque on small compressible solid spheres of finite mass. The torque does not depend explicitly on the compressibility of the sphere, but does depend on the density of the sphere. Our analysis also gives the proportional relationship between the torque and the near-boundary viscous absorption. The results are generalized to cases of standing orthogonal waves and acoustic vortex fields, and other small axisymmetric obstacles such as circular cylinders and disks. [Work supported by the 2013-14 F. V. Hunt Postdoctoral Fellowship (Zhang) and by ONR (Marston).]

9:05

3aBA4. The pull of finite acoustic beams. Farid G. Mitri (Area 52 Technol., Chevron, 5 Bisbee Court, Santa Fe, NM 87508, f.g.mitri@ieee.org)

The counterintuitive effect of pulling an object back towards a finite (bounded) circular source with the use of acoustic forces of progressive waves is demonstrated, which defines the acoustic "tractor beam" behavior [F. G. Mitri, "Near-field single tractor-beam acoustical tweezers," Appl. Phys. Lett. **103**(11), 114102, 2013; "Single Bessel tractor-beam tweezers," Wave Motion **51**(6), 986–993, 2014]. Experimental results and numerical predictions show the emergence of negative forces due to continuous (CW) and amplitude-modulated (AM) propagating waves. The circumstances under which a collimated beam emanating from a finite aperture can act as a tractor beam and related investigations are also discussed.

9:25

3aBA5. Applications of acoustic radiation force for microvascular tissue engineering. Diane Dalecki, Eric S. Comeau (Dept. of Biomedical Eng., Univ. of Rochester, 310 Goergen Hall, P.O. Box 270168, Rochester, NY 14627, ddalecki@ur.rochester.edu), and Denise C. Hocking (Pharmacology and Physiol., Univ. of Rochester, Rochester, NY)

We have developed a non-invasive ultrasound-based method to spatially pattern cells within three-dimensional (3D) hydrogels, and have demonstrated translation of this technology to microvascular tissue engineering. Acoustic radiation forces associated with an ultrasound standing wave field (USWF) can rapidly organize a variety of cell types into distinct spatial patterns within 3D hydrogels. USWF-induced patterning of endothelial cells into distinct multicellular planar bands can lead to the rapid formation of complex microvessel networks throughout the volume of the collagen hydrogel. Recent efforts have focused on optimizing acoustic exposure parameters to control resultant 3D microvessel morphology, producing multilayered composite constructs with complex, physiologically relevant vascular morphologies, and developing dual ultrasound transducer systems to noninvasively direct cell patterning within injectable collagen hydrogels in situ. Engineering microvessel networks that structurally and functionally mimic native microvasculature is critical for the fabrication and survival of a broad range of bioengineered tissues, and for the fabrication of small, vascularized tissue constructs for use as realistic cost-effective *in vitro* models for drug discovery and testing.

9:45–10:00 Break

Contributed Papers

10:00

3aBA6. Acoustic radiation force and particle dynamics in cylindrical resonators. Lev A. Ostrovsky (PSD, NOAA ESRL, 325 Broadway, R/PSD99, Boulder, CO 80305, lev.a.ostrovsky@noaa.gov)

Acoustic radiation force is widely used in biomedical studies and medical diagnostics. Among the promising areas of its application is concentration and stirring of particles and bubbles in ultrasonic resonators (e.g., [1]). In a number of cases, cylindrical resonators have a practical advantage vs. plane resonators (acoustic interferometers) due to the axial energy concentration [2], [3]. Theoretical analysis of particle dynamics in such resonators is often very complicated. This presentation outlines recent results in modeling of dynamics of micro-particles and microbubbles in cylindrical resonators. In particular, concentration and separation of "heavy" and "light" particles and near-resonant bubbles are discussed. The role of ultrasound nonlinearity in the considered effects is also discussed. [1]. A. Sarvazyan and L. Ostrovsky, J. Acoust. Soc. Am. **125**, 3548–3554 (2009). [2]. A. Priev and A. Sarvazyan, J. Acoust. Soc. Am. **125**, 2593 (2009). [3]. L. Ostrovsky, A. Priev, V. Ponomarev, and Y. Barenholz, Proc. Meetings Acoust. **14**, 020002 (2013).

10:15

3aBA7. Acoustic radiation force on a particle in ultrasonic standing wave fields driven by a piezoelectric plate. Bart Lipkens (Mech. Eng., Western New England Univ., Springfield, MA), Ben Ross-Johnsrud, Minghe Liu (FloDesign Sonics, 380 Main St., Wilbraham, MA 01095, b.johnsrud@fdsonics.com), Yurii Ilinskii, and Evgenia Zabolotskaya (Appl. Res. Labs, The Univ. of Texas at Austin, Austin, TX)

Acoustic radiation forces on a particle or droplet in a liquid have been studied mostly for relative simple acoustic fields such as plane traveling waves, or planar standing waves. In this study, a more complicated acoustic field is considered, namely, the acoustic standing wave field generated by a rectangular piezoelectric plate with finite size. The piezo-electric plate is excited in its thickness mode. A three dimensional model of the vibration of the piezo-electric plate has been developed. The plate is loaded by an acoustic standing wave field in a fluid on one side, and is air backed on the other side. The three-dimensional acoustic standing wave field generated by the piezo-electric plate is calculated. The acoustic radiation force on a particle or droplet can then be calculated using a framework such as developed by Gor'kov or Ilinskii and Zabolotskaya. The theoretical calculations are compared with numerical predictions using a two-dimensional model of the piezo-electric plate and resonator. Finally, experimental results were obtained in a resonator with a one inch acoustic path length and driven by a 2 MHz PZT-8 one inch by one inch piezo-electric plate. Experimental results will be compared with theoretical and numerical predictions.

10:30

3aBA8. Enhancement of osteogenesis and stem cell differentiation by injectable nanocomposite and dynamic acoustic radiation force stimulation. Vaishnavi Shrivastava, Sunny Patel, Minyi Hu, Suphannee Pongkitwitoon, Balaji Sitharaman, and Yi-Xian Qin (Biomedical Eng., Stony Brook Univ., BioEng. Bldg., Rm. 215, Stony Brook, NY 11794, yi-xian.qin@stonybrook.edu)

Osteopenia affects mineral density, microstructure, and integrity of bone, leading to increased risk of fractures, as well as high rates of non-union, which affect patients' quality of life. Current treatments are ineffective, requiring invasive surgeries and/or long-term drug therapy. The objective of this study was to develop a novel noninvasive biomimetic treatment for rapid regeneration to promote cell differentiation and osteogenesis. An injectable orthopedic implant was designed by developing a thermosensitive scaffold incorporating carbon nanotubes and chitosan-β glycerophosphate hydrogels. An innovative biophysical stimulation using dynamic ultrasound radiation force (ARF) was used to induce carbon nanotube resonance for regulating osteogenic differentiation of stem cells. An assay on activity of ALP, a biomarker of osteogenesis, and a fluorescence-based live/dead cell assay were conducted to determine the best treatment for inducing rapid cell formation. The single-walled carbon nanotube scaffold and 60 mW/cm^2 ARF group was the best treatment, and enhanced ALP activity and stem cell differentiation by 430%. Cell viability was increased by 65% through the ARF treatment. This study developed a novel treatment for regulating cell proliferation using single-walled carbon nanotube nanocomposite implants and synergistic stimulation of stem cells with ultrasound. This rapid, noninvasive, cost-effective treatment may provide an innovative alternative for osteoporotic treatment and rapid fracture healing.

10:45

3aBA9. Acoustic separation of milk fat globules—Principles in large scale processing. Thomas Leong, Linda Johansson (Faculty of Sci., Eng. and Technol., Swinburne Univ. of Technol., John St. Hawthorn, Melbourne, Victoria 3122, Australia, tleong@swin.edu.au), Pablo Juliano (CSIRO Food and Nutrition, Melbourne, Victoria, Australia), and Richard Manasseh (Faculty of Sci., Eng. and Technol., Swinburne Univ. of Technol., Melbourne, Victorai, Australia)

The acoustic manipulation of particulates in standing wave fields is a well-established technique, with the fundamental theory describing the acoustic radiation forces first reported in 1934 by King. To date, there are few demonstrations of the technique on a volume-scale relevant to industrial application, due to difficulties in scaling the acoustic radiation forces over large distances. Other issues such as the onset of acoustic streaming with high acoustic power input may also impact upon the effectiveness of separation in large systems by disrupting the collection of particulates in the pressure nodal/antinodal regions. In recent work, we have demonstrated the capability of ultrasonic standing waves applied in a liter-scale vessel to concentrate and/or remove fat globules from whole milk with volume throughputs up to 30 L/h. By tuning parameters according to acoustic fundamentals, the technique can be used to specifically select milk fat globules of different sizes in the collected fractions. We report key design and operation principles at large scale using milk as a model system, which has potential to be applied to particulate fluids of biomedical relevance such as blood and lipids.

11:00

3aBA10. Understanding diffractive and reflective contributions to scattering and extinction of intersecting plane waves. Philip L. Marston (Phys. and Astronomy Dept., Washington State Univ., Pullman, WA 99164-2814, marston@wsu.edu)

Scattering and extinction of intersecting plane waves is relevant to many processes of interest, including the interpretation of bistatic scattering and the understanding of acoustical radiation forces. A special case that exhibits diverse behavior is the illumination of a sphere by an acoustic Bessel beam. Consider high frequency scattering in which the sphere radius exceeds the wavelength. Recent progress in understanding diffractive scattering contributions uses Babinet's principle and the Kirchhoff approximation [P. L. Marston, J. Acoust. Soc. Am. **135**, 1668–1671 (2014)]. The optical theorem for invariant beams [L. Zhang and P. L. Marston, J. Acoust. Soc. Am. **131**, EL329–EL335 (2012)] was used to evaluate the extinction. Comparison with the partial wave series for the extinction supports the utility of quasi-scaling involving the Bessel beam's conic angle. The reflective scattering contribution is subtler; however, evaluating it quantitatively explains an abrupt transition in the spacing of fringes in bistatic scattering patterns for spheres centered in Bessel beams. The transition is visible where the scattering angle passes through the beam's conic angle [P. L. Marston, J. Acoust. Soc. Am. **121**, 753–758 (2007)]. Diffractive and reflective contributions for smooth objects illuminated by intersecting waves are relevant to many other situations. [Work supported by ONR.]

11:15

3aBA11. Spatially uniform harmonic acoustic radiation force excitation using one-dimensional linear array. Mahdi Bayat, Azra Alizad, and Mostafa Fatemi (Physiol. and Biomedical Eng., Mayo Clinic, 200 1st St. SW, Rochester, MN 55905, bayat.mahdi@mayo.edu)

Low frequency harmonic force excitation can be used for both imaging and tissue characterization. This type of excitation can be achieved via

intersecting two beams that slightly differ in frequency. One requirement of this technique is creating focused intersecting beams using available hardware. Previous studies have suggested different configurations for creating harmonic radiation force using linear arrays. However, these methods rely on identical frequency response assumption across all elements of the aperture. In practice, this assumption can be easily violated, especially if the transmit frequencies are not around the center frequency of the probe. When using electronic steering and focusing at different locations, this can result in a non-uniform radiation force pattern due to frequency changes in the sub-aperture. When used for vibro-acoustography, this manifests as vertical streaks in the final image that can severely degrade the quality of the image. We present a novel technique that minimizes element transmit frequency switching which in turn results in a more uniform excitation intensity. The results of applying this aperture configuration for vibro-acoustography imaging of breast phantoms with hard inclusions show significant improvements over previous methods. [This work was supported by NIH Grant R01 EB17213.]

11:30

3aBA12. Sternal vibrations reflect hemodynamic changes during immersion: Underwater ballistocardiography. Andrew D. Wiens, Andrew Carek, and Omer T. Inan (Elec. and Comput. Eng., Georgia Inst. of Technol., 85 Fifth St. NW, Atlanta, GA 30308, andrew.wiens@gatech.edu)

Ballistocardiography (BCG) is a method for measuring the small vibrations of the body caused by the beating of the human heart. In this study, vibration measurements of the sternum for the purpose of noninvasive hemodynamic monitoring during total body immersion in water are recorded and examined for the first time. Three individuals wore a low-noise accelerometer while immersed in water of varying temperature up to the neck, and Valsalva maneuvers were performed. The resulting waveforms reveal distinct differences in signal morphology between three postures and two water temperatures, suggesting that underwater BCG could be applied in aquatic environments without a need for electrodes.

WEDNESDAY MORNING, 20 MAY 2015 — KINGS 1, 8:30 A.M. TO 11:55 A.M.

Session 3aED

Education in Acoustics and Physical Acoustics: Preparing Graduate Students for Careers in Acoustics

Daniel A. Russell, Cochair
Graduate Program in Acoustics, Pennsylvania State University, 201 Applied Science Bldg., University Park, PA 16802

E. Carr Everbach, Cochair
Engineering, Swarthmore College, 500 College Avenue, Swarthmore, PA 19081

Chair's Introduction—8:30

Invited Papers

8:35

3aED1. Careers in acoustics: A journey from research lab engineer to tenure-track faculty. Andrew R. Barnard (Mech. Eng. - Eng. Mech., Michigan Technolog. Univ., 815 R.L. Smith MEEM Bldg., 1400 Townsend Dr., Houghton, MI 49931, arbarnar@mtu.edu)

As they approach the end of their Bachelor's or Master's degree, many students ask themselves "Should I get my Ph.D. or take the money and move to industry?" There is a perception that it may be more difficult to get a job after obtaining a Ph.D. due to the dreaded "over-qualification." I have found the opposite in my career. In this talk, I will present my story of completing my Ph.D. and moving into a position at a university research laboratory, the Applied Research Lab at Penn State, and my subsequent transition into a tenure-track faculty position in the department of Mechanical Engineering—Engineering Mechanics at Michigan Tech University. Pros and Cons of both positions will be discussed as will my view on the importance of the Ph.D. researcher in the future of academia, government, and industry. I will conclude with my advice for current and future Ph.D. students interested in research and/or academics.

8:55

3aED2. Perspective from a college of engineering at a large, public research university. Anthony A. Atchley (Gradate Program in Acoust., Penn State Univ., College of Eng., 101C Hammond Bldg., University Park, PA 16802, atchley@psu.edu)

A career at a large, public research-intensive university is the gateway to a life-long opportunity to create new knowledge, develop realistic solutions to the world's greatest challenges, influence young minds, educate the next generation of leaders, and service society. The value proposition for such a privileged career opportunity, however, requires a life-long commitment to excellence, self-improvement, and reinvention. This presentation, based upon the author's experience as a faculty member, graduate program chair, and senior college-level administrator, is aimed at students who are considering faculty positions at such institutions. It will address getting the most out of your Ph.D. program and post-doctoral experience; the critical first few years as a faculty member; balancing the demands of and integrating teaching, research, and service; becoming skilled at writing proposals and desseminating research results; qualities you should seek in an institution you are considering and questions you should ask when interviewing.

9:15

3aED3. Publish or perish and funding or failure—The dark side of a career in academia. Ralph T. Muehleisen (Energy Systems, Argonne National Lab., 9700 S. Cass Ave., Bldg. 221, Lemont, IL 60439, rmuehleisen@anl.gov)

A career in academia can be extremely rewarding. It can be intellectually stimulating, financially rewarding, provide an extremely flexible work schedule, and give an opportunity for one to truly impact the world in a positive way. But, there is a dark side to academia that one should be aware of before jumping in. Academia can also be full of petty and insignificant politics, enormous egos, and unrealistic expectations. Perhaps the darkest side relates to the need to "publish or perish" and "find funding or fail" that can be found at many research universities. This presentation will provide an honest discussion of the ups and downs of academia from someone who found both success and failure.

9:35

3aED4. Navigating the track to tenure: A liberal arts college perspective. David T. Bradley (Phys. + Astronomy, Vassar College, 124 Raymond Ave., #745, Poughkeepsie, NY 12604, dabradley@vassar.edu)

Pursuing a career in the academy can be rewarding and challenging. Securing a tenure-track position at a liberal arts college can be particularly difficult since there are far fewer available positions, and because this type of institution is very different from the ones that grant the requisite Ph.D. degree for the job. A liberal arts college is an institution that is undergraduate focused, offering a breadth of courses in the humanities, social sciences, and natural sciences, while still providing the appropriate depth and rigor to prepare students for graduate school and industry. These colleges tend to have a stronger emphasis on teaching than larger universities, although professors are still expected to develop a thriving research program, typically only with the help of a few undergraduate research assistants. This presentation will discuss the unique aspects of being a professor at a liberal arts college while describing one successful path to the coveted tenured status.

9:55–10:10 Break

10:10

3aED5. Preparing for a career in academia: Managing students in research. Kent L. Gee, Tracianne B. Neilsen, Scott D. Sommerfeldt, and Timothy W. Leishman (Dept. of Phys. and Astronomy, Brigham Young Univ., N243 ESC, Provo, UT 84602, kentgee@byu.edu)

A new faculty member faces challenges associated with meeting and balancing various teaching, research, and citizenship demands. This includes managing students as part of developing a research program. Despite the vital importance of this skill, effective employee management is not something a student inherently learns in graduate school nor does it often receive attention as part of new faculty development workshops. This presentation discusses lessons learned regarding research student management. These include setting a scholarly goal at the outset with specific result-driven milestones and clear expectations of the "end game," adopting a management style that is best suited to each student's personality, adapting the project where possible to student strengths, and helping them learn to write as early as possible. Graduate students can be trained to become effective peer mentors of undergraduate students, increasing both a sense of teamwork and overall productivity. New faculty members will benefit from actively seeking mentorship by more experienced colleagues who have successfully built student-based research programs.

10:30

3aED6. An administrative perspective on preparing for careers in academia. Scott D. Sommerfeldt (Dept. of Phys., Brigham Young Univ., N181 ESC, Provo, UT 84602, scott_sommerfeldt@byu.edu)

As the dean of a college, one of my responsibilities is to interview each person that the departments in my college have some interest in hiring. As a result, I have interviewed dozens of candidates, some of whom were very strong and some of whom were far weaker. In this presentation, I will overview some aspects of career preparation that students can focus on that will help prepare them to be strong candidates if they are interested in pursuing a career in academia. I will also discuss some aspects of the interviewing process that students can prepare for in an effort to present themselves favorably during interviews.

10:50

3aED7. Preparing University of Mississippi graduate students for careers in acoustics. Richard Raspet (National Ctr. for Physical Acoust, NCPA, Univ. of MS, University, MS 38677, raspet@olemiss.edu)

The acoustics program at the University of Mississippi originated in the Physics Department and is still closely associated with physics. The physics department requires all students teach two semesters of laboratory under the supervision of a laboratory physicist. The required physics curriculum is extensive so that the acoustics students only receive two semesters of acoustics courses. This leads to careful course design so that our students are broadly literate in acoustics. The necessary compression of class work means that speaking and writing skills are taught to the student by their research advisor in preparing conference talks, written papers, and dissertations.

11:10–11:55 Panel Discussion

Session 3aMUa

Musical Acoustics: Acoustics of Percussion Instruments

Christopher Jasinski, Cochair
University of Notre Dame, 54162 Ironwood Road, South Bend, IN 46635

Andrew C. Morrison, Cochair
Natural Science Department, Joliet Junior College, 1215 Houbolt Rd., Joliet, IL 60431

Invited Papers

9:00

3aMUa1. The harmonic vistaphone. Garry M. Kvistad (Woodstock Percussion, Inc., 167 DuBois Rd., Shokan, NY 12481, garry@chimes.com)

Percussion instruments offer a wide range of frequencies from the lowest gongs to the highest bells. The natural modes of vibrations of all percussion instruments produce overtones, which are inharmonic. Builders of keyboard percussion instruments must carve bars in specific ways to include a few harmonic overtones of their choosing. Garry Kvistad, founder/owner of Woodstock Percussion, Inc., has built a set of tubes/rods, which he calls the Vistaphone, carefully tuned to the first 32 partials of the harmonic overtone series. In this presentation, Mr. Kvistad will demonstrate how polyrhythmic patterns played slowly can be sped up by means of a proprietary software program to yield musical intervals that correspond to the ratios of just intonation. After this introduction illustrating the relationship of pulse and pitch, the Vistaphone will be played to reinforce the phenomena of pure harmonic relationships. The listener will be able to experience the pure harmonic overtone series in just intonation which yields a rich bouquet of sustained sound with a multitude of harmonically related difference/summation tones. The only way to obtain this effect acoustically is to build a unique instrument of this type.

9:20

3aMUa2. Coupling of drumhead vibrations: Experimental and numerical analysis. Randy Worland and Benjamin Boe (Phys., Univ. of Puget Sound, 1500 N. Warner, Tacoma, WA 98416, worland@pugetsound.edu)

Many musical drums, such as snare drums, bass drums, and tom toms, consist of two stretched membranes at opposite ends of a cylindrical shell. Vibrations of the two heads are coupled acoustically by the enclosed air and mechanically by the shell itself. The degree of coupling varies with modal frequency and shape, and depends on several factors including the geometry of the drum shell (diameter and length) and the tensions of the two heads. Experimental measurements of coupled vibrations on drums of various sizes have been made using an electronic speckle-pattern interferometer to image the deflection shapes, amplitudes, and relative phases of both heads simultaneously. These measurements are compared with a finite element model of the system that illuminates the role of the vibrational patterns of the enclosed air in the coupling process.

9:40

3aMUa3. From laboratory to concert hall: The invention and production of a new drum tuning system. Rohan Krishnamurthy (Music, Ohlone College, 544 Sunrise Circle, San Francisco, CA 94105, rohan.krishnamurthy@rochester.edu)

I discuss the motivations behind the invention and production of my new drumhead tuning system. The design originated with the *mridangam*, an ancient and popular pitched drum from South India, which has been constructed the same way for centuries. The mridangam is notoriously difficult to tune and replace drumheads, which consists of multiple layers of leather and a complex iron oxide and starch loading. My new design makes tuning and head replacement more user-friendly and durable, and can be applied to any drum that needs to be fine-tuned. I present my research comparing the acoustics of the traditional mridangam design with my new design. I also explore the challenges and opportunities involved in the international manufacturing process that is currently underway. The presentation concludes with an interactive performance on the mridangam.

10:00

3aMUa4. A method for predicting the performance of a steelband orchestra in arbitrary environments. David Chow and Brian Copeland (Dept. of Elec. and Comput. Eng., The Univ. of the West Indies, St. Augustine, Trinidad and Tobago, davidachow@gmail.com)

One of the challenges in steelpan performance is in determining the best placement of the various instruments in an orchestra so as to achieve the level performance desired by the musical arranger. This research attempts to develop a method for analyzing the performance of the tenor steelpan in arbitrary acoustic spaces using the room acoustic modeling software, Enhanced Acoustic Simulation for Engineers (EASE). When provided with an acoustic characterization of a speaker system, particularly the magnitude and phase radiation data, and the room in which it is located, EASE can be used to predict the resulting acoustic sound-field. EASE also requires an electrical equivalent of

the instrument's sound source and, through its auralization feature, allow the user to actually hear an actual performance. EASE uses a variety of reconstruction techniques, such as ray tracing and boundary element method, to provide the most accurate responses. The study will verify the approach by comparing simulation results on direct sound coverage, total SPL, Clarity (C80), Centre Time, Reverberation, and Early decay times of the instrument with real time measurements in the usual 1/3-band octave frequency resolution. Although the work focuses on the tenor steelpan, it can be easily extended to predict the performance of an entire ensemble of steelpans in any given space.

Contributed Paper

10:20

3aMUa5. Ancient Aztec drum, the Huehuetl. Sergio Beristain (IMA, ESIME, IPN, P.O.Box 12-1022, Narvarte, Mexico City 03001, Mexico, sberista@hotmail.com)

The Aztecs and other cultures in Mesoamerica used to build and play during their festivities and warriors gatherings, simple drums made out from a hollow tree trunk, carved in the inside in order to make a fairly uniform thick surface to form the resonant chamber, topped by an ocelot skin, and open in the bottom, usually by three supports made from the same tree piece, called huehuetl, which means that these drums were of different shapes and sizes, some of these trunks were highly decorated by carving in the surface three dimensional images related to animals, faces, warriors, and other figures. These drums were excited on the top either by bare hand or wood sticks, varying the sound by the strength of the hit, and the area of the ocelot skin actually hit. Here are presented some pictures of ancient huehuetls together with the acoustic analysis of the sounds produced with a small replica of this percussion instrument.

WEDNESDAY MORNING, 20 MAY 2015 BALLROOM 1, 11:00 A.M. TO 12:00 NOON

Session 3aMUb

Musical Acoustics: Acoustics of Percussion Instruments II: Concert

Garry Kvistad and Rohan Krishnamurthy will be demonstrating in concert the acoustics of various percussion instruments. All are welcome to attend this performance.

WEDNESDAY MORNING, 20 MAY 2015 KINGS 5, 8:00 A.M. TO 11:40 A.M.

Session 3aNS

Noise, Architectural Acoustics, and ASA Committee on Standards: Noise with Tonal Components in the Built Environment

Lily M. Wang, Chair

Durham School of Architectural Engineering and Construction, University of Nebraska - Lincoln, PKI 101A, 1110 S. 67th St., Omaha, NE 68182-0816

Chair's Introduction—8:00

Invited Papers

8:05

3aNS1. An overview of issues in quantification of the tonalness of sounds. Patricia Davies (Ray W. Herrick Labs., School of Mech. Eng., Purdue Univ., 177 South Russell St., West Lafayette, IN 47907-2099, daviesp@purdue.edu)

Audible pitches in noise increase annoyance, and tone corrections to noise level metrics have been used in noise impact quantification for over 50 years. These corrections are a function of the prominence of the tone relative to the nearby spectral components. Frequency masking, the width of the tonal feature, and the frequency may also be taken into account. All measures are derived from

spectral estimates. When a single or well isolated tonal components are present in noise, sounds are stationary and sufficient data is available for good spectral estimation, many tonalness metrics (Prominence Ratio, Tone to Noise Ratio, Tone Audibility, and Aures Tonality) work well. More challenging is quantification of tonalness when: the frequency is varying with time, either slowly or quickly; there is a harmonic series that is predominantly heard holistically not as a series of individual pitches; or there is beating because of the presence of multiple tones within a critical band. In music, the characteristics of harmonic families help us identify different instruments and manipulation of harmonic series in machinery noise can make tonal sounds more or less pleasant. The challenges of quantifying tonal contributions and their impact in these more complicated cases will be described

8:25

3aNS2. Tonal noise characteristics from HVAC equipment and how to improve practically and technically. Jyuhn-Jier J. Wang (Ingersoll-Rand PLC, 3600 Pammel Creek Rd., Bldg. 12-1, LaCrosse, WI 54601, jjwang@trane.com)

All HVAC equipment has rotating components, such as compressors and fans. These critical components often exhibit annoying tonal characteristics. Although there are many engineering methods to improve the tonal characteristics, there exists no commonly accepted tonal metrics in HVAC equipment specifications, which typically only include sound power or sound pressure levels at few operating conditions. However, the equipment sound level can vary greatly by the operating conditions—the range may be as much as 10 dB or more, depending on compressor/fan types, sizes, and speed range. Although HVAC manufacturers understand the impact of the equipment tonal characteristics to the acoustical comfort of built environment, lack of customer requirement in the annoyance of the tonal characteristics makes it very challenging to be assessed and improved upon with the other competing functional requirements. This presentation will show typical tonal noise characteristics of chillers, air handlers, and transport HVAC equipment, and then propose methods to improve practically and technically from equipment manufacturer's point of view.

8:45

3aNS3. Tonal noise in heating, ventilation, and air-conditioning equipment in commercial and residential buildings. Curtis Eichelberger, James E. Bender, Paul F. Bauch, and Jarom H. Giraud (Bldg. Efficiency, Johnson Controls, 631 Richland Ave., York, PA 17403, curtis.eichelberger@jci.com)

All heating, ventilation, and air-conditioning (HVAC) equipment that moves air or compresses refrigerant produces noise with tonal content. The challenge for the equipment manufacturer and building designer is to make sure that this tonal noise is isolated from the occupied spaces in a building. This paper provides an overview of the tonal content in airborne and structureborne sound produced by heating, cooling, and ventilation equipment. Examples of the tonal noise in HVAC components and packaged equipment will be presented. The challenge to manufacturers in measuring and rating equipment sound will also be covered. An overview of methods for testing and rating HVAC equipment sound will be presented, with emphasis on the uncertainty in characterizing the tonal content.

9:05

3aNS4. Transformers in the built environment. Felicia Doggett (Metropolitan Acoust., LLC, 40 W. Evergreen Ave., Ste. 108, Philadelphia, PA 19118, f.doggett@metro-acoustics.com)

Tonal noise in the built environment can be distracting, annoying, and at times make it practically impossible to live or work in a building. One of the worst offenders of tonal noise in buildings is electrical transformers. Whereas the airborne sound output of transformers is not particularly high, the vibration and subsequent structure-borne noise can be extreme. In large commercial buildings, transformers are typically installed directly on structural concrete floors. When these same transformers are located on upper floors of buildings, structure-borne sound transmission can be significant on the occupied floors below. For the most part, the transmission loss of the concrete floors is more than sufficient in attenuating airborne sound but structureborne sound can carry throughout the floors below for large distances from the transformer location. The best way to truly solve the problem is by isolating the transformer. So, how easy is it to elevate and isolate transformers that weigh as much as a truck? This presentation focuses on case studies of transformer noise in buildings that are incorrectly installed, what was subsequently done in an effort to remediate the problems, and whether or not it was successful.

9:25

3aNS5. Rumble strip sound level evaluation—Comparison of three designs. David Braslau (David Braslau Assoc., Inc., 6603 Queen Ave. S, Ste. N, Richfield, MN 55423, david@braslau.com) and Edward Terhaar (Wenck Assoc., Inc. , Maple Plain, MN)

Results of sound level monitoring from three types of longitudinal rumble strips installed along the edge of two-lane rural roads are discussed. This study was in response to objections by landowners about unwanted noise caused by vehicles traveling over rumble strips. The ultimate study goal is to provide maximum tactile and tonal sound levels at the driver position while minimizing exterior sound levels. Both exterior and vehicle interior sound levels were measured for three longitudinal edge-of-pavement rumble strip designs—California, Pennsylvania, and Minnesota. Simultaneous digital audio files were also recorded. Three vehicles types were tested at 30, 45, and 60 mph. Comparison of exterior levels and interior sound levels showed that the Pennsylvania design is the quietest, both interior and exterior. Interior levels from the Minnesota and California designs are similar but exterior levels are higher for the Minnesota design. The California design generates two tonal peaks. The Minnesota design generates only one higher tonal peak. Studies to date have not accurately addressed the distance at which a rumble strip signal can be heard. Sound levels were projected perpendicular to the roadway suggesting detectability at over a mile from the Minnesota design in a typical rural environment.

9:45–10:00 Break

10:00

3aNS6. Subjective evaluation of loudness and preference for noise containing audible tones. Stephan Toepken, Steven van de Par, and Reinhard Weber (Acoust. Group, Oldenburg Univ., Carl-von-Ossietzky-Str.9-11, Oldenburg 26129, Germany, stephan.toepken@uni-oldenburg.de)

Noise in living environments often contains unwanted tonal components that contribute to the unpleasant and intrusive character of a sound and usually lead to an increased annoyance. In the measurement and assessment of noise immissions after the German DIN 45681:2005-3 standard, the higher annoyance due to audible tonal components is covered by so called tone adjustments. Depending on the SNR of a tonal component within each critical band, the level of a sound can be charged with a penalty of up to 6 dB. In this study, the psychoacoustic penalty levels for sounds containing tonal components are directly determined in listening tests. The points of subjective equality (PSEs) for loudness and preference are determined using a matching procedure. The level of the test sound with tonal components is varied until it is equally loud / preferred as a tone-free reference sound that is constant in level. The level difference between test and reference sound at the PSEs is a direct measure of the penalty level. In listening tests with pure multi-tone sounds the penalty levels were found to be well above 6 dB, indicating that for these stimuli the German DIN standard is underestimating the penalty level.

10:20

3aNS7. Multidimensional characteristics of annoyance perception to tonal building mechanical noises. Joonhee Lee and Lily M. Wang (Durham School of Architectural Eng., Univ. of Nebraska - Lincoln, 1110 S. 67th St., Omaha, NE 68182-0816, joonhee.lee@huskers.unl.edu)

Tones in noises from building mechanical systems can be a leading cause of complaints in an indoor environment. One of the greatest challenges to quantifying annoyance perception from these noises is the lack of information on how assorted acoustic characteristics relate to perceptual annoyance. Recent evidence suggests that annoyance perception by noises should be treated as a multidimensional rather than a unidimensional problem. Thus, the aim of this study is to explore multidimensional aspects of annoyance perception specific for building mechanical noises with tonal components. Tone frequency, tonal strength, presence of harmonics, and time fluctuation characteristics are investigated using actual noise recordings as well as artificially synthesized signals. Two subjective tests were implemented with the noise signals. Part A of the experiment is designed to identify prevailing acoustic characteristics for annoyance perception. The dominant acoustic characteristics were determined by multidimensional scaling analysis technique. A multidimensional annoyance model is subsequently proposed based on the test results. Part B is conducted to specifically investigate perceptual weighting of individual tones to overall contributions towards annoyance perceptions when complex tones are present in signals. The results of this test help to increase the accuracy of the developed annoyance model.

10:40

3aNS8. Psychoacoustically based tonality model for the evaluation of noise with tonal components. Roland Sottek (HEAD Acoust. GmbH, Ebertstr. 30a, Herzogenrath 52134, Germany, roland.sottek@head-acoustics.de)

For many years in various product assessments, tonality measurement procedures have been applied to identify and quantify prominent tonal components. The perception and evaluation of sound events containing such components has become increasingly important, e.g., in the field of vehicle acoustics for the assessment of tonality due to alternative drives, or in Information Technology (IT) devices due to hard disk drive noise. Additionally, many products include fans, e.g., IT devices, household appliances, and air-conditioning systems in buildings. These fans may emit prominent tonal sounds. The effective characterization of noise with tonal components is a challenge in acoustic and sound quality measurement. Existing methods for tonality calculation often show problems when applied to technical sounds where tonality perception is caused by different physical mechanisms like pure tones, multiple tones, narrowband noises, and even broadband noise showing very steep spectral slopes. To address this multitude of perceptual phenomena in a holistic approach, a new perceptually accurate tonality assessment method has been developed based on the hearing model of Sottek, evaluating the nonlinear and time-dependent specific loudness distributions of both tonal and broadband components via the autocorrelation function. The model has been validated by many listening tests. Its background and current state are presented.

Contributed Paper

11:00

3aNS9. Acoustic sensitivity phenomena found in people with cancer and thyroid problems. Bonnie Schnitta (SoundSense, LLC, 46 Newtown Ln., Ste. One, East Hampton, NY 11937, bonnie@soundsense.com)

Throughout my career in acoustic engineering, I have observed two interesting dynamics. Many of the people that I encountered with extreme noise sensitivity, that is easily disturbed by noise 1–2 dB above background (typically unperceivable by most), were found to be in ill health. Once I recognized this phenomena, I began to take note that there was an increase in these numbers specific to those suffering from cancer or a thyroid problem. I began to survey these individuals and most identified having been sensitive to noise long before their initial diagnosis, further developing my theory that noise sensitivity may indeed be a symptom for certain diseases, and a warning sign for early treatment. Second, those effected by ill health were specifically more bothered by lower frequency tones than those in good health. Many of these people are plagued by their noise sensitivity and describe it as a persistent nuisance that is often associated with inability to sleep with the tones present. This paper will discuss the knowledge of these facts, as well as a means to create a better healing environment in hospitals for patients that are noise sensitive.

11:15–11:40 Panel Discussion

Session 3aPP

Psychological and Physiological Acoustics: Spatial Hearing and Localization

Christopher Brown, Chair
University of Pittsburgh, 4033 Forbes Tower, 3600 Forbes at Atwood, Pittsburgh, PA 15260

Contributed Papers

8:00

3aPP1. Across-channel processing of interaural level differences. Matthew Goupell (Hearing and Speech Sci., Univ. of Maryland, 0119E Lefrak Hall, College Park, MD 20742, goupell@umd.edu)

Interaural level differences (ILDs) are produced when sounds occur away from the midline. The attenuation at the ear that is farther from the source location is frequency dependent and sometimes non-monotonic, resulting in inconsistent ILDs across frequency. It is unclear how the binaural system processes inconsistent ILDs. Ten normal-hearing listeners were presented one or three narrowband noises that varied in center frequency. Consistent or inconsistent ILDs were applied to the bands and listeners were asked to perform a left-right discrimination task. Diotic level roving was introduced to diminish listeners' ability to utilize monaural cues. Results show that thresholds were nearly frequency independent for one noise band. For three bands, thresholds became frequency dependent with the worst performance at 4 kHz, suggesting across-frequency ILD processing is a complex operation. These results have implications for people who do not heavily rely on low-frequency interaural time differences to localize sounds, such as bilateral cochlear-implant users.

8:15

3aPP2. Reverberation increases perceptual calibration to reliable spectral peaks in speech. Christian Stilp, Paul W. Anderson, Ashley Assgari, Gregory Ellis, and Pavel Zahorik (Univ. of Louisville, 308 Life Sci. Bldg., Louisville, KY 40292, christian.stilp@louisville.edu)

Preceding acoustic context influences speech perception, especially when it features a reliable spectral property (i.e., relatively stable or recurring across time). When preceding sounds have a spectral peak matching F_2 of the following target vowel, listeners decrease reliance on F_2 and increase reliance on changing, more informative cues for vowel identification. This process, known as auditory perceptual calibration, has only been studied in anechoic conditions. Room reverberation smears spectral information across time, presumably giving listeners additional "looks" at reliable spectral peaks, which should increase the degree of perceptual calibration. Listeners identified vowels (varying from [i] to [u] in F_2 and spectral tilt) presented in isolation, then following a sentence filtered to share a spectral peak with the target vowel's F_2. Calibration was measured as the change in perceptual weights (standardized logistic regression coefficients) across blocks. Listeners completed sessions where stimuli were diotic or processed by virtual room acoustics to simulate high reverberation (T_{60} = 2.97 s). Perceptual calibration was significantly greater (larger decreases in F_2 weight, larger increases in tilt weight) for reverberant stimuli than diotic stimuli. Interestingly, hearing reverberant stimuli first attenuated calibration in subsequent diotic stimuli. Results demonstrate perceptual sensitivity to the persistence of reliable spectral peaks across time.

8:30

3aPP3. The effect of manipulating interaural level differences on lateralization by bilateral cochlear implant users. Christopher A. Brown (Univ. of Pittsburgh, 4033 Forbes Tower, 3600 Forbes at Atwood, Pittsburgh, PA 15260, cbrown1@pitt.edu)

Bilateral cochlear implant (BCI) users are relatively sensitive to interaural level differences (ILDs), but have been shown to be less sensitive to interaural time differences (ITDs). A proposed method for improving spatial release from masking for BCI users was recently demonstrated to do so, but was tested in only one spatial configuration: when one sound source was to the left, and the other to the right (Brown, Ear Hear. 2014). Briefly, instantaneous ITDs were measured and converted to ILDs using a linear function in which the applied ILD increased linearly with increases in the estimated ITD. Follow-up lateralization data showed that the effect of the proposed method was to hyper-lateralize sound sources, which proved effective in the configuration originally tested but may be less effective, or even deleterious, in other spatial configurations. In the current study, exponential functions are used, in which the applied ILDs are small when the ITDs are small, and increase more rapidly as the ITD moves away from midline. Results show significantly reduced RMS error in a lateralization paradigm with exponential functions.

8:45

3aPP4. Front-back confusions when sources and listeners rotate. William Yost and Xuan Zhong (Speech and Hearing Sci., ASU, PO Box 870102, Tempe, AZ 85287, william.yost@asu.edu)

Tonal stimuli presented from directly in-front or directly in-back of a stationary listener are often miss-located in that sounds presented from in-front are occasionally perceived as being in-back of the listener; and often sounds presented in-back of a listener are perceived as being in-front. These types of front-back/back-front confusions in the azimuth plane were investigated as a function of the type of stimuli (e.g., tones, speech, and wideband noise) used and as a function of the sound or listener rotating around an azimuthal 24-loudspeaker array. Front-back confusions/back-front confusions depended on the type of stimulus; and on the number of loudspeaker positions the sound rotated through and the rate of sound or listener rotation. For example, if the sound rotated through two loudspeakers from in-front to in-back, there were a large number of confusions; if the sound rotated through each of the 24 loudspeakers on the circular array there were almost no confusions. [Research supported by an AFOSR grant.]

9:00

3aPP5. Binaural masking-patterns with short signals. Bjoern Luebken (Dept. of Experimental Audiol., Otto-von-Guericke Univ. Magdeburg, Leipziger Str. 44, Magdeburg 39120, Germany, bjoern.luebken@med.ovgu.de), Ifat Yasin (UCL Ear Inst., London, United Kingdom), G B. Henning (Colour and Vision Group, The Inst. of Ophthalmology, London, United Kingdom), and Jesko L. Verhey (Dept. of Experimental Audiol., Otto-von-Guericke Univ. Magdeburg, Magdeburg, Germany)

In masking-pattern experiments, thresholds are commonly measured for a sinusoidal signal masked by a narrow-band noise as a function of the

3a WED. AM

signal frequency. For a given spectral separation of signal and masker, beating between signal and masker may provide temporal fluctuation cues in addition to energy cues to detect the signal. In a dichotic condition, significantly broader masking patterns were found. It was hypothesized that this is due to the fact that the modulation cues do not play a major role in binaural processing. The present study investigates how masking patterns depend on signal duration. Masking patterns were not only measured for long (600 ms) signals but also for short signals (12 ms), where modulation cues should hardly play a role in signal detection even for monaural detection. The results show broader masking patterns for short signals than for long signals. In addition the binaural benefit does not change as much for the short signals than for the long signals when the signal frequency is varied. A binaural equalization cancellation model predicts the duration dependence of the masking patterns when a modulation analysis is assumed for the monaural pathway only.

9:15

3aPP6. When does binaural hearing benefit from ongoing envelope fluctuations? G. Christopher Stecker and Anna C. Diedesch (Hearing and Speech Sci., Vanderbilt Univ. School of Medicine, 1215 21st Ave. South, Rm. 8310, Nashville, TN 37232-8242, g.christopher.stecker@vanderbilt.edu)

A number of classic and recent studies suggest that listeners' access to ongoing binaural information in sounds with steady envelopes is significantly poorer than for sounds with stochastic or slowly modulated ongoing envelopes. For some types of binaural cues—e.g., envelope interaural time differences (ITD)—that result is consistent with the requirement of salient envelope features for accurate cue encoding. For interaural level differences (ILD) and fine-structure ITD, however, no such requirement exists and the result is therefore more surprising. Regardless, the overall results suggest that binaural-cue encoding-in general-relies heavily upon the occurrence of infrequent envelope fluctuations, such as sound onsets and slow (e.g., syllabic-rate) amplitude modulations. Here, the results of relevant studies are reviewed, along with conceptual and physiological models of sampling binaural cues during envelope fluctuations. Potential advantages for binaural hearing in reverberation and competing backgrounds are considered, as is the impact of such mechanisms on the localization of strongly modulated targets such as speech. Overall results suggest a general degradation of binaural information-regardless of cue type-in the absence of envelope fluctuations. Instead, the binaural system emphasizes representations of temporally sparse, but highly localizable, onset-like events in the auditory scene. [Work supported by NIH R01-DC011548.]

9:30

3aPP7. Auditory localization performance in the azimuth for tactical communication and protection systems. Jeremy R. Gaston, Timothy Mermagen, Ashley Foots, and Kelly Dickerson (Dept. of the Army, US Army Res. Lab., Attn: RDRL-HRS, Aberdeen Proving Ground, MD 21005, jgaston2@gmail.com)

An important consideration in assessing auditory abilities is the effect personal protective equipment (PPE) can have on performance. Tactical Communication and Protection (TCAP) systems are a type of PPE that is becoming more common in military applications. These types of systems provide radio communications, while also protecting the user from hazardous noise through passive and active attenuation. These systems have an active acoustic pass-through mechanism that utilizes an external microphone and internal earphone to restore environmental hearing. Past research has shown that this type of pass-through mechanism can negatively affect localization ability. In the present study, six participants localization ability was tested for two different signals (AK-47 impulse, 500 ms white noise) across an array of eight loudspeakers arranged in a ring around the test participant. The systems tested consisted of three types of in-ear TCAPs, one over-the-ear TCAPs, and one passive nonlinear hearing protection device. All systems were tested with the Advanced Combat Helmet (ACH) donned. In general, localization performance was worst for the AK-47 impulse signal. Across signals, the over-the-ear TCAPs had significantly more front/back confusions and greater localization error than the in-ear systems.

9:45–10:00 Break

10:00

3aPP8. Perceptual effects of update-delay in room auralization. Samuel W. Clapp and Bernhard U. Seeber (Audio Information Processing, Technische Universität München, Arcisstr. 21, München 80333, Germany, samuel.clapp@tum.de)

A real-time auralization system, permitting sources and receivers to move about a simulated acoustic space, offers the opportunity to study auditory perception in realistic listening scenarios. However, such a system requires substantial computing resources to achieve the low latency needed for the simulation to adjust to new conditions without introducing artifacts. To use computing resources most efficiently from a perceptual standpoint, the perceptual effects of the simulation update delays must be understood. Simulated early reflections are typically determined via the image source method. When the source or receiver moves, the positions of the image sources relative to the receiver must be updated, with higher orders requiring longer calculation times. The aim of this study is to examine the perception of sound sources in simulated reverberant rooms when a source moves to a new position with only a partial update of image source locations. The parameters tested are perceived localization and apparent source width, as a function of the highest order of image sources updated and the distance traveled by the moving source. The results are meant to aid designers of real-time auralization systems by providing a perceptual basis for the necessary update rates for image source locations. They also aim to provide insight into the perception of auditory images that are incongruous with a real room, where there is a dissociation between source and reflection positions. [Funded by BMBF 01 GQ 1004B.]

10:15

3aPP9. The influence of signal type on the internal representation of a room in the auditory system. Elizabeth Teret, Jonas Braasch, and M. Torben Pastore (Rensselaer Polytechnic Inst., School of Architecture (Architectural Acoustics), Rensselaer Polytechnic Inst., 110 8th St. - Greene Bldg., Troy, NY 12180, terete@rpi.edu)

Currently, architectural acousticians make no real distinction between a room impulse response and the auditory system's internal representation of a room. With this lack of a good model for the auditory representation of a room, it is indirectly assumed that our internal representation of a room is independent of the sound source needed to make the room characteristics audible. In a perceptual test, we investigate the extent to which this assumption holds true. Listeners are presented with various pairs of signals (music, speech, and noise) convolved with impulse responses for different rooms. They are asked to evaluate the differences between rooms and disregard differences between the source signals. Multidimensional scaling is used to determine the extent to which the source signal influences the internal representation of the room and which room acoustical characteristics are important/perceivable for each sound type.

10:30

3aPP10. Rate effects in interaural and sequential level difference perception. Bernhard Laback (Acoust. Res. Institute, Austrian Acad. of Sci., Wohllebengasse 12-14, Vienna 1040, Austria, bernhard.laback@oeaw.ac.at), Mathias Dietz, and Stephan Ewert (Medizinische Physik, Universität Oldenburg, Oldenburg, Germany)

The relative weighting of post-onset interaural level difference (ILD) cues decreases with increasing modulation rate of a signal. It is unclear, however, whether overall ILD sensitivity decreases with increasing rate, particularly if stimulus duration and loudness are kept constant. Moreover, it is unclear if the rate effect arises also in monaural sequential level discrimination. In experiment 1, ILD-based lateralization discrimination thresholds and sequential level difference (SLD) discrimination thresholds were measured using bandpass-filtered pulse trains (4 kHz) with rates of 100, 400, and 800 pulses/s. From 100 to 400 pulses/s, both ILD and SLD thresholds decreased, contrary to a rate limitation. From 400 to 800 pulses/s ILD thresholds increased while SLD thresholds remained constant. The latter result is consistent with a high-rate limitation being specific to binaural hearing. Experiment 2 evaluated whether this rate limitation is due to the loss of transmitted modulation at high rates. The ILD thresholds for an unmodulated 4-kHz pure tone were indeed higher than those for pulse-

trains. An auditory nerve model (Zilany *et al.*, 2014, JASA **135**, 283–286) combined with an interaural discharge rate comparison stage qualitatively predicted the nonmonotonic pattern of ILD thresholds. Overall, the results suggest that modulation can be beneficial for ILD perception.

10:45

3aPP11. Just noticeable differences of listener envelopment and apparent source width. Stefan Klockgether and Steven van de Par (Medical Phys. and Acoust., Carl von Ossietzky Univ., Carl-von-Ossietzky-Str. 9-11, Oldenburg, Lower Saxony 26129, Germany, stefan.klockgether@uni-oldenburg.de)

Listener envelopment (LEV) and apparent source width (ASW) are attributes which describe the spatial perception of a sound in a room. They are linked to several parameters of physically measurable binaural room impulse responses (BRIR), e.g., the interaural cross-correlation (IACC). Both the LEV and ASW decrease with increasing IACC. For this study the IACC of the BRIR of different rooms was directly manipulated to systematically change the spatial impression of a sound in a room. The BRIRs were convolved with anechoic music signals, and the resulting stimuli were presented to subjects in a psychoacoustic experiment to estimate the just noticeable differences of LEV and ASW. For that purpose, the subjects had to rate the different stimuli for both perceptual attributes in seven steps from "no envelopment" to "completely enveloped" or from "small" to "wide" respectively. In a second step, the stimuli which were rated the same, were used in a paired comparison paradigm to obtain a more precise estimation of the ability to distinguish between similar stimuli. The results of this study are used in a psychoacoustically motivated model to estimate differences in perceived spaciousness of sounds in rooms by the prediction of the perceived envelopment and source width.

11:00

3aPP12. A Bayesian framework for the estimation of head-related transfer functions. Griffin D. Romigh (Air Force Res. Labs, 2610 Seventh St., Area B, Bldg. 441, Wright Patterson AFB, OH 45433, griffin.romigh@us.af.mil), Richard M. Stern (Carnegie Mellon Univ., Pittsburgh, PA), Douglas S. Brungart (Walter Reed National Military Medical Ctr., Bethesda, MD), and Brian D. Simpson (Air Force Res. Labs, Dayton, OH)

While high-fidelity spatial auditory displays require individualized head-related transfer functions (HRTFs), much of the physical structure contained within an HRTF is similar across most individuals. This suggests that a Bayesian estimation technique, which uses sample observations to bias an a priori model towards an individual, may provide benefits in terms of efficiency. Therefore, the current work proposes a Bayesian HRTF framework that utilizes HRTFs in the form of their real spherical harmonic representation. When combined with assumptions of normality, the resulting technique is shown to enable the accurate estimation of an individualized HRTF from a small set of spatially distributed measurements. Moreover, the model provides a convenient way to analyze which components of an HRTF vary most across individuals, and can therefore be used to create very efficient HRTF measurement strategies. A perceptual localization test confirmed that similar localization performance could be attained with an HRTF estimated from as few as 12 spatial measurements, even when confined to the median-sagittal plane.

WEDNESDAY MORNING, 20 MAY 2015 KINGS 3, 8:00 A.M. TO 12:00 NOON

Session 3aSA

Structural Acoustics and Vibration: General Topics in Structural Acoustics and Vibration II

Kuangcheng Wu, Cochair
Ship Survivability, Newport News Shipbuild, 4101 Washington Ave., Newport News, VA 23693

Jeffrey Cipolla, Cochair
Weidlinger Associates, Inc, 1825 K St NW, Suite 350, Washington, DC 20006

Invited Papers

8:00

3aSA1. Acoustic radiation from an infinite submerged, line-driven plate with attached finite plate. Robert M. Koch (Chief Technol. Office, Naval Undersea Warfare Ctr., Code 1176 Howell St., Bldg. 1346/4, Code 01CTO, Newport, RI 02841-1708, Robert.M.Koch@navy.mil)

The structural acoustics and vibration related to submerged, structurally driven elastic plates is germane to a wide range of applications in U.S. Navy undersea vehicle and system designs. Of particular interest to a recent specific Naval design application was the examination of effects on the acoustic radiation from a very large, fluid-loaded elastic plate joined with a "welded," or strongly attached, smaller elastic plate subjected to force-loading excitation somewhere on the large plate. This paper presents the results following an examination of several theoretical approaches to predicting the radiated sound level (both levels and directivity) of an idealized infinite plate and perfectly attached thin, finite plate submerged in a heavy fluid and driven by a line-load at varying locations on the infinite plate. Parametric studies are also presented to both highlight the underlying basic physical mechanisms and also to investigate the effect on sound radiation of varying system parameters. Specifically, results of parametric variations in (a) location of the line force on the plate with respect to the attached finite plate, (b) length/size of the attached plate, (c) mass and/or stiffness of the attached plate, (d) frequency of the harmonic line-load, and, time permitting, and (e) the mechanical theory of the attached thin plate, are all presented herein.

8:20

3aSA2. Causal and passive interpolation of structural acoustic matrices. James G. McDaniel and Andrew S. Wixom (Mech. Eng., Boston Univ., 110 Cummington Mall, Boston, MA 02215, jgm@bu.edu)

The present work investigates the causal and passive interpolation of structural acoustic matrices in the frequency domain. Interpolations significantly ease computational burdens involved in computing these matrices from models. For example, one may wish to compute a small admittance matrix, relating forces to velocities at points on a structure, from a large finite element model with over one million degrees of freedom. In many structural acoustic systems, the elements of such matrices vary smoothly with frequency and therefore interpolation can be accurate and effective. In the time domain, causality requires no effect before its cause. In the frequency domain, this takes the form of Hilbert transform relations that relate the real and imaginary parts of scalar transfer functions. Passivity requires that the net power dissipated by the structure over a cycle of vibration be positive semidefinite. The present work develops series expansions in which these conditions may be implicitly satisfied by the proper choice of basis functions. Coefficients of in the series expansion are found by matching the series to the known values at the interpolation points. Examples demonstrate situations in which the satisfaction of these physical conditions yields more accurate interpolations.

8:40

3aSA3. A hybrid method for predicting heavy fluid loading of structures. Micah R. Shepherd and John B. Fahnline (Appl. Res. Lab, Penn State Univ., PO Box 30, Mailstop 3220B, State College, PA 16801, mrs30@psu.edu)

Measuring the resonance frequencies and damping of structures in heavy fluids can be difficult and time consuming. To alleviate this burden, a hybrid approach is proposed by combining in-air experimental modal analysis with a boundary element estimate of the acoustic resistance and reactance matrices. The method was validated on a rectangular metal plate by comparing the natural frequencies estimated using the hybrid method to the natural frequencies measured in water. The modal parameters were first measured in air with the mass-normalized mode shapes estimated using the complex mode indicator function. The modal parameters were then updated using rational polynomials curve fitting. The measured modes were then used as *in-vacuo* basis functions to compute modal resistance and reactance matrices in water and the fluid-loaded vibration and sound radiation was estimated. The natural frequencies of the plate submerged in water were then measured and compared to the frequencies predicted by the hybrid method showing good agreement.

9:00

3aSA4. Jacobian-Free Newton Krylov based iterative strategies in frequency-domain analyses. Jeffrey Cipolla (Weidlinger Assoc., Inc, 1825 K St NW, Ste. 350, Washington, DC 20006, cipolla@wai.com) and Abilash Nair (Weidlinger Assoc., Inc, New York, NY)

This study examines the theory and performance of Jacobian-Free Newton Krylov (JFNK) methods for the efficient, iterative solution of steady state dynamics of linear vibrating systems in the frequency domain. Currently, most commercial FEM algorithms employ use direct factorization of large system matrices to achieve steady state solution. Such approaches are usually quite demanding on memory and CPU requirements. Some implementations exploit iterative solutions, but still use large, assembled system matrices, requiring significant memory. The methods investigated in this work avoid the formation of a matrix completely, minimizing memory requirements and enabling much larger problems to be performed on desktop computers. Here, the Conjugate-Gradient (CG) and Transpose Free Quasi Minimal Residual (TFQMR) algorithms are studied as possibilities. These methods are of particular interest for adaptation to finite element software which uses explicit transient dynamics, because such software's optimal architecture prevents the formation and solution of a stiffness (or a tangent stiffness) matrix. In order to demonstrate the advantages of these algorithms, we choose examples that use the JFNK-CG and JFNK-TFQMR technique to show computational advantages over regular matrix based solutions, with significant decrease in memory requirements.

Contributed Papers

9:20

3aSA5. Random sound pressure fluctuations Induced by non-linear damping in a reactor column. Hasson M. Tavossi (Phys., Astronomy, and GeoSci., Valdosta State Univ., 2402, Spring Valley Cir, Valdosta, GA 31602, htavossi@valdosta.edu)

Self-excited non-harmonic oscillations due to non-linear damping in the system are produced in a reactor column. The uniform steady flow is converted spontaneously into random fluctuations in sound pressure, under the especial experimental conditions, in a reactor column that includes a thin layer of dissipative porous medium. The resulting large-amplitude non-harmonic pressure fluctuations in the air-flow are similar to the bifurcation in chaotic systems; where two or more energy states can occur simultaneously, and the system oscillates between them. Experimental results demonstrate this abrupt change in flow-regime, from steady-flow to random fluctuations. Our results show that a low-pressure shock-wave- is established that precedes the self-excitation fluctuation in the system. Results also show the existence of a threshold for flow-velocity beyond which this transition from steady to pulsating non-harmonic pressure fluctuations occur. A numerical model is developed for this behavior in terms of; FFT peak frequencies, flow-velocity, characteristic numbers, relaxation-time, and system nonlinear damping.

9:35

3aSA6. Numerical hybrid TMM-modal FEM method prediction of the vibroacoustic of sandwich panels with add-on damping. Imen Rzig and Noureddine Atalla (mécanique, univesité de Sherbrooke, 2500 Boulevard Univ., Sherbrooke, Quebec J1K2R1, Canada, imen.rzig@usherbrooke.ca)

This paper discusses the numerical modeling of the vibroacoustic response of sandwich-composite panels with add-on damping, under acoustic excitation, diffuse acoustic field (DAF). A modal synthesis approach is used for the calculation of the structural response and the Rayleigh's integral is used for the acoustic response. Since the panel has a viscoelastic core, a methodology is presented to handle efficiently the modeling of the frequency depended properties of the viscoelastic layer. A hybrid TMM-modal FEM method is used to predict the acoustic response in high frequency, using the equivalents properties of panel, which are calculated from strain energies' panel. Next, a parameters' study on the effect of the viscoelastic layer location is presented. In particular, three locations are compared: within the Honeycomb core, within the skins and added to the skin with a constraining layer. The effects of the excitation type on the vibration and acoustic response are also discussed using the hybrid: TMM-modal FEM method. Key words: Sandwich NIDA, Modal FEM method, TMM method, viscoelastic damping, acoustic response, equivalent properties.

9:50–10:15 Break

10:15

3aSA7. Reconstructing excitation forces acting on a baffled plate using near-field acoustical holography. Pan Zhou (College of Power and Energy Eng., Harbin Eng. Univ., 5050 Anthony Wayne Dr., Wayne State Univ., Detroit, MI 48202, sean.f.wu@gmail.com), Sean F. Wu (Mech. Eng., Wayne State Univ., Detroit, MI), and Wanyou Li (College of Power and Energy Eng., Harbin Eng. Univ., Harbin, Heilongjiang, China)

This paper presents numerical simulations of using nearfield acoustical holography technology to reconstruct the excitation forces acting on the surface of a baffled plate. The reason for choosing a baffled plate is that analytic solutions to the vibration responses and acoustic radiation are readily available. To acquire a better understanding of the interrelationship between excitation forces and vibro-acoustic responses of the plate, the structural intensities and power flows due to individual excitation forces and their interactions are calculated. The relative contributions of the excitations toward overall structural vibration responses of the plate are analyzed by taking the normal modes transform into the wavenumber domain. Results show that the wavenumber contents of vibration responses of a plate due to a line, point, and distributed forces have very distinct characteristics. Therefore, by analyzing the vibration responses in the wavenumber domain, it is possible to reveal the impacts of individual excitation forces on the resultant structural vibration of a plate.

10:30

3aSA8. Magnetic excitation and identification of flexural modes of a circular plate. Timothy D. Daniel, Phil L. Marston (Phys. and Astronomy Dept., Washington State Univ., Pullman, WA 99164, timothy.daniel@email.wsu.edu), Ahmad T. Abawi (HLS Res., La Jolla, CA), and Ivars Kirsteins (NUWC, Newport, RI)

A spatially localized, oscillating, magnetic field was used to induce resonant vibrations in an aluminum circular plate without direct mechanical contact. Experiments were done with the plate in air, at a free surface, and fully submerged in water. This allowed the effect of fluid loading on the plate's spectrum to be examined for both the fully loaded and half loaded case. Since the magnetic field is spatially localized, identification of the mode shapes is possible by scanning the source along the target and measuring the varying response of the plate. Additional experiments were done with an open-ended fluid filled copper circular cylinder in which both resonant frequencies and mode shapes were identified. Both targets responded at twice the oscillation frequency of the applied field [B. T. Hefner and P. L. Marston, J. Acoust. Soc. Am. **106**, 3340–3347 (1999)]. The response corresponds to the oscillation frequency of the Maxwell stress associated with combined applied field and induced eddy-current field. Excitation of both plate and cylinder was found to be measurable with a steady state and a tone burst source. [Work supported by ONR.]

10:45

3aSA9. Localized generation of a vibration pattern on a thin plate by using traveling wave control with periphery actuator array. Jung-Han Woo and Jeong-Guon Ih (Mech. Eng., KAIST, Korea Adv. Inst. of Sci., Guseong-dong, Yuseong-gu, Daejeon 305701, South Korea, j.h.woo@kaist.ac.kr)

Vibrations are rendered on the display panel of a mobile electronic device to transfer a tactile sensation to users for effective information. Previous vibration control has usually focused on the suppression of an entire area. The only choice for actuator position is the boundary of the panel due to the utilization of central part for other electric components by consideration of the practical application. Traveling wave control is adopted as the basic principle to generate a rendered vibration pattern. Proper weighting for both amplitude and phase for each actuator in the array is determined by applying the general inverse method with the relationship velocity responses and input signals of actuators. Experiments are conducted on a thin the tempered glass panel of a commercial tablet PC employing small moving-coil actuators. Good performance to fulfill the rendered 2x2 and 3x3 grid distribution of vibration magnitude was observed with more than 97 % success ratio. Also, Regularization provides 75% reduction of the control effort. Minimum number and optimum position of actuators for rendering target vibration pattern can be chosen by consideration of the independency of actuator array.

11:00

3aSA10. Building leakage detection and quantification using statistically optimized nearfield acoustic holography technique. Kanthasamy Chelliah, Ganesh G. Raman (Illinois Inst. of Technol., 10 w 32nd St., Ste. 243, Chicago, IL 60616, kchellia@iit.edu), Ralph T. Muehleisen (Argonne National Lab., Argonne, IL), Hirenkumar Patel (Illinois Inst. of Technol., Chicago, IL), and Eric Tatara (Argonne National Lab., Argonne, IL)

This paper presents a building infiltration detection and quantification technique using statistically optimized nearfield acoustic holography (SONAH) technique. A model building with known cracks on its wall was investigated in this study. The model building housed a synthetic acoustic source. A hologram measurement was performed outside the model building and the sound pressure levels on the walls were reconstructed. The correlation between the reconstructed pressure levels and the area of the crack was obtained. It was found that the acoustic technique can successfully detect and quantify the leakages from the building model. Two different frequencies of reconstructions are compared and it was found that the lower frequency reconstructions were more accurate. Effects of various regularization methods, phase matching, and quantization error are discussed. Various methods to suppress the wrap-around error in the reconstructions will be addressed.

11:15

3aSA11. Vibrational analysis of hollow and foam-filled graphite tennis rackets. Kritika Vayur (Graduate Program in Acoust., Penn State Univ., 729 S. Allen St., State College, PA 16801, kuv126@psu.edu) and Daniel A. Russell (Graduate Program in Acoust., Penn State Univ., University Park, PA)

The game of tennis is plagued with wrist and elbow injuries. Most players use light, stiff hollow graphite rackets. A recent foam-filled racket design appears to reduce the risk of wrist injury. A vibrational analysis of a wide variety of rackets was conducted in attempt to understand the science behind this racket's apparent success. Damping rates were measured for the bending and torsional modes for a wide variety of hollow and foam-filled rackets. No significant difference among the rackets was found, suggesting the benefit of the foam-filled racket design is not due to damping. An extensive modal analysis was conducted for several rackets over the frequency range of 100 Hz to 1.5 kHz to identify the bending, torsional, and string modes. In this paper, the damping rates, mode shapes, and frequencies, will be compared for several hollow and foam-filled rackets. Results will be interpreted in light of the relevant forces between the racket, ball, and arm.

11:30

3aSA12. Frequency eigenvalues of the breathing mode of a submerged empty thin spherical shell: Variation with the ratio of shell thickness to radius. Marshall V. Hall (Marshall Hall Acoust., 9 Moya Crescent, Kingsgrove, New South Wales 2208, Australia, marshallhall@optushome.com.au)

The frequency eigenvalues of the breathing mode of a submerged empty thin spherical shell are the three roots of a cubic algebraic equation. Compared with the quadratic equation for an *in-vacuo* shell, the polynomial contains two imaginary terms in the ratio of sound-speed in the ambient fluid to plate velocity of the shell material. All three roots of the equation are complex, due to radiation into the ambient fluid. For a shell submerged in a fluid similar to water, two of the eigenvalues' real parts, and two of their imaginary parts, do not vary monotonically with shell thickness. To illustrate this phenomenon, an example of a steel shell submerged in water is considered. For a thickness to radius ratio of 0.1, one of the non-dimensional eigenvalues has a real part of 1.57, close to the *in-vacuo* result of 1.61. As the thickness ratio decreases to 0.0143, this real part decreases to zero and remains there for lower ratios. Another eigenvalue, whose real part is always zero above 0.0143, has a positive real part at ratios less than 0.0141, with a maximum at 0.009. Over the interval from 0.0141 to 0.0143, however, the real parts of all three eigenvalues are zero.

11:45

3aSA13. Bias of vibrational energy at high frequencies in circular and rectangular plates. Thomas D. Boyer (Appl. Res. Lab, Penn State Univ., 200 GTWT, University Park, PA 16802, tub139@psu.edu) and Micah R. Shepherd (Appl. Res. Lab, Penn State Univ., State College, PA)

Experimental modal analysis can be used to measure the modal parameters and vibrational energy in structures. This presentation will compare biases in the total vibration energy at high frequencies for a circular and rectangular aluminum plate that are freely supported. Both rectangular and cylindrical coordinates were used to create the set of excitation points for the circular plate, while only rectangular coordinates were used for the rectangular plate. Singular value decomposition was then used to obtain the natural frequencies and mode shapes in each of these plates. The obtained modes were used to synthesize the vibration response and to compute the vibration energy. Biases were found at high frequency using a direct integration method to compute the vibration energy. The dependencies of the bias on the plate geometry and set of excitation points will be discussed.

WEDNESDAY MORNING, 20 MAY 2015 — COMMONWEALTH 2, 8:00 A.M. TO 11:20 A.M.

Session 3aSC

Speech Communication: Celebration of Kenneth N. Stevens' Contributions to Speech Communication: Past, Present, Future I

Helen Hanson, Cochair
ECE Dept., Union College, 807 Union St., Schenectady, NY 12308

Stefanie Shattuck-Hufnagel, Cochair
MIT, 36-511 MIT, 77 Mass Ave., Cambridge, MA 02139

Chair's Introduction—8:00

Invited Papers

8:05

3aSC1. Ken Stevens and linguistic phonetics. Patricia Keating (Linguist, UCLA, Los Angeles, CA 90095-1543, keating@humnet.ucla.edu)

While almost all of Ken Stevens's research has been influential in linguistic phonetics—especially his work on the quantal nature of speech and on enhancement theory—this presentation will focus more on aspects not covered by others in this session. A noteworthy aspect of Ken's career is that he frequently collaborated with linguists, notably in research on phonetic features and their structure. In work with Blumstein and with Halle, he provided acoustic correlates of place of articulation, laryngeal, and nasalization features. Much of this work is summarized in his 1980 paper in *JASA*, "Acoustic correlates of some phonetic categories." In work with Keyser, he suggested an overall organization of features to define major classes of sounds. This work was published in linguistics journals, e.g., their 1994 paper in *Phonology*, "Feature geometry and the vocal tract." Ken's importance to linguistics, as an engineer interested in linguistic sound systems and eager to work with phoneticians and phonologists, cannot be overestimated, and is a legacy continued by many of his students.

8:20

3aSC2. Distinctive features, quantal theory, and enhancement. Helen Hanson (ECE Dept., Union College, 807 Union St., Schenectady, NY 12308, helen.hanson@alum.mit.edu)

Ken's background in physics and engineering and keen interest in phonetics gave him a rare ability to suggest a quantitative basis for distinctive features. His awareness that the acoustic correlate of a smoothly varying articulatory parameter might have a sudden change in value led to his development of Quantal Theory. In addition to providing a basis for distinctive contrasts, it provides a means for a continuous-time signal to reflect the discrete nature of the underlying representation of speech sounds. Later work led to two types of quantal relations: those based on aeromechanical properties of the vocal tract, which lead to manner features, and those based on coupling between cavities, which lead to articulator-bound features. Ken also suggested that each feature has a defining articulatory state, and thus, a defining acoustic correlate. Neighboring segments or higher-level phenomena may threaten the salience of defining acoustic correlates. Enhancement theory was developed to account for the fact that the defining articulatory or acoustic correlates are threatened in some environments. At such times, enhancing gestures that bolster the salience of the defining acoustic correlates are brought into play. Much of the work produced by Ken's students was aimed at finding support for these theories.

8:35

3aSC3. The role of subglottal resonances in speech processing algorithms. Abeer Alwan (Elec. Eng., UCLA, 405 Hilgard Ave., Los Angeles, CA 90095, alwan_99@yahoo.com), Steven Lulich (Speech and Hearing Sci., Indiana Univ., IN), and Harish Ariskere (Xerox-India, Bangalore, India)

Inspired by Ken Stevens' work on speech production, we investigated the role of the subglottal resonances (SGRs) in several machine-based speech processing algorithms. The subglottal acoustic system refers to the acoustic system below the glottis, which consists of the trachea, bronchi, and lungs. The work is based on the observations that the first and second subglottal resonances (Sg1 and Sg2) form phonological vowel feature boundaries, and that SGRs, especially Sg2, are fairly constant for a given speaker across phonetic contexts and languages. After collecting an acoustic and subglottal database of 50 adults and 48 children, analyzing the database, and developing algorithms to robustly estimate SGRs, we were able to use these resonances to improve the performance of a variety of speech processing algorithms including: recognition of children's speech in limited-data situations, frequency-warping for adult speech recognition, speaker recognition and speaker verification, and automatic height estimation. [Work supported in part by the NSF.]

8:50

3aSC4. Influences of Ken Stevens on research in speech production. Joseph S. Perkell (Res. Lab. of Electronics, Massachusetts Inst. of Technol., Cambridge, MA 02445, perkell@mit.edu)

Ken Stevens has had a profound influence on many aspects of speech research. His contributions to the understanding of speech range from the theoretical through a vast body of hypothesis-driven studies to important developments in methodology. His theory of the quantal nature of speech has important implications for our understanding the nature of sound structures of languages. It embodies the concept that sounds are specified by combinations of distinctive features, which are based on universal properties of the production and perception mechanisms. Over a period of five decades, he, the many people he mentored and others tested hypotheses stemming from this theory using modeling and running experiments on relations among vocal-tract configurations, articulatory movements, tissue properties and aerodynamics—and the influences of these factors on perceptually relevant characteristics of the resulting speech sounds. Investigators who spent time in Stevens' Speech Communication Group also learned valuable principles for the conduct of scientific research. Examples of research on speech production will be presented to illustrate some of these points. [Research supported by the NIDCD, NIH.]

9:05

3aSC5. Ken Stevens' influence on aerodynamic models of fricatives. Christine H. Shadle (Haskins Labs., 300 George St., New Haven, CT 06511, shadle@haskins.yale.edu)

In the late 1950s, John Heinz, under Ken Stevens' supervision, used a constriction in a duct to model fricatives, recording the far-field sound spectra. Ken elaborated the model over the next 15 years: The constriction's location corresponded to place. The flowrate and constriction area were used to compute the Reynolds number, indicating the degree of turbulence of the flow; the pressure drop predicted the far-field sound. However, as I showed when Ken supervised my thesis in the 1980s, an obstacle downstream of the constriction created more turbulence noise, and was a better approximation to sibilants. It could still be modeled by a circuit analog with independent source and filter. Similar models of interdentals and palatals were not as successful; geometric details not included in the area function affected the acoustic output. Other limitations of early models include: circuit analogs use only the speed of sound, but the much-slower convection velocity plays a role in noise modulation in voiced fricatives; noise source changes during transitions are oversimplified; quasi-static modeling of fricatives excludes the noise produced by articulator movement. Ways of incorporating these mechanisms of fricative production in simple, conceptually powerful models are needed.

9:20

3aSC6. Ken Stevens' contributions to the field of communication disorders. Ray D. Kent (Waisman Ctr., Univ. of Wisconsin-Madison, Rm. 491, 1500 Highland Ave., Madison, WI 53705, kent@waisman.wisc.edu)

Speech is a robust faculty and most people rely on it daily, overcoming temporary difficulties such as laryngitis, dental work, or stage fright. Even slips of the tongue are relatively rare, occurring once in about a thousand spoken words. But for all its robustness, speech is vulnerable to a number of disorders that can be severe enough to interfere with efficient and natural communication. A communication disorder can be defined as a disruption or limitation in the production of speech that is fluent, intelligible, and natural. Through direct and indirect influences, Ken Stevens became a valued ally in the effort to bring relief to people with these disorders. In a highly selective account of the contributions of Ken and his colleagues, this talk examines three general areas: technological advances for the assessment and treatment of speech and voice disorders, basic theories that illuminated the path to understanding speech communication in health and disease, and quantitative research focused on speech communication in individuals with hearing loss or dysarthria (both of which have multisystem consequences on the production of speech and both of which can result in serious communication difficulties).

9:35

3aSC7. Looking into the future by looking back at the past: Observations about Ken Stevens' views on speech perception. David B. Pisoni (Psychol. and Brain Sci., Indiana Univ., 1101 E 10th St., Bloomington, IN 47405, pisoni@iupui.edu)

There has always been an interest in the perception of speech and non-speech signals. As Ken pointed out 35 years ago (Stevens, 1980), speech signals possess several highly distinctive acoustic properties that set them apart from other auditory signals. He argued that the study of the acoustic properties of classes of speech sounds that occur in natural languages may provide insights about the "special" response characteristics of the auditory system. He observed that all speech sounds have a small number of general properties in common: (1) they have a short-time spectrum that contains peaks and valleys; (2) they have up-and-down variations in amplitude as a function of time; and (3) the short-time spectrum changes over time. I summarize studies with normal-hearing and hearing-impaired listeners with cochlear implants that provide converging support for the conclusion that human listeners have developed highly efficient perceptual processing strategies to make optimal use of minimal acoustic information in the speech signal. [Work supported by grants from NIDCD to Indiana University.]

9:50–10:05 Break

10:05

3aSC8. Analysis by synthesis techniques. William Idsardi (Linguist, Univ. of Maryland, 1401 Marie Mount Hall, College Park, MD 20742, idsardi@umd.edu)

In the late 1950s and early 1960s, Ken Stevens and Morris Halle proposed *analysis by synthesis* (A-by-S) as a general model for speech perception by humans and machines. The leading idea in A-by-S is that people can generally both speak and listen, and therefore speech is both action and perception. Thus, the perception of a speech fragment can be the answer to the question, "how would I have said that?," that is, to recover the action that would produce that event (or an appropriately scaled version of the event). This idea has had lasting influence in speech perception models and systems (even if they employ other terms), as well as in more general accounts of perception. In this talk, we will offer a brief history of the idea of A-by-S, focusing on the general architecture of such systems and in what space (auditory or articulatory) signal comparison is carried out. In doing so, we will briefly compare and contrast A-by-S with other influential ideas, including the motor theory of speech perception, modularity and the special nature of speech, mirror neurons, the memory-action-perception loop, Bayesian inference, and general auditory cognition.

10:20

3aSC9. Ken Stevens, motor theory, and infant MEG brain data. Patricia K. Kuhl (Inst. for Learning & Brain Sci., Univ. of Washington, Box 357920, Seattle, WA 98195, pkkuhl@u.washington.edu)

ASA meetings provided a special time to talk to Ken Stevens and share data and opinions about topics of mutual interest. On one of these occasions, Ken uttered a sentence that stuck with me, "I hate to tell you this, but I think I'm becoming a motor theorist!" He then explained *Analysis by Synthesis*. In this talk, I'll share newly published MEG brain data (Kuhl *et al.*, *PNAS*, 2014), suggesting that young infants may indeed be engaging in something like analysis by synthesis as they listen to us speak. I'll use this and other findings to discuss Ken's important contributions to Speech Communication.

10:35

3aSC10. The MIT Lexical Access Project: A window into Stevens' model of feature-cue-based processing. Jeung-Yoon Choi and Stefanie Shattuck-Hufnagel (MIT, 50 Vassar St., Rm. 36-523, Cambridge, MA 02139, sshuf@mit.edu)

In his JASA (2002) paper, Ken Stevens proposed a model of human speech recognition based on extracting acoustic cues to the distinctive feature contrasts of the speaker's intended words. Following Halle (1995), he distinguished between cues to manner features (e.g., abrupt spectral events associated with constrictions/widenings of the vocal tract, called landmarks) vs. cues to other features related to voicing and place, and proposed that landmark detection is an early step in perception. Landmark cues are particularly useful: they are reliably produced, robustly detectable, and highly informative about the structure and lexical content of an utterance; they also identify adjacent regions rich in cues to additional features. During his working life, Ken's students developed many of the modules required to detect feature cues, meanwhile discovering important aspects of their systematic context-governed variation. Current work aims at (1) completing the speech analysis system based on detection of feature cues and parameter values, (2) evaluating its performance, and (3) comparing its performance to human perceptual behavior. A system based on Ken's insights will have implications for human speech processing models, for knowledge-based approaches to ASR, and for a deeper understanding of the mechanisms underlying clinical speech problems as well as language learning.

10:50

3aSC11. Ken Steven's research and influence on automatic speech recognition. Carol Y. Espy-Wilson (Elec. and Comput. Eng., Univ. of Maryland, A.V. Williams Bldg., College Park, MD 20742, espy@umd.edu)

This talk will address Dr. Kenneth Steven's considerable influence in the area of speech recognition. Central to Ken's model of lexical access is the relationship between acoustics and articulation. Binary distinctive features are identified from the continuous signal based on acoustic cues that are tied to articulation. As such, variability in this parameterization is reduced from that seen in the physical signal. The acoustic features are then matched against the lexicon where there is generally only one representation for each word. This lexical representation consists of a bundle of distinctive features. In contrast, state-of-the-art recognizers use acoustic features that are not tied to articulation. Consequently, they are highly variable, resulting in the recognizer's over-reliance on sophisticated machine-learning tools and statistical signal processing methodologies that are data intensive and often fail to generalize. Further, multiple lexical representations of each word in terms of acoustic units called tri-phones are needed to cope with pronunciation variability. While Ken's model has not been fully realized, it has motivated research into extraction of the distinctive features, investigations of alternate articulation-based representations, and marrying these representations with well-established recognition back-ends to study gains in robustness. Examples from studies and their promise relative to state-of-the-art methods will be given.

11:05

3aSC12. Ken Stevens' influence on the development of paradigms for speech synthesis. Brad H. Story (Speech, Lang., and Hearing Sci., Univ. of Arizona, 1131 E. 2nd St., P.O. Box 210071, Tucson, AZ 85721, bstory@email.arizona.edu)

Synthetic speech has long been used as a means of understanding both speech production and speech perception, as well as for technological applications such as text-to-speech devices. Paradigms for developing speech synthesis systems have included electrical circuits, digital filters, and computational models that replicate either the structure or acoustic characteristics of the voice source and vocal tract. This presentation will focus on how Ken Stevens' investigations of speech, spanning more than five decades, have directly influenced essentially every paradigm of speech synthesis, including formant synthesis, articulatory synthesis, and speech production modeling. [Work supported by NIH R01-DC011275 and NSF BCS-1145011.]

Session 3aSPa

Signal Processing in Acoustics: Smartphone Acoustic Signal Processing Student Competition (Poster Session)

Kevin Cockrell, Cochair
Applied Physical Sciences Corp., 2488 Historic Decatur Rd., Ste. 100, San Diego, CA 92106

Joshua Atkins, Cochair
Beats Electronics, LLC, 1431 Ocean Ave., Apt. 1018, Santa Monica, CA 90401

The ASA Technical Committee on Signal Processing in Acoustics is sponsoring a competition for students to develop concepts for smartphone applications ("apps") that make novel use acoustic signal processing. This session will include a poster describing each smartphone app concept. A panel of volunteer judges will judge each entry on the following equally weighted criteria: novelty, feasibility of implementation, technical rigor, and how well the concept is presented. Cash prizes for top three entries of US$1000, US$500, and US$300 per team will be awarded.

WEDNESDAY MORNING, 20 MAY 2015 BRIGADE, 9:00 A.M. TO 10:45 A.M.

Session 3aSPb

Signal Processing in Acoustics and Psychological and Physiological Acoustics: Cognitive Signal Processing

Grace A. Clark, Chair
Grace Clark Signal Sciences, 532 Alden Lane, Livermore, CA 94550

Chair's Introduction—9:00

Invited Papers

9:05

3aSPb1. Random linear packet coding for half duplex underwater channels. Rameez Ahmed and Milica Stojanovic (ECE, Northeastern Univ., 360 Huntington Ave., Boston, MA 02115, rarameez@ece.neu.edu)

Random linear packet coding is considered for underwater acoustic communications in situations where reliability is essential, i.e., where data packet delivery has to be guaranteed with some probability. We develop an adaptive power and rate control strategy for a half duplex acoustic channel in which the receiver can inform the transmitter about the channel state. We consider transmission in cycles, each long enough to conform to the channel's coherence. In each cycle, the receiver feeds back the channel gain and the noise power, and the transmitter adjusts its settings accordingly. Based on the experimental measurements, we model the channel gain as a log-normally distributed fading process, and provide a system optimization procedure based on a chosen criterion, such as minimum average energy per bit or maximum average throughput. System performance is compared analytically and numerically to that of a non-adaptive system, showing benefits of closed-loop, channel-aware transmission.

9:25

3aSPb2. Non-Gaussian correlated process sampling for Bayesian cognitive target classification. Grace A. Clark (Grace Clark Signal Sci., 532 Alden Ln., Livermore, CA 94550, clarkga1@comcast.net)

A cognitive signal processing system (for example, in radar or sonar) is one that observes and learns from the environment; then uses a dynamic closed-loop feedback mechanism to adapt the illumination waveform so as to provide system performance improvements over traditional systems. Current cognitive radar algorithms are designed only for target impulse responses that are Gaussian distributed to achieve mathematical tractability. Our research generalizes the cognitive radar target classifier to deal effectively with arbitrary non-Gaussian distributed target responses. Given exemplars of target impulse responses, our Bayesian illumination waveform design algorithm requires the ability to draw complex correlated samples from a target distribution specified by *both* an arbitrary desired probability density function and a desired power spectral density. This capability is realized using kernel density estimation and an extension of a new simple and efficient nonlinear sampling algorithm by Nichols *et al*. Simulations using non-Gaussian target impulse response waveforms demonstrate very effective target classification performance. We discuss practical issues with the application of the algorithms to real-world problems.

Contributed Papers

9:45

3aSPb3. Determination of material parameters to recreate realistic audio quality for sounding materials in virtual sound reproduction based on modal analysis. Muhammad Imran and Jin Yong Jeon (Architectural Eng., Hanyang Univ., 222 Wangsimni-ro, Seongdong-gu, Seoul 133-791, South Korea, mimranh1@gmail.com)

The modal synthesis method has been investigated to synthesize realistic virtual sounds for rigid bodies. The current methods to create realistic audio quality of sounding materials lack the ability to automatically determine material parameters. In this study, an approach is introduced to estimate material parameters that capture the inherent quality of sounding materials and extract perceptually significant features from the recorded example impulse responses. This approach is based on modal analysis using extracted material features to identify the best material parameters for linear modal synthesis (LMS). The synthesized audio produced from LMS preserves the same quality of material sound as recorded sound clips. A perceptual study was also conducted, revealing that the results of this approach are comparable with real recorded sounds in terms of perception of the materials.

10:00

3aSPb4. Decision-feedback equalizer based on channel estimation in the signal subspace. Tsih C. Yang (Dept. of Information Sci. and Electron. Eng., Zhejiang Univ., 38 Zhe Da Rd., Hangzhou, Zhejiang 310027, China, tsihyang@gmail.com) and San Ho Huang (College of Marine Sci., National Sun Yat-sen Univ., Kaohsiung, Taiwan)

The performance of a channel-equalizer is determined by how well the equalizer knows about the channel. For a decision-feedback equalizer based on explicit estimation of the channel impulse response (CIR), the symbol soft-decision-error (SDE) can be directly linked to the channel-estimation error. Due to channel fluctuation and the presence of noise, conventional channel-estimation lacks the required accuracy to achieve minimal SDE (<-10 dB) for a single-receiver channel. One of the approach is to use spatial diversity, where the (effective) CIR for maximum-ratio combining of the multiple-channels, often referred to as the q function, is well-structured and behaved, and relatively accurately estimated. An alternative approach, as proposed in this paper, is to explore the signal subspace for a single-receiver channel, when the multi-paths are cross-correlated as in many underwater acoustic propagation channels. Since the signal occupies a small fraction of the total space, tracking the signal in the subspace has been shown to yield much smaller (by 10–15 dB) signal prediction error (the error between the received-data and predicted-data based on the estimated CIR) and SDE in the training mode. This paper extends the method to the decision mode for a semi-time invariant channel.

10:15

3aSPb5. Improving headphone spatialization for stereo music. Muhammad Haris Usmani (School of Music, Carnegie Mellon Univ., 5000 Forbes Ave., Pittsburgh, PA 15213, usmani@cmu.edu), Ramón Cepeda Jr., Thomas M. Sullivan (Dept. of Elec. and Comput. Eng., Carnegie Mellon Univ., Pittsburgh, PA), and Bhiksha Raj (School of Comput. Sci., Carnegie Mellon Univ., Pittsburgh, PA)

Music is mixed and mastered for playback on near- and far-field speakers, which presents a problem to the growing population of listeners who listen to music primarily on headphones. Playing legacy stereo mixes on headphones places the stereo image inside the listener's head and makes the image appear ultra-wide. While this is helpful for separating the center and stereo content, it has a detrimental effect on the spatialization of the music. It makes headphone listening unnatural and uncomfortable, although listeners have learned to accept it. In this work, we develop a system that processes stereo signals to provide a better sound image to headphone listeners. The sound image is improved by adding the necessary cues to the signal so as to externalize the perceived soundstage by making it similar to the soundstage experienced inside a mixing studio. In order to work with all headphones, our system tries to maintain the mastering equalization curve of the original stereo content. We employ head-related transfer functions, de-correlation, models of reverberation, and 1/3-octave-band equalization to realize our system. The results of pilot subjective evaluations suggest that our system makes music more natural and comfortable to listen to, although with some loss of quality.

10:30

3aSPb6. Multi-modal mapping of complex data using interactive visual and aural cues. JoAnn C. Kuchera-Morin (Media Arts and Technol. (MAT), California Nanosystems Inst., Univ. of California, Santa Barbara, Santa Barbara, CA 93106, jkm@create.ucsb.edu)

At the AlloSphere Research Facility at the University of California, Santa Barbara, our research encompasses multimodal mapping of big and complex data sets. Interactive visual and sonic cues are used to navigate through this complex information. Sonic cues are used to reinforce visual cues as well as used independently, depending on the nature and structure of the data being mapped. In this presentation, I will discuss various research projects and their different or similar mapping configurations depending upon the information being extracted form the data set. We have found that sonic representation is essential for expanding our field-of-view of a large and/or complex data set, as information outside of our visual field-of-view can be sonically represented, thus facilitating visual navigation to that area of the data.

Session 3aUW

Underwater Acoustics: Historical Perspectives on the Origins of Underwater Acoustics III

David L. Bradley, Cochair
Penn State University, PO Box 30, State College, PA 16870

Thomas G. Muir, Cochair
Applied Research Laboratories, University of Texas at Austin, P/O. Box 8029, Austin, TX 78713

Chair's Introduction—9:00

Invited Papers

9:10

3aUW1. Clay and Medwin: An intuitive approach to the math and physics. Mohsen Badiey (Univ. of Delaware, 261 S. College Ave., Newark, DE 19716, badiey@udel.edu) and Tim Stanton (Woods Hole Oceanographic Inst., Woods Hole, MA)

It is interesting to look back to the first encounter with a field that one would eventually make a career. This is frequently a combination of a mentor and some initial learning in an academic environment. Clarence ("Clay") Clay and Herman ("Hank") Medwin provided such an experience for numerous researchers in the field of ocean acoustics. Being active researchers and mentors, they published a sequence of two books, both entitled *Acoustical Oceanography* (1977 and the revised version in 1998), to introduce the field to those interested in using sound to remotely sense the oceans' interior and boundaries. In addition to acousticians, readership includes marine biologists, geologists, and physical oceanographers. Although each book may look relatively simple at first glance, they present an intuitive approach to complex ocean acoustic problems with the right balance between physics, math, and real-world applications. There are also many references to more in-depth treatments of the topics. Each book provides a range of diverse topics including principles of underwater sound, sonar systems, signal processing, nonlinear acoustics, scattering from objects and creatures in the sea, the seafloor, acoustic waveguides, and many other subjects.

9:30

3aUW2. Six decades of evolution in Underwater Acoustics at the Applied Physics Laboratory, University of Washington. Kevin L. Williams, Dajun Tang, Peter H. Dahl, Eric I. Thorsos, Darrell R. Jackson, and Terry E. Ewart (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, williams@apl.washington.edu)

Professor Joe Henderson of the University of Washington physics department formed the Applied Physics Laboratory during WWII. The lab's initial efforts were to redesign the magnetic influence exploders used in US torpedoes. One of the lab's first Underwater Acoustics (UA) successes was development of transducers used in the Bikini Atoll Able test (1946). Those transducers, used to trigger other instrumentation, proved critical. Combining UA and torpedo expertise brought APL-UW to the forefront of tracking range design, construction and deployment in Dabob Bay, Nanoose, and St. Croix in the 1950s and 1960. Understanding the torpedo behavior seen in tracking ranges required measuring both the ocean environment and the acoustics within that environment. Making those measurements, as well as development and testing of models based on those measurements, also became standard operating procedure at APL, led in the 50's by Murphy and Potter. This blueprint of applied research motivating basic research, and the pursuit of basic research via ocean experiments and high fidelity modeling, continues to this day. The presentation will follow this evolution. APL-UW ocean experiments carried out during that time, as well as notable APL-UW research papers, technical reports, computer codes and textbooks, will be used as guideposts.

9:50

3aUW3. Underwater acoustics at Applied Research Laboratories, The University of Texas at Austin. Thomas G. Muir, Clark S. Penrod, and Chester M. McKinney (Appl. Res. Labs., The Univ. of Texas at Austin, P/O. Box 8029, Austin, TX 78713, muir@arlut.utexas.edu)

Several University of Texas physicists and acousticians served at the Harvard Underwater Sound Laboratory during World War II. Among them was Associate Director Paul Boner who founded Applied Research Laboratories (ARL:UT) upon his return to Texas in 1945. The underwater acoustics program began in 1949 under sponsorship of the Naval Ordnance Laboratory in support of passive sea mine sensors. The talk will focus on growth of the program in the early years including basic research, advancements in sonar engineering and signal processing, torpedo, mine and mine countermeasures (for both ocean and riverine environments), work on ship signatures, sediment physics and properties, target strength, submarine and anti-submarine sonar for both active and passive modes, *in-situ* calibration of ship transducers, environmental acoustics, long-range propagation and modeling, and nonlinear acoustics. ARL:UT's participation and support of the university's missions of research, education, and public service will also be discussed.

10:10–10:30 Break

10:30

3aUW4. Penn State's Applied Research Laboratory contributions to underwater acoustics. Edward G. Liszka (Appl. Res. Lab., Penn State Univ., P.O. Box 30, State College, PA 16804, egl4@psu.edu)

The Applied Research Laboratory (ARL) at the Pennsylvania State University has had a rich history of research and development in the field of underwater acoustics dating from its founding in 1945. The author reviews some of the important contributions made over the intervening decades in sub-fields such as sound propagation, scattering, transducers and arrays, signal processing, autonomous systems decision making, structural vibration and sound radiation, and hydro acoustics. The Laboratory was established by the US Navy as a derivative of the Harvard Underwater Sound Laboratory following World War II. Dr. Eric Walker, the first Director, was charged with continuing research and development on acoustically guided torpedoes following his highly successful effort during the war. In 1949, the Garfield Thomas Water Tunnel was constructed to further the understanding of flow-driven underwater acoustic phenomena, such as cavitation and flow induced vibration as examples of early achievements. Due to a growing need for scientists and engineers educated in acoustics, the Navy encouraged Penn State to establish its Graduate Degree Program for Acoustics in 1965. While the Program covers a broad scope of acoustic disciplines, it has a served a significant role in the development of the underwater acoustics field.

10:50

3aUW5. Underwater acoustics at the Johns Hopkins University Applied Physics Lab. Bruce K. Newhall and James W. Jenkins (Appl. Phys. Lab., Johns Hopkins Univ., 11100 Johns Hopkins Rd., Laurel, MD 20723, bruce.newhall@jhuapl.edu)

The Johns Hopkins University Applied Physics Lab (JHU/APL) was founded in 1942. JHU/APL began research in underwater acoustics in 1970 and became known for full scale ocean testing. JHU/APL developed the first technique to accurately measure the shape of a towed array. Towed array testing culminated in 1991–1992 with the deployment of a 5 km aperture measuring signal coherence and beam noise statistics in the Pacific, Atlantic, and Mediterranean. In 1982, JHU/APL began investigations into low frequency (LF) active sonar. Initially, airguns and explosives were employed to measure bottom and surface scattering strengths. Then, tests with a stationary controlled source were conducted from 1986–1989, activating both stationary and towed receiver arrays. In 1989, JHU/APL outfitted the Cory Chouest, adding a two story back deck superstructure. The lower level housed a three aperture LF source array and long towed receiver array, while the upper story berthed 50 scientists and engineers. This ship conducted a series of measurements of scattering strengths of the ocean surface, bottom and volume from 200–1000 Hz. Today JHU/APL employs 5000 staff of which about 150 are scientists and engineers working in underwater acoustics. Studies have expanded from LF beginnings to the full spectrum of acoustic frequencies.

Contributed Papers

11:10

3aUW6. A history of the study of low frequency sound absorption in the sea—An active 50 years and it isn't over yet. David G. Browning (139 Old North Rd., Kingston, RI 02881, decibeldb@aol.com), Vernon P. Simmons (335 Burgundy Rd., Healdsburg, CA), and Peter D. Herstein (24 Mohegan Rd., Charlestown, RI)

In 1965, Thorp published a compilation of low frequency data from SOFAR propagation that indicated absorption below 10,000 Hz was greater than predicted. It was suggested this may be an artifact, but the data were shown to fit a 1 kHz relaxation mechanism. NUWC (formerly NUSC) began an extensive at-sea measurement program, ranging from Hudson Bay (jointly with Canadian DREP) to Lake Tanganyika, while several other laboratories obtained data in other oceans. Yeager and Fisher, using the T-jump method at the Case-Western Laboratory, determined there was a 1 kHz relaxation in seawater associated with boron. The data collected at-sea showed, however, a variation between ocean areas, notably values from the North Pacific were 1/2 those in the North Atlantic. This puzzled supporters and encouraged critics. At Scripps, Simmons and Fisher, using a resonating sphere, showed that the boron relaxation did indeed cause the absorption of low frequency sound. Browning and Mellen discovered the key to the at-sea variation was *p*H, and at NUWC Mellen confirmed this dependence in the laboratory using Simmons' sphere. This direct connection between low frequency absorption and ocean acidification has produced several recent papers on possible implications, and increasing global warming suggests an interesting future.

11:25

3aUW7. A historical perspective: Frank Andrews and acoustics at The Catholic University of America. Joseph F. Vignola (Mech. Eng., The Catholic Univ. of America, 620 Michigan Ave., NE, Washington, DC 20064, vignola@cua.edu), Teresa J. Ryan (Eng., East Carolina Univ., Greenville, NC), Diego Turo, Shane Guan (Office of Protected Resources, National Marine Fisheries Service, Washington, DC), and John Judge (Mech. Eng., The Catholic Univ. of America, Washington, DC)

The Catholic University of America has a long history in the field of acoustics dating back to the 1930s. Some notable acousticians associated with Catholic University during the early development of the acoustics program there include Father Frances Fox, Karl Herzfeld, Theodore (Ted) Litovitz, among many others. Herzfeld and Litovitz are responsible for an enormous amount of pioneering work in the field of acoustic dissipation. Under the direction Litovitz, Acoustics moved from its original home in the Physics Department to Engineering in the early 1960s. Litovitz in turn recruited Frank Andrews. Andrews took the helm of acoustics at Catholic University and expanded the scope of the program from largely research focused to include a broader emphasis on both graduate teaching and developing the critical connection to practical Navy interests. This presentation will describe the history of acoustic research at Catholic University with a focus on Frank Andrews' accomplishments and contributions to Catholic University and the acoustics community at large.

11:40–12:00 Panel Discussion

Session 3pAA

Architectural Acoustics: Uses, Measurements, and Advances in the Use of Diffusion and Scattering Devices

Ronald Sauro, Chair
NWAA Labs, Inc., 90 Tower Blvd., Elma, WA 98541

Chair's Introduction—1:15

Invited Papers

1:20

3pAA1. Acoustical optimization of curvilinear shapes used in modern architecture. Peter D'Antonio (RPG Diffusor Systems, Inc., 651C Commerce Dr., Upper Marlboro, MD 20774, pdantonio@rpginc.com) and Richard H. Talaske (Talaske | Sound Thinking, Oak Park, IL)

Classic architecture is characterized by beautiful statuary, coffered ceilings, columns, and relief ornamentation. These beautiful features coincidentally also provided excellent sound diffusion. As architecture evolves into using less ornate surfaces, the acoustic fallout is that modern rooms may not have good sound diffusion. As a result, there is a need for scattering surfaces that compliment contemporary architecture, the way that the afore-mentioned surfaces complemented classic architecture. In order to generate these modern sound diffusing surfaces, a software program called the Shape Optimizer was developed utilizing an accurate prediction method, a diffusion coefficient metric to evaluate performance and an intelligent search engine which can quickly and efficiently navigate through the myriad shape possibilities. One of the authors (Talaske) will also present an acoustician's perspective on how to successfully meet the challenge of marrying the acoustical scattering requirements with the aesthetic goals expressed by the architect and/or interior designer. Several examples will be given by both authors illustrating the use of curvilinear diffusors and the potential of providing diffusing and absorbing surfaces with the same topology.

1:40

3pAA2. The effects of materials on the performance of acoustic diffusers. Richard L. Lenz (RealAcoustix LLC, 2637 N. Washington Blvd., #125, N. Ogden, UT 84414, RL@RealAcoustix.com)

Over the years, many different theories about the effects of material mass, finishes, coverings such as fabrics, porosity, and other physical attributes of diffuser design have been put forth. These theories have included excess absorption and/or reduced diffusion being caused by different woods or other materials being used as well as different types of paints and finishes. This paper is a study on the veracity of those theories and the effects of assorted materials in the construction and implementation of acoustic diffusers. The study will include the use of a specific diffuser design utilizing a periodic design with specific frequency ranges better known as quadratic residue diffusers. Tests will be conducted to look at specific absorption results and the correlation to diffusion efficiency, if any. In addition, different styles of QRD's will be utilized to see if any designs can overcome the various attributes in design used in diffusion. Standard and customized testing procedures will be used to derive the information presented in the paper.

2:00

3pAA3. Particle simulations advance insight into diffusion device design, efficiency, and diffuse field development and propagation. James J. DeGrandis (Acoust. First, 2247 Tomlyn St., Richmond, VA 23230-3334, jim@acousticsfirst.com)

After the development of a demonstration to illustrate acoustic diffusion using Ping-Pong balls (DeGrandis, 2011, "Sound Diffusion Explained," http://acousticsfirst.com/educational-videos-acoustic-sound-diffusion.htm), the idea of virtualizing the concept to gain more insight into diffusion from a particle physics perspective became a goal. Using open-source, Ray tracing modeling, and physics simulation software, the efficiency of different diffusor designs was evaluated. Simulations included individual elements and arrays in varied distribution and varied environments, and their impact on the propagation of sound, as well as the visualization and prediction of the development of an acoustically diffuse field. Integrating a simulated particle emitter to model the excitation, environmental physics to model atmospheric conditions, and 3d models with adjustable surface dampening properties, usable simulations with calculable and predictable properties could be run. New data were collected based on spatial diffusion, isometry, and the speed of which the excitation was broken down from a contiguous wave front into a diffuse particle field. Simulations varied from individual unit efficiency to complex room modeling with multiple emitters and scattering surfaces. This is a collection of the findings from that exercise.

2:20

3pAA4. Diffusion and scattering: A critique of existing standards and data. Ronald Sauro (NWAA Labs, Inc., 90 Tower Blvd., Elma, WA 98541, audio_ron@msn.com)

This presentation will look at the existing standards used to measure scattering and diffusion. It will point out scientific problems, unreasonable assumptions, and methodology problems in these standards. Scattering has been assumed to be a separate property of unevenly shaped materials, and therefore, a method was needed to "measure" this property. It will be pointed out that scattering is really

a part of the sound absorption characteristics of materials, i.e., diffraction and can be measured as part of the sound absorption characteristic. Scientific problems and unreasonable assumptions in the standard will also be pointed out. Diffusion can be characterized by the directivity of the energy as well as the magnitude and phase of the energy radiating from the diffuser. The methodologies used to create diffusion can be described as geometric or diffractive. The existing standards do not measure the magnitude or phase of the radiated energy. The results derived from the existing standards approximate the energy losses using scattering and the directivity measured is only used to compare diffusers but cannot be used in acoustic simulation programs.

2:40

3pAA5. A better method or methodology for the measurement of diffusers. Ronald Sauro (NWAA Labs, Inc., 90 Tower Blvd., Elma, WA 98541, audio_ron@msn.com)

This presentation is a follow-on to the presentation "Diffusion and Scattering: A critique of existing standards and data (Goodbye, Scattering)." In this presentation, we will present a method of displaying 3D directivity data for diffusers, pointing out the similarities between the directivity data for diffusers and loudspeakers, and how those similarities can be used to answer the problem of creating usable data for acoustical simulation programs. We will point out that other acoustic and electro-acoustic organizations have solved many of the problems with characterizing the spatial characteristics of sound sources and how diffusers can meet those same criteria. These solutions offer a roadmap for us to be able to characterize diffusers as well. We will also present a way of measuring these diffusers using generally accepted methods, and introduce a method of presenting this data in an easily understood way with a format that can be used in acoustic simulation programs.

WEDNESDAY AFTERNOON, 20 MAY 2015 RIVERS, 1:30 P.M. TO 2:45 P.M.

Session 3pAB

Animal Bioacoustics: Biosonar

Adrienne M. Copeland, Chair
University of Hawaii at Manoa, P.O. Box 1106, Kailua, HI 96734

Contributed Papers

1:30

3pAB1. Validating side scan sonar as a fish survey tool over artificial reefs in the Gulf of Mexico. Michael Bollinger and Richard Kline (Biological Sci. Dept., Univ. of Texas at Brownsville, 1 W University Blvd., Brownsville, TX 78520, michael.bollinger127@utb.edu)

Artificial reef placements are becoming an important part of fisheries management strategies worldwide due to the loss of natural reefs, and in the Western Gulf of Mexico due to the scarcity of hard structures and vertical relief near shore, which are essential habitat for reef fish. Current visual survey techniques can be crippled by low visibility and unpredictable currents in the Gulf of Mexico, but hydroacoustics can provide a solution to these problems. This study focuses on using ground-truthed side scan sonar technology to determine fish community biomass. Through fish abundance surveys and *in situ* fish sampling, we developed a fish survey protocol using side scan sonar to quantify fish assemblages over artificial reefs. The effectiveness of this technology for management purposes was also demonstrated by comparing it with visual census methods.

1:45

3pAB2. A design for a biomimetic dynamic sonar head. Philip Caspers, Yanqing Fu, and Rolf Mueller (Mech. Eng., Virginia Tech, 1110 Washington St., SW, MC 0917, Blacksburg, VA 24061, pcaspers@vt.edu)

The biosonar system of horseshoe bats (family Rhinolophidae) has been shown to employ unusual dynamics upon the emission as well as the reception of the ultrasonic pulses. Non-rigid changes to the shapes of the noseleaves (emission baffles) as well as the outer ears (pinnae, reception baffles) have been demonstrated to affect the properties of the emitted and received ultrasonic signals. These effects have been found in the results of numerical simulations as well as experimentation with physical prototypes. In the present work, a next-generation prototype of a biomimetic sonar head inspired by horseshoe bats is being developed. The goals for this system are to create more comprehensive and life-like dynamic baffle shape geometries as well as a better acoustic coupling between the ultrasonic transducers and the time-variant baffle shapes. Particular attention has been paid to geometry of the transition between nostrils and the noseleaf baffle. A single biomimetic system that incorporates these dynamic emission and reception baffles will enable an experimental investigation of how these two dynamic stages could be used in an integrated fashion to enhance sonar performance in real-world sonar sensing scenarios.

2:00

3pAB3. Echolocation beam shape of the Risso's dolphin (*Grampus griseus*). Adam B. Smith (Marine Mammal Res. Program, Hawaii Inst. of Marine Biology, 1933 Iwi Way, Honolulu, HI 96816, adambsmi@hawaii.edu), Laura Kloepper (Neurosci., Brown Univ., Providence, RI), Wei-Cheng Yang (Veterinary Medicine, National Chiayi Univ., Chiayi, Taiwan), Wan-Hsiu Huang, I-Fan Jen (Farglory Ocean Park, Hualien, Taiwan), and Paul E. Nachtigall (Marine Mammal Res. Program, Hawaii Inst. of Marine Biology, Kaneohe, HI)

Studies have shown that echolocation signals of some odontocete species are projected in both single and sometimes vertically dual-lobed beam shapes. In this study, the echolocation beam of a Risso's dolphin (*Grampus griseus*) was measured from a captive individual ensonifying an underwater target. Clicks were recorded with an array of 16 hydrophones and the two dimensional beam shape was explored using frequency-dependent

amplitude plots. Click source parameters were comparable to those already described for this species, while analysis revealed primarily single-lobed, and occasionally vertically dual-lobed, beam shapes. Center frequencies of click signals increased with increasing sound pressure level, while the −3 dB beam radius decreased with increasing center frequency. This study is the first to measure the beam shape of echolocation signals in *G. griseus*, which exhibits forms similar to those found in the bottlenose dolphin and false killer whale

2:15

3pAB4. The effect of mouth gape angle on beam size for echolocating *Eptesicus fuscus*. Laura Kloepper, Rebecca Wojciechowicz, and James Simmons (Dept. of Neurosci., Brown Univ., 185 Meeting St., Box GL-N, Providence, RI 02912, laurakloepper@gmail.com)

The sonar beamwidth of mouth-emitting echolocating bats is hypothesized to change according to mouth gape angle, with a narrowing of the beam predicted with wider mouth openings. To investigate the relationship between mouth gape angle and beam size, we recorded both the emitted sonar beam as well as the mouth opening for four bats during a target detection task. Sonar signals were recorded with an array of microphones, and bat mouth dynamics was recorded with an action camera recording at 240 frames per second. The video frames corresponding to each pulse emission were extracted and the mouth gape angle was measured. Sound intensity, −3 dB beam angle, and frequency characteristics were calculated from the microphone array and compared to the mouth gape angles to determine the influence of mouth opening on echolocation signals.

2:30

3pAB5. Comparing the biosonar signals of free swimming dolphins with those of a stationary dolphin in a net pen. Whitlow W. Au, Adrienne Copeland (Hawaii Inst. of Marine Biology, Univ. of Hawaii, 46-007 Lilipuna Rd., Kaneohe, HI 96744, wau@hawaii.edu), Stephen W. Martin, and Patrick W. Moore (National Marine Mammal Foundation, San Diego, CA)

The biosonar signals of two free-swimming Atlantic bottlenose dolphins performing a complex sonar search for a bottom target in San Diego Bay were compared with the biosonar signals of a dolphin performing a target discrimination task in a net pen. A bite-plate device that the dolphins carried supported a hydrophone that extended directly in front of the dolphin. A biosonar measuring tootbox (BMT) attached to the bite plate measured the outgoing biosonar signals while the dolphins conducted sonar searches. Both of the free-swimming dolphins used different biosonar search strategy in solving the problem and their biosonar signals reflect the difference in strategy. The dolphin stationed in a hoop in the pen while echolocating on a target 6 m away and reported if the indentation on a spherical target was directed toward it. The signals were parameterized by determining the peak-to-peak source levels, source energy flux density, peaked frequency, center frequency, rms bandwidth, rms duration, and the Q of the signals. Some of the characteristics of the r signals were similar for the free-swimming and stationary dolphins while some were significantly different suggesting biosonar signals used by free-swimming animals may be different than signals used by captive dolphins in restrictive environments.

WEDNESDAY AFTERNOON, 20 MAY 2015 — KINGS 1, 1:00 P.M. TO 2:05 P.M.

Session 3pAO

Acoustical Oceanography: Acoustical Oceanography Prize Lecture

Andone C. Lavery, Chair
Applied Ocean Physics and Engineering, Woods Hole Oceanographic Institution, 98 Water Street, MS 11, Bigelow 211, Woods Hole, MA 02536

Chair's Introduction—1:00

Invited Paper

1:05

3pAO1. Monitoring deep ocean temperatures using low-frequency ambient noise. Karim G. Sabra (Mech. Eng., Georgia Inst. of Technol., 771 Ferst Dr., NW, Atlanta, GA 30332-0405, karim.sabra@me.gatech.edu)

In order to precisely quantify ocean's heat capacity and ocean's influence on climate change, it is important to accurately monitor ocean temperature variations, especially in the deep ocean (i.e., at depths ~1000 m), which cannot be easily surveyed by satellite measurements. Indeed, to date, deep ocean temperatures are most commonly measured using free-drifting oceanographic floats (e.g., ARGO floats), but this approach only provides a limited spatial and temporal resolution because of the sparseness of the existing global network of oceanographic floats. On the other hand, acoustic thermometry has already been demonstrated as an important modality for measuring ocean temperature and its heat capacity using low-frequency sound. However, current implementations of acoustic thermometry requires the use of active sources; aside from the technology issues of deploying such sources, there is also the ongoing issue of the potential disturbance of marine animals. We will demonstrate a totally passive acoustic method of acoustic thermometry based only on coherent processing of low-frequency ocean noise (f<50Hz) and whose results are in agreement with classical point measurements obtained from oceanographic floats. We will discuss how passive acoustic thermometry could improve global monitoring of deep ocean temperature variations through implementation using a global network of hydrophone arrays.

Session 3pBA

Biomedical Acoustics: Biomedical Acoustics Best Student Paper Competition (Poster Session)

Kevin J. Haworth, Chair
University of Cincinnati, 231 Albert Sabin Way, CVC3940, Cincinnati, OH 45209

The ASA Technical Committee on Biomedical Acoustics offers a Best Student Paper Award to eligible students who are presenting at the meeting. Each student must defend a poster of her or his work during the student poster session. This defense will be evaluated by a group of judges from the Technical Committee on Biomedical Acoustics. Additionally, each student will give an oral presentation in a regular/special session. Up to three awards will be presented to the students with USD $500 for first prize, USD $300 for second prize, and USD $200 for third prize. The award winners will be announced at the meeting of the Biomedical Acoustics Technical Committee. Below is a list of students competing, with abstract numbers titles. Full abstracts can be found in the oral sessions associated with the abstract numbers.

All entries will be on display and all authors will be at their posters from 1:00 p.m. to 3:00 p.m.

2aBA3. Radial excursions of bound and non-bound targeted lipid-coated single microbubbles
Student author: Tom van Rooij

2aBA4. Heat and mass transfer effects on forced radial oscillations in soft tissue
Student author: Carlos Barajas

2aBA10. Effect of blood viscosity and thrombus composition on sonoreperfusion efficacy
Student author: John Black

2aBA12. Stress and strain fields produced by violent bubble collapse
Student author: Lauren Mancia

2pBA13. Broadband ultrasound scanning and frequency sweep measurement for trabecular bone with novel wideband single crystal transducer
Student author: Jian Jiao

2pBA14. Ultrasonic wave velocities in radial direction of bovine cortical bone
Student author: Yuma Nishimura

2pBA15. FDTD simulations of ultrasonic wave propagation in the cortical bone with heterogeneity
Student author: Toshiho Hata

3aBA8. Enhancement of osteogenesis and stem cell differentiation by injectable nanocomposite and dynamic acoustic radiation force stimulation
Student author: Vaishnavi Shrivastava

3aBA12. Sternal vibrations reflect hemodynamic changes during immersion: Underwater ballistocardiography
Student author: Andrew D. Wiens

4aBA6. Compensating for Scholte waves in single track location shearwave elasticity imaging
Student author: Jonathan Langdon

4aBA9. Three-dimensional finite difference models of shear wave propagation in isotropic, homogeneous soft tissue
Student author: Yiqun Yang

4pBAa4. Laser-induced vaporization and acoustic-induced cavitation of droplets
Student author: Jaesok Yu

4pBAa5. Numerical investigation of the nonlinear attenuation and dissipation of acoustic waves in a medium containing ultrasound contrast agents
Student author: Amin Jafari Sojahrood

4pBAb2. A sum-of-harmonics time-domain method to distinguish harmonic and broadband signals during passive acoustic mapping of ultrasound therapies
Student author: Erasmia Lyka

4pBAb8. Designing ultrasound fields to control the morphology of engineered microvessel networks
Student author: Eric S. Comeau

4pBAb9. A nonlinear derating method for estimating high-intensity focal pressures in tissue
Student author: Seyed Ahmad Reza Dibaji

5aBA1. Numerical evaluation of absorbing boundary layers for the transient Khokhlov-Zabolotskaya-Kuznetsov Equation
Student author: Xiaofeng Zhao

5aBA2. An improved interpolation approach for rapid simulations of pulse-echo ultrasound imaging
Student author: Leslie P. Thomas

5aBA3. Toward monodisperse ultrasound-triggered phase-shift emulsions using differential centrifugation
Student author: Kyle Stewart

5aBA4. Characterizing the pressure field in a modified microbubble flow cytometer: Using a laser Doppler vibrometer to validate the internal pressure
Student author: Cheng-Hui Wang

5aBA7. On the use of local speckle field as a correction factor for shear modulus estimates based on multiple-track-locations methods
Student author: Laurentius O. Osapoetra

5aBA8. Iterative reconstruction of the ultrasound attenuation coefficient from backscattered signals
Student author: Natalia Ilyina

5aBA11. Three-dimensional pulsation of rat carotid artery bifurcation observed using a high-resolution ultrasound imaging system
Student author: Changzhu Jin

5aBA13. Novel use of ultrasound imaging to decode activity of forearm muscles for upper extremity prosthetic control
Student author: Nima Akhlaghi

WEDNESDAY AFTERNOON, 20 MAY 2015 BALLROOM 4, 1:00 P.M. TO 2:05 P.M.

Session 3pID

Interdisciplinary: Hot Topics in Acoustics

Michelle C. Vigeant, Chair
Graduate Program in Acoustics, The Pennsylvania State University, 201 Applied Science Bldg., University Park, PA 16802

Chair's Introduction—1:00

Invited Papers

1:05

3pID1. Hot topics in speech communication: Acoustics of regionally accented speech. Robert A. Fox and Ewa Jacewicz (Speech and Hearing Sci., The Ohio State Univ., 110 Pressey Hall, 1070 Carmack Rd., Columbus, OH 43210-1002, fox.2@osu.edu)

Pronunciation patterns of native speakers of American English vary across the country in a systematic way, being shaped in part by social and cultural factors in a particular geographic area. The detectable regional (or dialect) variation in speech can considerably alter speech recognition and perceptual processing. The intelligibility of regionally accented speech has its source in acoustic variation present in dynamic sound structures, which convey an unprecedented richness of acoustic phonetic details along multiple dimensions in space and time. The challenge for speech communication research is to understand which complex acoustic structures are essential in shaping auditory sensitivity to regional variation and which are coexistent and necessarily redundant. New directions in this research area include acoustic explorations of complex interactions among spectral, source, and temporal components whose unique balance is modified by regional and social influences. Recent advances in these acoustic explorations will be presented.

1:25

3pID2. Hot topics in noise. Scott D. Sommerfeldt (Dept. of Phys., Brigham Young Univ., N181 ESC, Provo, UT 84602, scott_sommerfeldt@byu.edu)

There are a number of diverse areas of interest in TC Noise. Many of these have developed in response to adverse human reaction to various noise sources. This paper will briefly overview some of these current topics to give some exposure to what is happening in "Noise." These include wind turbine noise, which has become of concern with the development of wind farms near residents; soundscapes, which is focused on engineering a desirable environment in the settings where we live; high amplitude jet noise, which is becoming of greater concern as source levels increase, and humans and animals are exposed to higher sound pressure levels; and active noise control, which continues to mature and is beginning to show up in more applications in society.

1:45

3pID3. Hot topics in physical acoustics. Joel Mobley (Phys. and Astronomy/NCPA, Univ. of MS, PO Box 1848, 1034 NCPA, University, MS 38677, jmobley@olemiss.edu)

From the astronomical to the molecular, acoustics is central to phenomena across a vast spectrum of length scales. This talk provides an overview of recent developments in physical acoustics across the size continuum, while looking more in depth in three areas. At the larger end of the scale, the generation of the infrasound by cyclonic storms is considered, specifically looking at the infrasonic signatures that can be used to track and characterize these systems. At intermediate length scales, advances in acoustic metamaterial (AM) research are examined. Specific examples will include energy-harvesting metasurfaces and soft matter systems. Moving to the small end of the scale, phoxonic crystals (PxC) are discussed. PxC's possess both optical and acoustic band gaps and can be used to trap light and sound together, enhancing the strength of acousto-optic interactions.

WEDNESDAY AFTERNOON, 20 MAY 2015 KINGS 2, 1:00 P.M. TO 2:20 P.M.

Session 3pMU

Musical Acoustics: General Topics in Musical Acoustics I

Randy Worland, Cochair
Physics, University of Puget Sound, 1500 N. Warner, Tacoma, WA 98416

Whitney L. Coyle, Cochair
The Pennsylvania State University, 201 Applied Science Building, University Park, PA 16802

Jack Dostal, Cochair
Physics, Wake Forest University, P.O. Box 7507, Winston-Salem, NC 27109

Chair's Introduction—1:00

Contributed Papers

1:05

3pMU1. Acoustic characteristics of a guitar. Corey A. Taylor (Graduate Program in Acoust., The Penn State Univ., 130 Famstead Ln., Apt. #171, State College, PA 16803, cat240@psu.edu) and Daniel A. Russell (Graduate Program in Acoust., The Penn State Univ., University Park, PA)

A variety of tests were done on an acoustic guitar to help characterize and quantify the radiated sound. Modal analysis was performed to determine resonant low frequency modes of the guitar and the motion of the air in the sound hole. Mechanical impedance and mobility measurements were taken at the bridge. String decay and radiated decay times were measured at several fret positions to study the interaction between modal resonances and decay times. Many of these individual tests have been performed on acoustic instruments before. This paper will combine the analysis of all tests and measurements to show that the modal frequencies play a large role in affecting the sound quality of an acoustic guitar.

1:20

3pMU2. Using spectral analysis to evaluate flute tone quality. Ron Yorita (Comput. Sci., California Polytechnic State Univ., PO Box 834, Morro Bay, CA 93443, ron.yorita@gmail.com) and John Clements (Comput. Sci., California Polytechnic State Univ., San Luis Obispo, CA)

Many skilled flutists place a high priority on "good" tone quality, or timbre. Unlike pitch and rhythm, timbre is difficult to objectively quantify. This project explores (1) how tone quality is described by skilled flutists, (2) whether the harmonic spectrum has some correlation with tone quality, (3) whether certain harmonic spectra are preferred, or considered "good." Thirty-one flutists ranging from high school students to professionals were recorded. A set of samples was used in surveys and interviews to capture descriptors and ratings of tone quality. All of the recorded samples were analyzed using application programs, Harmonic Analysis Tools (HAT), created for this study. HAT uses digital signal processing techniques to produce "spectral signatures". The signatures consist of the harmonic content, pitch, and amplitude of a sample. The outcome of this research is a baseline set of some often used descriptors. In addition, results showed some correlation between harmonic spectra and descriptors. There were also trends in preferences with respect to certain spectral characteristics. An unexpected finding was that University students showed divergent timbre preferences compared to highly experienced flutists.

1:35

3pMU3. A full double tuba in F/BBb. Frederick J. Young (Communications and Arts, Univ. of Pittsburgh at Bradford, 800 Minard Run Rd., Bradford, PA 16701, youngfj@youngbros.com)

Described here is a design for an F/BB^b tuba. The valves and keys needed for a full double tuba take a lot of space. There exists a full and complete double tuba in BB^b/EEE built on a very large instrument frame and is more than 5 feet tall and 2 feet wide [1]. However, the sizes of F tuba frames are too small to accommodate 5 double and 2 switch valves and their associated tubing. Using a standard F tuba frame, the lack of space has been solved by having four double valves and two switch valves. The fourth double valve is unusual because it is descending on the F side and ascending on the BB^b side of the instrument. An ascent to a BB □tuba is made by the ascending part of that valve. The valve slide lengths have been optimized to yield intonation such that $-5<$ intonation error < 5 cents over a five semitone descent from an open note. Various acoustic measurements are reported for the instrument which will be exhibited. [1] F. J. Young, "Five valve compensating brass wind instrument," Proc. Meetings Acoust. **11**, 035004 (2011).

1:50

3pMU4. Experimental investigation of individual musicians on tonal quality for saxophones and clarinets. Charles E. Kinzer (Dept. of Music, Longwood Univ., Farmville, VA 23901, kinzerce@longwood.edu), Walter C. McDermott, and Stanely A. Cheyne (Phys. and Astronomy, Hampden-Sydney College, Hampden-Sydney, VA)

An experimental investigation designed to quantify the effects of individuals on the tonal quality of a given saxophone, mouthpiece and reed combination, as well as a clarinet, mouthpiece, and reed combination has been performed. Several musicians, using the same instrument of each configuration, played a single note from each of these instruments which was digitally recorded. A Fourier transform of the recorded waveforms was used to compare the frequency spectrum from each musician. The results of the comparison will be reported and discussed.

2:05

3pMU5. Air columns for demonstration of upstream/downstream symmetry in reed wind instruments. Peter L. Hoekje (Phys. and Astronomy, Baldwin Wallace Univ., 275 Eastland Rd., Berea, OH 44017, phoekje@bw.edu)

The acoustics of a reed wind instrument such as a clarinet or oboe, and including the lip-reed instruments such as the trumpet, are classically described by the input impedance of the instrument air column and the dynamical flow control characteristics of the reed valve. The behavior of the system is largely dependent on the resonances of the instrument air column. A fuller explanation however includes both the downstream (usually the instrument) and the upstream (usually the player's wind way) input impedances in symmetry. To demonstrate this, a clarinet-like instrument with no resonance peaks can be played using the resonances of the player's wind way. Another demonstration uses a resonant tube for the upstream air column, with the playing frequency or fundamental based upon a resonance of this upstream tube. The downstream air column has a resonance whose frequency and damping can be set independently of that other resonance. This allows exploration of the effect of a resonance at a harmonic of the playing frequency.

WEDNESDAY AFTERNOON, 20 MAY 2015 — KINGS 5, 2:00 P.M. TO 3:15 P.M.

Session 3pNS

Noise: Environmental Noise and Noise Control Elements

Peter Newman, Chair

Recreation, Park and Tourism Management, The Pennsylvania State University, University Park, PA 16802

Contributed Papers

2:00

3pNS1. A spatially explicit estimate of environmental noise exposure in the contiguous United States. Daniel Mennitt (Elec. and Comput. Eng., Colorado State Univ., 1201 Oakridge Dr., Ste. 100, Fort Collins, CO 80525, daniel_mennitt@partner.nps.gov), Kurt Fristrup, and Lisa Nelson (National Park Service, Fort Collins, CO)

Environmental noise is widespread across the United States, the spatial patterns of which are dependent on a complex linkage of environmental and socioeconomic factors. Chronic exposure brings with it adverse consequences to terrestrial organisms; effects on human health and well-being include hypertension, cardiovascular disease, sleep disturbance, cognitive impairment, and annoyance. Assessments of noise exposure are essential to understand the extent of impact as well as inform land use planning and noise abatement strategies. Using extensive empirical data and a geospatial framework, we modeled the day-night average sound level (L_{dn}). The dominant factors driving sound levels are land use, climate, population, and proximity to traffic corridors. Model predictions were mapped to reveal the spatial distribution of expected sound pressure levels across the contiguous United States at a resolution of 270 m. The expected L_{dn} was compared

with localized population density to estimate the number of inhabitants exposed to levels that put them at increased risk for non-auditory health effects. The magnitude and extent of noise exposure suggests a substantial opportunity to enact measures that will improve the quality of life for many Americans.

2:15

3pNS2. Visitor assessment of anthropogenic noise at Grand Canyon National Park. Pranav K. Pamidighantam and Paul Schomer (Schomer and Assoc., Inc., 2117 Robert Dr., Champaign, IL 61821, ppamidig@illinois.edu)

In the last decade, multiple studies have been conducted at national parks using the traditional uni-polar annoyance scale and the dose-response concept to determine the effects of aircraft noise on park visitors' experiences. This survey instead uses a bipolar "pleasantness" scale and breaks each visitor's hike into segments so as to understand the efficacy of the pleasantness scale as a metric for visitor perceptions of aircraft noise. This survey was conducted on three different trails at the Grand Canyon: the Hermit trail, a backcountry trail with a large amount of aircraft noise; the Widforss trail, a backcountry trail with minimal aircraft noise; and the Bright Angel trail, a maintained trail with a small amount of aircraft noise but many hikers. Many visitors began hiking before aircraft started flying so the use of segments allowed for a longitudinal study between segments with and without aircraft noise. Through the use of these segments, correlations are drawn and regression models fit between pleasantness and various metrics, such as Leq or percent time audible. The results to date show "pleasantness" to be a viable alternative metric to quantify visitor perceptions of aircraft noise in national parks.

2:30

3pNS3. The impact of anthropogenic noise on wetland habitats. Adrienne Hopson-Costa, Regis Antonioli, Julia Crispim da Fontoura, and Ferenc de Szalay (Dept. of Biological Sci., Kent State Univ., 256 Cunningham Hall, Kent, OH 44242, ahopson2@kent.edu)

Impacts of anthropogenic noise is an important but poorly studied aspect of wetland ecology. Many aquatic animals use sound to attract mates, coordinate movements, defend territory, and detect predators. However, anthropogenic noise (e.g., from boat or automobile motors) are increasingly important in wetlands. Therefore, it is important to understand how anthropogenic noise pollution affects these habitats. I sampled wetland soundscapes in urbanized and rural areas in northeastern Ohio, and used Raven software to analyze the data. Above the water line, average sound power was 38.9db–73.9db, which was similar to below the water line (48.6db–78.5db). Frequencies in the band 0–5 kHz had the highest overall power both above and below the water. One of the most urbanized sites had above/below readings as high as 83.9/79.3 dB. In comparison, one of the most rural sites had readings as low as 47.6/66.8 dB. At both urbanized and rural sites, the 0–5 kHz band had the highest overall power. I further tested if wetland invertebrates could be impacted by anthropogenic sounds. I found that *Procambarus acutus* crayfish produced audible clicks (frequency range 6 kHz–45 kHz, average power 69–73.5 dB). Ongoing laboratory research is testing if motorboat noise affects *P. acutus* behavior and sound production.

2:45

3pNS4. Bayesian inverse analysis of multilayer acoustic porous media. Cameron J. Fackler (3M Personal Safety Div., 7911 Zionsville Rd., Indianapolis, IN 46268, cameron.fackler@mmm.com), Kirill V. Horoshenkov (Dept. of Mech. Eng., Univ. of Sheffield, Sheffield, United Kingdom), and Ning Xiang (Graduate Program in Architectural Acoust., Rensselaer Polytechnic Inst., Troy, NY)

Porous materials provide sound absorption and noise control in various applications. In many scenarios, different porous materials may be combined into multilayer absorbers to enhance the absorptive properties. We apply the Bayesian inference framework to analyze such multilayer porous materials, developing a method to determine simultaneously the number of constituent layers as well as the physical properties of each layer in a multilayer porous material. The model-based analysis combines a measurement of the acoustic surface impedance or absorption coefficient of a potentially multilayered material sample with a transfer-matrix formulation of multilayer porous material acoustic propagation models. For each sample to be analyzed, the number of layers considered in the propagation model is varied, and Bayesian evidence is computed for each case. Selecting the model with the highest evidence parsimoniously determines the number of layers present in the sample. Once the number of layers has been determined, Bayesian parameter estimation inversely determines the physical properties of each layer by estimating the input parameters of the multilayer propagation model. The proposed method automatically determines the number of layers and physical parameters of a multilayer material without any a priori knowledge of these values.

3:00

3pNS5. Modeling of acoustic resonators and resonator arrays for use in passive noise control. Matthew F. Calton and Scott D. Sommerfeldt (Brigham Young Univ., 266 N. 300 E. #26, Provo, UT 84606, mattcalton@gmail.com)

Acoustic resonators, such as the Helmholtz and quarter-wave resonator, can be used to attenuate unwanted noise in a space. Classic formulations can be used to approximate resonator performance for a given resonator configuration, but may lack sufficient accuracy for some applications. This research aims to improve the analytical characterization of resonators to provide better correlation to experimental results. Using higher order approximations and proper end corrections, more accuracy can be obtained in calculating the impedance and resonance frequency of a single resonator, which will then carry over into the overall configuration of the model. The impedance of a system of resonators in parallel is also considered, where the effects of acoustic coupling can be observed. Resonators with complex, non-ideal geometries are explored for applications where space is limited.

Session 3pPP

Psychological and Physiological Acoustics: Auditory Neuroscience Prize Lecture

Michael G. Heinz, Chair
Speech, Language, and Hearing Sciences & Biomedical Engineering, Purdue University, 500 Oval Drive, West Lafayette, IN 47907

Chair's Introduction—1:15

Invited Paper

1:20

3pPP1. Relating physiology to perception: The case of the notched-noise masker. Laurel H. Carney (Biomedical Eng., Univ. of Rochester, 601 Elmwood Ave., Box 603, Rochester, NY 14642, Laurel.Carney@Rochester.edu)

Sharpness of peripheral tuning is an essential factor in auditory signal processing. The notched-noise (NN) masking paradigm (Patterson, 1976, JASA) is an ingenious strategy used in numerous studies to estimate peripheral tuning bandwidths, filter shapes, and level-dependence in normal and impaired ears. Interpretation of NN results is based on the power-spectrum model of masking; however, the fact that NN thresholds are little influenced by a roving-level paradigm calls this interpretation into question (Lentz *et al*., 1999, JASA). Here, we explore neural cues for detection in NN in auditory-nerve (AN) and midbrain (inferior colliculus, IC) responses. Strong low-frequency fluctuations in AN discharge patterns associated with sharp spectral edges provide effective inputs for midbrain neurons tuned to low-frequency fluctuations. Addition of a tone target reduces the fluctuations in some frequency channels, creating a strong contrast in the fluctuation profile along the frequency axis. We explore the NN paradigm using computational models for AN and amplitude-modulation (AM) tuned IC neurons. Recordings from band-pass AM tuned neurons in the IC of awake rabbit support the hypothesis that the midbrain response profile can explain perception in the NN paradigm. Interpretation of NN masking results should include not only peripheral tuning but also central processing mechanisms.

Session 3pSA

Structural Acoustics and Vibration: General Topics in Structural Acoustics and Vibration III

Kuangcheng Wu, Cochair
Ship Survivability, Newport News Shipbuild, 4101 Washington Ave., Newport News, VA 23693

Jeffrey Cipolla, Cochair
Weidlinger Associates, Inc., 1825 K St. NW, Suite 350, Washington, DC 20006

Contributed Papers

1:00

3pSA1. Vibration analysis of damped stiffened beams and plates by complex modes and frequencies. Giora -. Rosenhouse (swantech, 89 Hagalil Str., Haifa 3268412, Israel, fwamtech@bezeqint.net)

Stiffened beams and plates are used in design of structures forced by machinery excitation, which result in flexural motion having distorted normal modes due to the effect of both stiffeners and damping. Such plates and beams are often damped externally or internally, which leads in turn to complex modes and natural frequencies. The modes, which are normal to the direction or axis of the stiffeners can be calculated by different techniques, such as the "three moments" method for stiffened plates or beams. The effect of the stiffeners is given by finding the changes in frequencies and normal modes as related to the ratio of the stiffeners'- and plate's or beam's rigidities. The paper gives insight into complex modes and the eigenvalue for each such mode and also their relation to the common definition of modal shapes magnitude and frequencies and concludes with solved examples for stiffened plates and beams models.

1:15

3pSA2. Acoustical field and vibration in hearing aid devices. Michael Salameh (Starkey Hearing Technologies, 6600 Washington Ave. S., Eden Prairie, MN 55344, michael_salameh@starkey.com)

One of the main considerations in the hearing aid design is the feedback from the receiver (speaker) to the microphone. Transducer characteristics, mechanical design, material properties, and test setup are among the factors that affect the feedback in hearing aid devices. Characterizing the vibration and the acoustical field generated from different components of the hearing aid device is one of the main challenges in the feedback study and control. In this paper, the acoustical field and the vibration in a hearing aid device are evaluated. Simulation and measurement data are presented and analyzed.

1:30

3pSA3. Resonant ultrasound spectroscopy with large attenuation and arbitrary geometry. Marcel Remillieux, TJ Ulrich, and Pierre-Yves Le Bas (Los Alamos National Lab., Geophys. Group (EES-17), Mail Stop: D446, Los Alamos, NM 87545, mcr1@lanl.gov)

Resonant ultrasound spectroscopy (RUS) is a powerful and established technique for measuring material properties. The first step of this technique consists of extracting resonance frequencies and attenuations from the vibrational frequency spectrum measured on a sample with free boundary conditions. An inversion technique is then used to retrieve the elastic tensor from the measured resonance frequencies. As originally developed, RUS has been mostly applicable to (1) weakly attenuating media where each resonance frequency can be clearly identified and (2) relatively simple geometries where analytical solutions exist. In this presentation, the possibility of using RUS in a highly attenuating medium and on a sample of arbitrary geometry is explored. The Kumaresan-Tufts algorithm is used to fit a sum of exponentially damped sinusoids with closely spaced frequencies to the measured frequency spectrum. The inversion of the elastic tensor is achieved with a genetic algorithm, which allows searching for a global minimum within a discrete and relatively wide solution space. The accuracy of the proposed approach is evaluated against numerical data for which the solutions are known *a priori*. [This work was supported by the U.S. Dept. of Energy, Fuel Cycle R&D, Used Fuel Disposition (Storage) campaign.]

1:45

3pSA4. High order modal approximation techniques in the frequency domain. Andrew S. Wixom and James G. McDaniel (Mech. Eng. Dept., Boston Univ., Boston, MA 02215, awixom@bu.edu)

This paper presents high order modal methods that increase the accuracy of the standard modal truncation scheme with an emphasis on frequency-domain accuracy. The existing Mode Acceleration and Modal Truncation Augmentation methods are compared alongside a new interpretation utilizing the interpolating polynomial. This technique is motivated by taking limits of the modal sum, introducing a residual that appears as a Laurent polynomial. The results presented focus on the case when the frequency range of interest is in the middle of the system's spectrum of eigenvalues, meaning that at least some of the system's natural frequencies lie below the lower bound of the frequency range. This case allows for a possible error in the Modal Truncation Augmentation method that is demonstrated here. Also, convergence properties of the methods are discussed and demonstrated.

2:00

3pSA5. "Negative" acoustic signal propagation in a media with resonant density and compressibility response functions. Valentin Burov, Konstantin Dmitriev, and Sergei Sergeev (Phys., Moscow State Univ., Leninskie Gory, Moscow 119991, Russian Federation, sergeev@aesc.msu.ru)

The properties of acoustical double-negative media strongly depend on the frequency of a propagating wave. Particularly, the effective density and compressibility of a medium can be negative only at the certain frequency band. So it is very important to consider nonstationary processes in such media. The numerical simulations of the medium characterized by resonant-type density and compressibility response functions was performed in the article. It was based on the deduced matrix equation of a Lippmann-Shwinger type. The layer of "resonant" medium was placed inside an infinite background with constant density and compressibility. The incident impulse had a gauss form modulated by a certain carrier frequency. The effective density and compressibility of the layer change from both positive to both negative values while the impulse propagates through it. The phase

velocity also becomes negative and the group velocity remains positive satisfying the causality principle. In the case of long enough impulse duration the layer behaves like it is made of a double-negative medium.

2:15

3pSA6. Vibration isolation design of railroad tracks within Ankara high speed train station. Salih Alan and Mehmet Caliskan (Dept. of Mech. Eng., Middle East Tech. Univ., Ankara 06800, Turkey, caliskan@metu.edu.tr)

Ankara serves as a hub for high speed railway operations within Turkey. A new station building of nine stories in total, which also incorporates a shopping mall and a hotel inside, is currently under construction. Need arises for controlling vibrations due to passing trains through the building. Finite element model of the building is developed by commercial software. The interaction of the building with the foundation is represented by a number of spring and damper combinations whose characteristics are obtained from the existing knowledge of soil-structure interactions. The dynamic stiffness characteristics of isolating layer under the railway slabs are sought. The FRF's between several floors of the building and the base of the track are obtained from the finite element model, to be coupled with FRF's of the train-track-isolating layer system. Vibration isolation design is carried with respect to vibration criteria in Turkish environmental noise regulations.

2:30

3pSA7. Reduction of low frequency scattering from a cylindrical elastic structure using a multi-element multi-path design. David Raudales and Donald B. Bliss (Mech. Eng., Duke Univ., Edmund T. Pratt Jr. School of Eng. Duke University, Box 90300, Hudson Hall, Durham, NC 27708, dr78@duke.edu)

Acoustic scattering of an incident plane wave off an elastic cylinder can be reduced by a multi-element multi-path (MEMP) design. Such a design replaces a single structural element with a system of elastically coupled subsystems, and utilizes the inherent mechanical properties of these coupled substructures, rather than damping, to tune its dynamic response. MEMP structures provide an innovative approach by increasing the number of wave transmission paths in this higher order system, thereby introducing physical phenomena not present in single element structures. Previous analytical and experimental work on vibration transmission in MEMP thin beams and shells demonstrates substantial wide-band attenuation. Current research considers the application of the method to underwater scattering from 2-D concentric thin cylindrical shells with azimuthally continuous radial elastic coupling. Modal decomposition of the thin shell equations is employed for mode-by-mode impedance matching between the Bessel function harmonics of the incident fluid planar wave and the polynomial functions of the mechanical structure. Quasi-static spring and mass impedance matching with the monopole and dipole modes ensures scattering reduction at lower frequencies, while novel resonance matching between the fluid and structure extends reduction into higher frequency ranges. Very promising results reveal possible applications in underwater naval submersibles.

2:45

3pSA8. Refinements to the relaxation model for sound propagation in porous media. D. K. Wilson (U.S. Army Cold Regions Res. and Eng. Lab., Engineer Res. and Development Ctr., 72 Lyme Rd., Hanover, NH 03755-1290, D.Keith.Wilson@usace.army.mil)

The relaxation formulation for sound propagation in porous media [D. K. Wilson, *J. Acoust. Soc. Am.* **94**(2), 1136–1145 (1993)] was intended to provide a phenomenologically correct model with relatively simple equations. This paper presents several refinements, which connect the relaxation model more clearly to others in the literature, and clarify interpretation of the parameters. In particular, the relaxation function is applied to the dynamic tortuosity (rather than the inverse of the complex density), and the vorticity and thermal relaxation times are related directly to the length scales of Allard and Champoux [J.-F. Allard and Y. Champoux, *J. Acoust. Soc. Am.* **91**(6), 3346–3353 (1992)]. When formulated in this manner, the relaxation and Allard-Champoux models coincide exactly in the low-frequency/small-pore and high-frequency/large-pore limits, although interpolation between these limits differs somewhat. The revised relaxation model can be readily transformed into a causal time-domain model, involving convolutions between the acoustic fields and a relaxation response function. The basic formulation has five parameters: porosity, static flow resistivity, tortuosity, and two relaxation times. A sequence of model reductions is described, leading finally to two-parameter (porosity and static flow resistivity) equations for the impedance and complex wavenumber, which provide a useful generalization of the Delany-Bazley equations.

WEDNESDAY AFTERNOON, 20 MAY 2015 COMMONWEALTH 2, 2:15 P.M. TO 3:15 P.M.

Session 3pSC

Speech Communication: Celebration of Kenneth N. Stevens' Contributions to Speech Communication: Past, Present, Future II

Helen Hanson, Cochair
ECE Dept., Union College, 807 Union St., Schenectady, NY 12308

Stefanie Shattuck-Hufnagel, Cochair
MIT, 36-511 MIT, 77 Mass Ave, Cambridge, MA 02139

Open Microphone Session

Plenary Session and Awards Ceremony

Judy R. Dubno,
President, Acoustical Society of America

Presentation of Certificates to New Fellows

Robert D. Celmer – For contributions to undergraduate education in acoustics

Frederick Gallun – For contributions to cognitive issues in hearing impairment

Kirill V. Horoshenkov – For contributions to outdoor sound propagation, remote sensing, and acoustics of porous materials

Jeffrey A. Ketterling – For contributions to cavitation luminescence and very high frequency sensing and imaging

Jennifer J. Lentz – For contributions on hearing loss and the perception of complex sound

Bart Lipkens – For contributions to undergraduate education and practical applications of nonlinear acoustics

Arlene C. Neuman – For contributions to classroom acoustics and hearing aid development

Sunil Puria – For contributions to middle- and inner-ear biomechanics and their practical applications

Purnima Ratilal – For contributions to bioacoustics and underwater acoustic scattering and reverberation

Juan Tu – For contributions to the physical acoustics of ultrasound contrast agents

Introduction of Award Recipients and Presentation of Awards

Aaron Moberly, 2015 Research Grant in Speech Science of the
American Speech Language and Hearing Foundation

Lily M. Wang, 2015 Student Council Mentoring Award

Richard Ruby, John Larson, and Paul Bradley, The American Institute of Physics Prize
for Industrial Applications of Physics

Laurel H. Carney, William and Christine Hartmann Prize in Auditory Neuroscience

Karim S. Sabra, Medwin Prize in Acoustical Oceanography

Matthew W. Urban, R. Bruce Lindsay Award

Henry Cox, Helmholtz-Rayleigh Interdisciplinary Silver Medal

Gerhard M. Sessler, Gold Medal

Barbara G. Shinn-Cunningham, Vice President's Gavel

Judy R. Dubno, President's Tuning Fork